Administrator Guide for Avaya Communication Manager

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About this book

Overview

Avaya Communication Manager is the centerpiece of Avaya applications. Running on a variety of Avaya S8XXX Servers and DEFINITY Servers, and providing control to Avaya Media Gateways and Avaya communications devices, Communication Manager can be designed to operate in either a distributed or networked call processing environment.

Communication Manager carries forward all of a customer's current DEFINITY capabilities, plus offers all the enhancements that enable them to take advantage of new distributed technologies, increased scalability, and redundancy. Communication Manager evolved from DEFINITY software and delivers no-compromise enterprise IP solutions.

Communication Manager is an open, scalable, highly reliable and secure telephony application. The software provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking.

Purpose of this book

This book describes the procedures and screens used in administering the most recent release of Communication Manager running on any of the following:

- Avaya S8XXX Servers
  - DEFINITY servers
  - S8100, S8300, S8400, S8500, or S87XX Servers
- Avaya S8XXX Servers configured as a Local Survivable Processor (LSP).
- Avaya media gateways
  - MCC1 or SCC1 Media Gateways
  - G250, G350, G600, G650, or G700 Media Gateways

Newer releases of Communication Manager contain all the features of prior releases.

Intended audience

This document is intended for system administrators and managers, for users interested in information about specific features, and Avaya personnel responsible for planning, designing, configuring, selling, and supporting the system.

Contents

This document includes the following chapters:

- Chapter 1: System Basics tells you how to log in and log off, set permissions for others who use the administration terminal, set daylight savings rules, set the system date and time, post messages, and back up the information you administer.
- Chapter 2: Planning the System provides information on system-wide functions. It explains how to read, use, and make simple changes to your dial plan, and how to assign feature access codes.
- Chapter 3: Managing Telephones explains how to add, swap, and remove telephones, and how to customize a telephone, for Communication Manager administrators.
- Chapter 4: Managing Telephone Features explains how to administer feature buttons for your users' telephones.
- Chapter 5: Managing Attendant Consoles explains attendant console feature buttons, and tells you how to change, move, or add attendant consoles.
- Chapter 6: Managing Displays provides information on the messages that appear on the read-out screen on display telephones.
- Chapter 7: Handling Incoming Calls shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is unavailable.
- Chapter 8: Routing Outgoing Calls explains how Avaya Communication Manager handles outgoing calls and tells you how to modify call restrictions and your routing plan.
- Chapter 9: Managing Multimedia Calling describes the Multimedia Applications Server Interface (MASI), and provides instructions on administration, monitoring, and troubleshooting. This section also provides information on Multimedia Call Handling (MMCH), which enables users to control voice, video, and data transmissions using a telephone and PC.
- Chapter 10: Setting Up Telecommuting provides information on system-wide settings and individual user administration for telecommuting.
- Chapter 11: Enhancing System Security provides information on analyzing and setting up basic system security, preventing toll fraud, using logins and permissions and passwords, and dealing with security violations.
- Chapter 12: Managing Trunks contains procedures for working with analog and digital trunks. Specialized trunks such as APLT, tandem, release-link, DMI-BOS and ISDN trunk groups are not covered in this manual.
- Chapter 13: Managing Announcements tells you how to record, save, copy, restore and delete announcements.

- Chapter 14: Managing Group Communications shows you how to administer your system so users can page other users or use their telephones as intercoms. You can also give specific users permission to monitor other users' calls or to interrupt active calls with important messages.
- Chapter 15: Managing Data Calls describes the system features available to enable data communications.
- Chapter 16: Administering Avaya Servers describes how to administer an Avaya S87XX Server and an Avaya G700 Media Gateway and the Avaya S8300 Server.
- Chapter 17: Collecting Call Information provides information on account codes, and on tracking and collecting billing information about calls.
- Chapter 18: Telephone Reference describes many of the telephones that you can connect to Communication Manager. It also describes the unique features and buttons for each telephone series to help you administer your user telephones.
- Chapter 19: Screen Reference provides a brief description and a graphic representation of the Communication Manager screens used for administration. It also lists the valid values for fields on the screens, and describes when and why to use each value.

Conventions used

Become familiar with the following terms and conventions. They help you use this book with Communication Manager.

- A "screen" is the display of fields and prompts that appear on a terminal monitor.

  See the Screen Reference chapter for an example of a screen and how it is shown in this book.
- We use the term "telephone" in this book. Other Avaya books might refer to telephones as phones, voice terminals, stations, or endpoints.
- Keys and buttons are printed in a bold font: **Key**.
- Titles of screens are printed in a bold font: **Screen Name**.
- Names of fields are printed in a bold font: **Field Name**.
- Text (other than commands) that you need to type into a field are printed in a bold font: **text**.
- Commands are printed in a bold constant width font: **`command`**.
- Variables are printed in a bold constant width italic font: ***`variable`***.
- We show complete commands in this book, but you can always use an abbreviated version of the command. For example, instead of typing **`list configuration station`**, you can type **`list config sta`**.

- If you need help constructing a command or completing a field, remember to use **Help**.
  - When you press **Help** at any point on the command line, the system displays a list of available commands.
  - When you press **Help** with your cursor in a field on a screen, the system displays a list of valid entries for that field.
- Messages that the system displays are printed in a constant width font: **system message**.
- To move to a certain field on a screen, you can use the **Tab** key, directional arrows, or the **Enter** key on your keyboard.
- If you use terminal emulation software, you need to determine what keys correspond to **Enter**, **Return**, **Cancel**, **Help**, and **Next Page** keys.
- We show commands and screens from the newest release of Communication Manager. If your system has an older version of Communication Manager installed, substitute the appropriate commands for your system and see the manuals you have available.
- The status line or message line can be found near the bottom of your monitor. This is where the system displays messages for you. Check the message line to see how the system responds to your input. Write down the message if you need to call the helpline.
- When a procedure requires you to press **Enter** to save your changes, the screen clears. The cursor returns to the command prompt. The message line shows “**command successfully completed**” to indicate that the system accepted your changes.

Admonishments

Admonishments that might appear in this book have the following meanings:

Note:

Draws attention to information that you must heed.

Tip:

Draws attention to information that you might find helpful.

CAUTION:

Denotes possible harm to software, possible loss of data, or possible service interruptions.

WARNING:

Denotes possible harm to hardware or equipment.

DANGER:

Denotes possible harm or injury to your body.

SECURITY ALERT:
Denotes when system administration might leave your system open to toll fraud.

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Related Resources

The following documents provide additional information.

Administration for Network Connectivity for Avaya Communication Manager, 555-233-504

ATM Installation, Upgrades, and Administration using Avaya Communication Manager, 555-233-124

Avaya Application Solutions -- IP Telephony Deployment Guide, 555-245-600

Avaya Business Advocate User Guide, 07-300653

Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide, 07-600779

Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide, 07-600780

Avaya Communication Manager Advanced Administration Quick Reference, 03-300364

Avaya Communication Manager Basic Administration Quick Reference, 03-300363

Avaya Communication Manager Basic Diagnostics Quick Reference, 03-300365

Avaya Remote Feature Activation (RFA) User Guide, 03-300149

Avaya Toll Fraud and Security Handbook, 555-025-600

Converged Communications Server Installation and Administration, 555-245-705

DEFINITY Communications Systems Generic 2.2 and Generic 3 Version 2 DS1/CEPT1/ISDN PRI Reference, 555-025-107

DEFINITY Enterprise Communications Server Release 1.1 Getting Started with the Avaya R300 Remote Office Communicator, 555-233-769

Feature Description and Implementation for Avaya Communication Manager, 555-245-205

Hardware Description and Reference for Avaya Communication Manager, 555-245-207

Installation, Upgrades and Additions for Avaya CMC1 Media Gateways, 555-233-118

Maintenance Alarms for Avaya Communication Manager, Media Gateways and Servers, 03-300430

Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers, 03-300431

Maintenance Procedures for Avaya Communication Manager, Media Gateways and Servers, 03-300432

Overview for Avaya Communication Manager, 03-300468

Reports for Avaya Communication Manager, 555-233-505

System Capacities Table for Avaya Communication Manager on Avaya Media Servers, 03-300511

Using the Avaya Enterprise Survivable Servers (ESS), 03-300428

What's New in Avaya Communication Manager, 03-300682

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If you need additional help, the following resources are available. You might need to purchase an extended service agreement to use some of these resources. See your Avaya representative for more information.

Go to the Avaya Web site at http://www.avaya.com/support:

- If you are within the United States, click the **Escalation Management** link. Then click the appropriate link for the type of support you need.
- If you are outside the United States, click the **Escalation Management** link. Then click **International Services**, which includes telephone numbers for the international Centers of Excellence. Or contact your local Avaya authorized dealer for any additional help and questions.

Chapter 1: System Basics

Logging into the System

You must log in before you can administer your system. If you are performing remote administration, you must establish a remote administration link and possibly assign the remote administration extension to a hunt group before you log in. The members of this hunt group are the extensions of the data modules available to connect to the system administration terminal. For information about setting up remote administration, contact your Avaya technical support representative. When not using the system, log off for security purposes.

Logging in from a system terminal

This procedure provides instructions for logging in from a system terminal, not a remote terminal.

To log into the system:

1. Enter your login name. Press **Enter**.
2. Enter your password. Press **Enter**.

   For security, your password does not display as you type it.
3. Enter the kind of terminal you have or the type your system emulates. Press **Enter**.

   The Command prompt displays.

   Note:

   If you enter the wrong terminal type, it can lock up your system. If the system is not responding to keyboard commands, type `newterm`. Press **Enter**. Enter the correct terminal type on the new screen. Press **Enter**. If this does not work, turn the power off only on the terminal and then turn it back on. The terminal reboots and you can log in again.

Logging in for remote administration

To log in for remote administration:

1. Dial the Uniform Call Distribution (UCD) group extension number.

   Note:

   The UCD group extension number is assigned when you set up remote administration.

   - If you are off-premises, use the Direct Inward Dialing (DID) number, a Listed Directory Number (LDN) (you must use a telephone), or the trunk number dedicated to remote administration.
   - If you are on-premises, use an extension number.

     If you dialed a DID number, dedicated trunk number, or extension, you receive data tone or visually receive answer confirmation.

     If an LDN was dialed, the attendant will answer.

     a. Ask to be transferred to the UCD group extension number.

        You receive data tone or visually receive answer confirmation.

     b. Transfer the voice call to your data terminal.

   The Login prompt displays.

2. Complete the steps for Logging into the System on page 23.

   For information about setting up remote administration, contact your Avaya technical support representative.

See also Chapter 11: Enhancing System Security. For a complete description of the Security Violation Notification feature, see "Security Violation Notification" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Accessing the Avaya S8XXX Server

To administer an Avaya S8XXX Server, you must be able to access it. Personal computers and services laptop computers equipped with a network PCMCIA card, Avaya Site Administration (ASA), and a Web browser are the primary support access for system initialization, aftermarket additions, and continuing maintenance.

You can access an Avaya S8XXX Server in one of three ways:

- directly
- remotely over the customer's local area network (LAN)
- over a modem

A direct connection and over the customer's LAN are the preferred methods. Remote access over a modem is for Avaya maintenance access only.

Accessing the Avaya S8XXX Server Directly

You can access an Avaya S8XXX Server directly by plugging a computer into the services port which defaults to port 2 (Eth1) on the back of the server. You must use a crossover cable with an RJ45 connector on each end. Plug the other end into the network connector (NIC card) on the your computer. You might need a NIC card adapter.

Once connected, you can administer the server using three tools:

- Web interface for server-specific administration.
- ASA for various features of Avaya Communication Manager
- An SSH client, like PuTTY, and an IP address of 192.11.13.6.

Web Interface - You can access the server Web interface either by connecting directly to the services port on the Avaya S8XXX Server, or by connecting over the customer network.

Connected to the services port - To use the server Web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type **192.11.13.6**. Press **Enter**.
3. When prompted, log in to administer the Avaya S8XXX Server and the features of Avaya Communication Manager.

Connected to the customer network - To use the server Web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type the active server name or IP address. Press **Enter**.
3. When prompted, log in to administer the Avaya S8XXX Server and the features of Avaya Communication Manager.

   You can also connect directly to an individual server using its name or IP address.

Accessing the Avaya S8XXX Server remotely over the network

You can access the Avaya S8XXX Server from any computer connected through the LAN. To access either server, use the IP address assigned to the server you want to access. You can also use the active server address to connect automatically to the server that is active. Once connected, you can administer the server using three tools:

- Web interface for server-specific administration and call processing features
- Avaya Site Administration for Communication Manager (Only available on the active Communication Manager server)
- An SSH client, like PuTTY, and an IP address of 192.11.13.6.

Using Avaya Site Administration

Avaya Site Administration features a graphical user interface (GUI) that provides access to SAT commands as well as wizard-like screens that provide simplified administration for frequently used features. You can perform most of your day-to-day administration tasks from this interface such as adding or removing users and telephony devices. You can also schedule tasks to run at a non-peak usage time. ASA is available in several languages.

The S8300, S8400, or S87XX Server can be used to download Avaya Site Administration. A downloadable version of this package can be accessed through the S8300, S8400, or S87XX Server Web Interface. This software must be installed on a computer running a compatible Microsoft Windows operating system such as Windows 95, 98, NT 4.0, Millennium Edition, Windows 2000, or Windows XP. Once installed, it can be launched from a desktop icon, from the P330 Device Manager, or through a link in the S8300 Server Web Interface.

Installing Avaya Site Administration

If you do not have ASA on your computer, make sure your personal computer (PC) or laptop first meets the following minimum requirements:

Operating systems	Processor/RAM
Windows 2000 Windows XP 2003 (Standard and Enterprise)	Pentium-class 300 MHz/64 MB Pentium-class 300 MHz/64 MB Pentium-class 300 MHz/64 MB
Graphics adapter	SVGA with minimum screen resolution of 800 x 600
Floppy disk drive	3-1/2 in. 1.44-MB floppy disk drive
CD-ROM	CD-ROM drive (required to install ASA from CD)
Available hard disk space	A minimum of 100-MB free hard disk space is required. The requirement for disk space depends on the size and number of configuration data sets.
Printer port	Standard PC or network printer port is required for connecting to a printer to produce station button labels.

Operating systems	Processor/RAM
Network adapter	Required for network access to the S87XX Server, AUDIX, and other network-connected systems.
Free serial ports	One free serial port capable of 9600-bps operation is required for a connection to each serial device (UPS). Avaya recommends that PCs have at least a 16550A UART or 16550A UART simulator (capable of 56 kbps DTE-speed connections). USB and internal modems should emulate this hardware. A second serial port is required for simultaneous connection to AUDIX through a serial connection.

Install ASA on your computer using the Avaya Site Administration CD. Place the ASA CD in the CD-ROM drive and follow the installation instructions in the install wizard.

ASA supports a terminal emulation mode, which is directly equivalent to using SAT commands on a dumb terminal or through an SSH session. ASA also supports a whole range of other features, including the graphically enhanced interface (GEDI) and Data Import. For more information see the Help, Guided Tour, and Show Me accessed from the ASA Help menu.

Starting Avaya Site Administration

To start ASA:

1. Start up ASA by double-clicking the ASA icon, or click **Start>Programs>Avaya Site Administration**.
2. In the **Target System** field, use the pull-down menu to select the desired system.
3. Click **Start GEDI**.

You now are connected to the desired system.

Configuring Avaya Site Administration

When Avaya Site Administration is initially installed on a client machine, it needs to be configured to communicate with Communication Manager on the Avaya S8XXX Server.

When you initially run ASA, you are prompted to create a new entry for the switch connection. You are also prompted to create a new voice mail system if desired.

Adding a Switch Administration Item

To add a switch administration item:

1. Click **File > New > Voice Mail System**.

   The system displays the **Add Voice Mail System** screen.
2. Enter a name in the **Voice Mail System Name** field.
3. Click **Next**. The connection type dialog box displays.
4. Click the **Network connection** radio button.
5. Click **Next**. The Network Connection dialog box displays.
6. Enter the IP address used to connect to the Avaya S8XXX Server.
7. Click **Next**. The Network Connection/Port Number dialog box displays.
8. In the **TCP/IP Port Number** field, enter port **5023**.
9. Click **Next**. The Network Connection/Timeout Parameters dialog box displays. Leave the default values for the timeout parameters.
10. Click **Next**. The login type dialog box displays.
11. Click the **"I want to login manually each time"** radio button.
12. Click **Next**. The Voice Mail System Summary dialog box displays.
13. Check the information, use the **Back** button to make corrections if necessary, and click the **Test** button to test the connection.
14. When the connection is successfully tested, click **Next** and then **Finish**.

   Note:

   In order for ASA to work properly with the ASG Guard II, the **Write (ms)** field on the **Advanced** tab of the **Connection Properties** screen must be set to a value of **5** (i.e., delay of 5 ms). ASG Guard II is an outboard appliance providing access security for Avaya products that do not have Access Security Gateway (ASG) software as a native application. For more information on ASG Guard II, contact your Avaya technical support representative.

Adding a new voice system

When you initially run ASA, you are prompted to create a new entry for the switch connection if you have not already done so. To set up the desired system, click the **New VOICE SYSTEM** icon (on the left of the toolbar row). Complete the **Add Voice System** wizard to build the system record. When completed, the record will be available in the **Target System** pull-down list.

Logging in with Access Security Gateway

Access Security Gateway (ASG) is an authentication interface used to protect the system administration and maintenance ports and logins associated with Avaya Communication Manager. ASG uses a challenge and response protocol to validate the user and reduce unauthorized access.

You can administer ASG authentication on either a port type or login ID. If you set ASG authentication for a specific port, it restricts access to that port for all logins. If you set ASG authentication for a specific login ID, it restricts access to that login, even when the port is not administered to support ASG.

Authentication is successful only when Avaya Communication Manager and the ASG communicate with a compatible key. You must maintain consistency between the Access Security Gateway Key and the secret key assigned to the Communication Manager login. For more information about ASG, see Using Access Security Gateway (ASG) on page 459.

Before you can log into the system with ASG authentication, you need an Access Security Gateway Key, and you need to know your personal identification number (ASG). The Access Security Gateway Key must be pre-programmed with the same secret key (such as, ASG Key, ASG Passkey, or ASG Mobile) assigned to the Avaya Communication Manager login.

Verify that the **Access Security Gateway (ASG)** field on the **System-Parameters Customer Options (Optional Features)** screen is set to **y**. If not, contact your Avaya representative.

Logging in with ASG

To log into the system with ASG:

1. Enter your login ID. Press **Enter**.

   The system displays the challenge number (for example, 555-1234) and system Product ID number (for example, 1000000000). The Product ID provides Avaya Services with the specific identifier of your Avaya MultiVantage communications application.

2. Press **ON** to turn on your Access Security Gateway Key.
3. Type your PIN. Press **ON**.

   The Access Security Gateway Key displays a challenge prompt.

4. At the challenge prompt on the Access Security Gateway Key, type the challenge number without the "-" character (for example, 5551234) from your screen. Press **ON**.

   The Access Security Gateway Key displays a response number (for example, 999-1234).

5. At the response prompt on your terminal, type the ASG response number without the "-" character (for example, 9991234). Press **Enter**.

   The Command prompt displays.

Note:

If you make 3 invalid login attempts, the system terminates the session. For more information, see the appropriate maintenance book for your system.

Login messages

Two messages may be displayed to users at the time of login.

- The Issue of the Day message appears prior to a successful login. In general, use the Issue of the Day to display warnings to users about unauthorized access. The client that is used to access the system can affect when, how, and if the user sees the Issue of the Day message.
- The Message of the Day (MOTD) appears immediately after a user has successfully logged in. In general, use the Message of the Day to inform legitimate users about information such as upcoming outages and impending disk-full conditions.

Using the system default Issue of the Day

The Communication Manager file /etc/issue.avaya contains sample text that may be used for the Issue of the Day message.

To use the system default Issue of the Day:

1. Log into the Communication Manager server and at the CLI enter

   cp /etc/issue.avaya /etc/issue

   cp /etc/issue.avaya /etc/issue.net

Setting Issue of the Day and Message of the Day

For more detailed information on setting login messages and interaction with individual access services, see the See the *Communication Manager Administrator Logins* White Paper on http://support.avaya.com/elmodocs2/white_papers/CM_Administrator_Logins.pdf.

In general, to administer the Issue of the Day and the Message of the Day, use
/bin/vi or
/usr/share/emacs to perform the following edits:

1. Configure **`etc/pam.d/mv-auth`** to include issue PAM module.
2. Edit **`/etc.issue`** and **`/etc.issue.net`** (if using telnet) to include the text for the Issue of the Day.

3. Edit **`etc/motd`** to include the text for the Message of the Day.

Strings not permitted in a Message of the Day (case sensitive)

[513]	Software Version	Login:	incorrect login	SAT cannot be executed on a standby server
513]	Password:	ogin	hallenge	
]	assword	ogin:	SAT	

Logging off the System

For security, log off any time you leave your terminal. If you use terminal emulation software to administer Communication Manager, log off the system and exit the emulation application before switching to another software package.

Instructions

To log off:

1. Type **`logoff`**. Press **Enter**.

   If the Facility Test Call or Remote Access features are administered, Alarm origination is disabled, or if you have busied out resources or active minor or major alarms, a security screen displays. You might want to take appropriate action (for example, disable these features or address any alarms) before you log off.

   If none of the above special circumstances exist, the system logs you off.

Figure 1: Logoff screen

```
Facility Test Call Administered
Remote Access Administered
Alarm Origination is currently disabled
Active major/minor alarm detected; be sure to resolve it

Proceed with Logoff? [n]
```

2. At the **Proceed with Logoff** prompt, type **y** to log off.

   If you log off with alarm origination disabled and the system generates an alarm, Avaya support services will not receive any notification of the alarm. For more information about alarms, see the maintenance book for your system.

Administering User Profiles and Logins

Authentication, Authorization and Accounting (AAA) Services allows you to store and maintain administrator account (login) information on a central server. Login authentication and access authorization is administered on the central server.

For details on administering user profiles and logins, see "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, and *Maintenance Commands for Avaya Communication Manager*, 03-300431.

Establishing Daylight Savings Rules

Avaya Communication Manager allow you to set the daylight savings time rules so that features, such as time-of-day routing and call detail recording (CDR), adjust automatically to daylight savings time. The correct date and time ensure that CDR records are correct. You can set daylight savings time rules to transition to and from daylight savings time outside of normal business hours, so the number of affected CDR records is small.

You can set up 15 customized daylight savings time rules. This allows Communication Manager administrators with servers in several different time zones to set up a rule for each. A daylight savings time rule specifies the exact time when you want to transition to and from daylight savings time. It also specifies the increment at which to transition (for example, 1 hour).

Instructions

Establishing daylight savings rules

To modify a daylight savings rule:

1. Type **`change daylight-savings-rules`**. Press **Enter**.

   The **Daylight Savings Rules** screen appears.

Figure 2: Daylight Savings Rules screen

```
change daylight-savings-rules                                  Page 1 of 2
DAYLIGHT SAVINGS RULES
Rule      Change Day                         Month   Date   Time    Increment

0:   No Daylight Savings

1:   Start: first  __Sunday_ on or after  March__  8_ at 2:00_     __01:00__
     Stop:  first  __Sunday_ on or after November  1_ at 2:00_
2:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
3:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
4:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
5:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
6:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
7:   Start: first  _________ on or after ________ __ at __:__
     Stop:  first  _________ on or after ________ __ at __:__
```

Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after March 8 at 2:00 a.m. with a 01:00 increment. Daylight Savings Time stops on the first Sunday on or after November 1 at 2:00 a.m., also with a 01:00 increment (used as a decrement when switching back to Standard time. This is the default.

The increment is added to standard time at the specified start time and the clock time shifts by that increment (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 03:00).

On the stop date, the increment is subtracted from the specified stop time (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 01:00).

Note:

You cannot delete a daylight savings rule if it is in use on either the **Locations** or **Date and Time** screens. However, you can change any rule except rule 0 (zero).

2. To add a Daylight Savings Time rule, complete the **Start** and **Stop** fields with the day, month, date, and time you want the system clock to transition to Daylight Savings Time and back to standard time.
3. Press **Enter** to save your changes.

Note:

Whenever you change the time of day, the time zone, or daylight savings rules, you must reboot the server for the changes to take effect. See the documentation for your system for information on rebooting the server.

Displaying daylight savings time rules

To display daylight savings time rules:

1. Type **`display daylight-savings-rules`**. Press **Enter**.

   The **Daylight Savings Rules** screen appears. Verify the information you entered is correct.

Setting Time of Day Clock Synchronization

Time of Day Clock Synchronization enables a server to synchronize its internal clock to UTC time provided by Internet time servers. Avaya uses the LINUX platform system clock connected to an Internet time server to provide time synchronization. The interface for these systems is web-based.

LINUX is used in:

- Avaya S8XXX Server IP-PNC
- Avaya S8XXX Server Fiber-PNC
- Avaya S8XXX Server

Before you start:

- A standard TCP/IP LAN connection is required to connect to the Internet time servers. If a LAN connection is not available, time sync will be done by setting the platform clock manually through the command line or web interface.

- On the target server running Communication Manager, verify if Daylight Savings Time is on.

  Note:

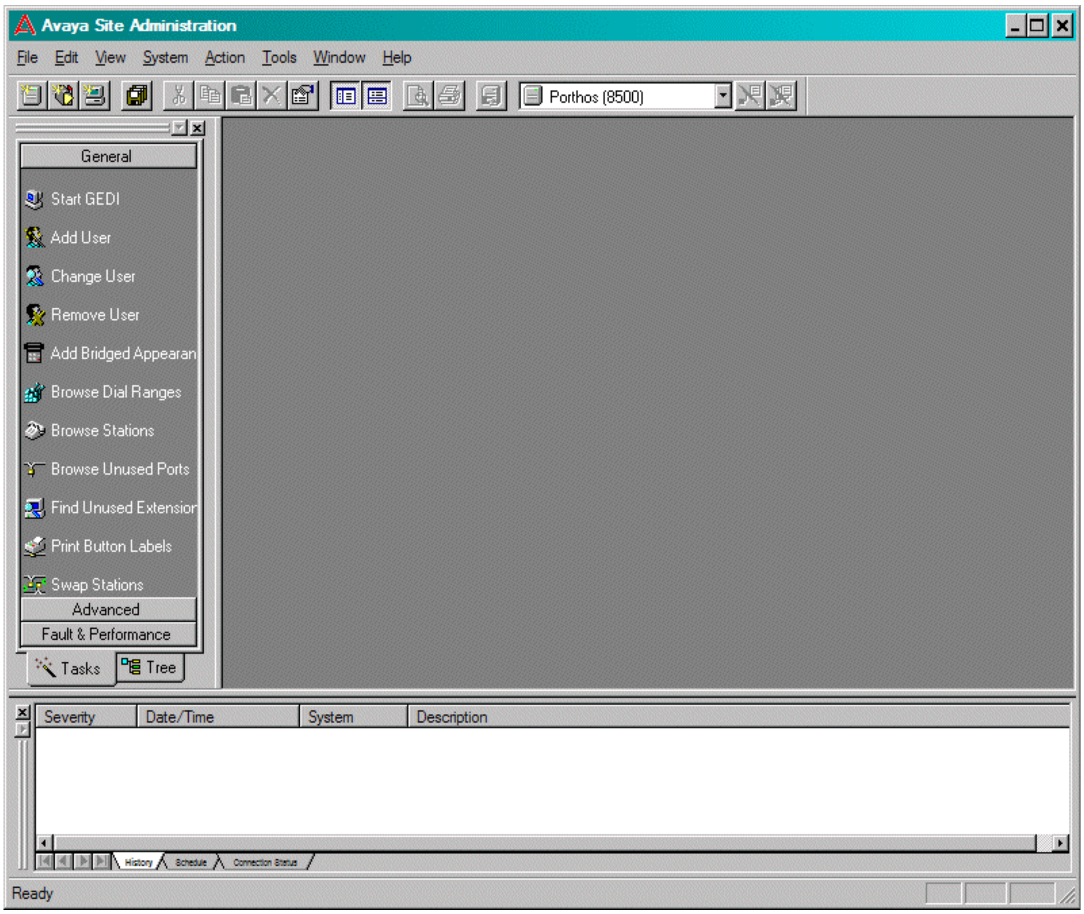
  These instructions are for Avaya DEFINITY CSI servers. For more information, see *Avaya Call Center Automatic Call Distribution (ACD) Guide*, 07-600779.

To set Time of Day Clock Synchronization:

1. Activate the **Avaya Site Administration** screen.
2. Click the **Fault & Performance** tab.

   The **Fault & Performance** icons display.

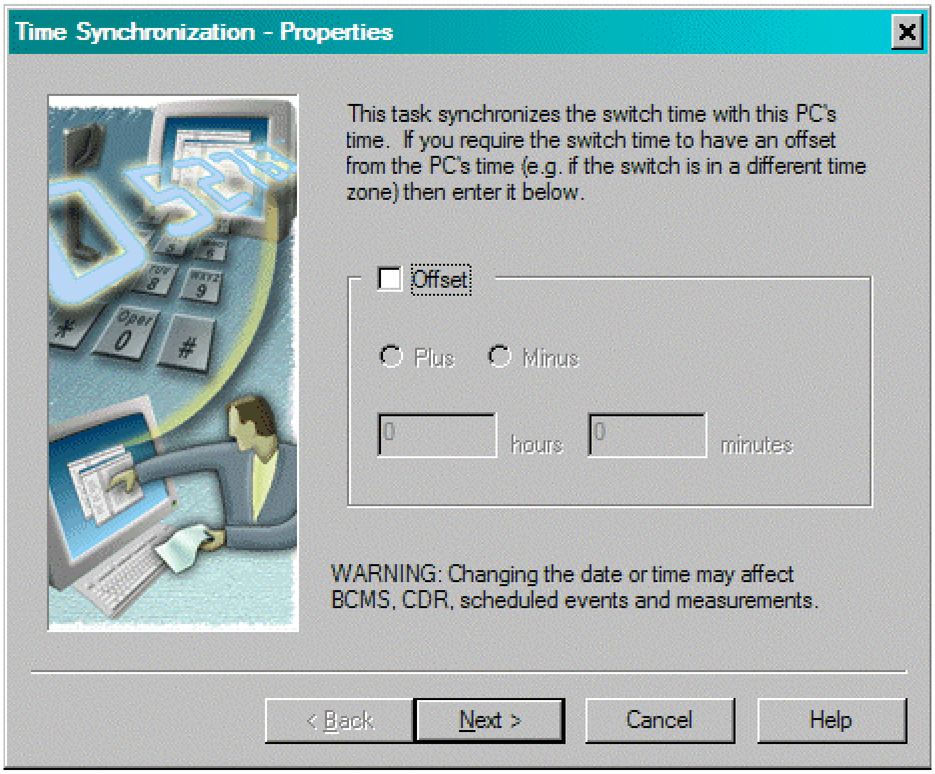
Figure 3: Avaya Site Administration menu



3. Click **Time Synchronization**.

   The **Time Synchronization - Properties** screen displays.

Figure 4: Time Synchronization - Properties screen



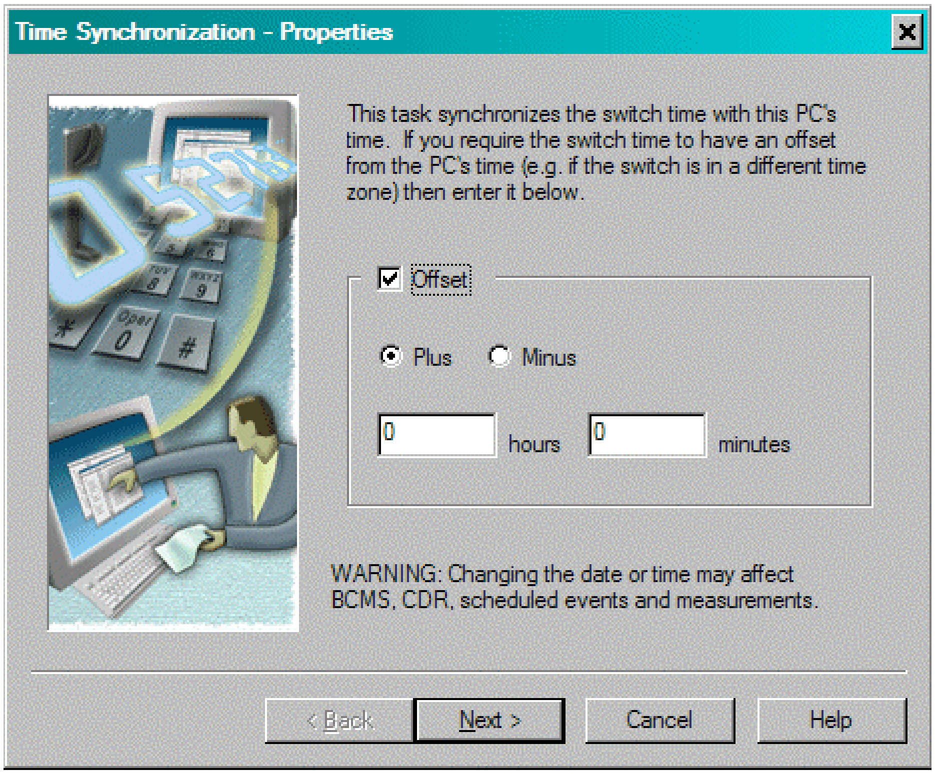
4. Click in the **Offset** box.

   The **Plus** and **Minus** radio buttons and the **Hours and Minutes** fields display.

5. Click **Next**.

   The **Time Synchronization - Properties** screen displays.

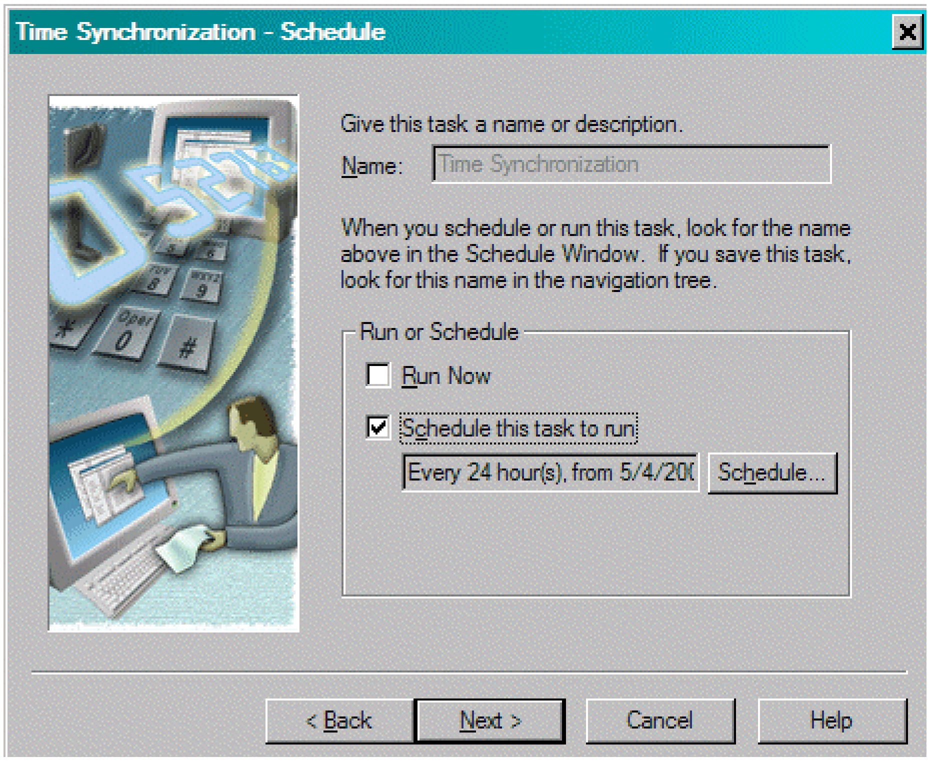
Figure 5: Time Synchronization - Properties screen



6. Click **Plus** to add hours to the remote station (located to the west of the system time) or click Minus to subtract hours to the remote station (located to the east of the system time).
7. In the **hours** field, enter the number of hours to be added or subtracted to synchronize with the remote site.
8. Click **Next**.

   The **Time Synchronization - Schedule** displays.

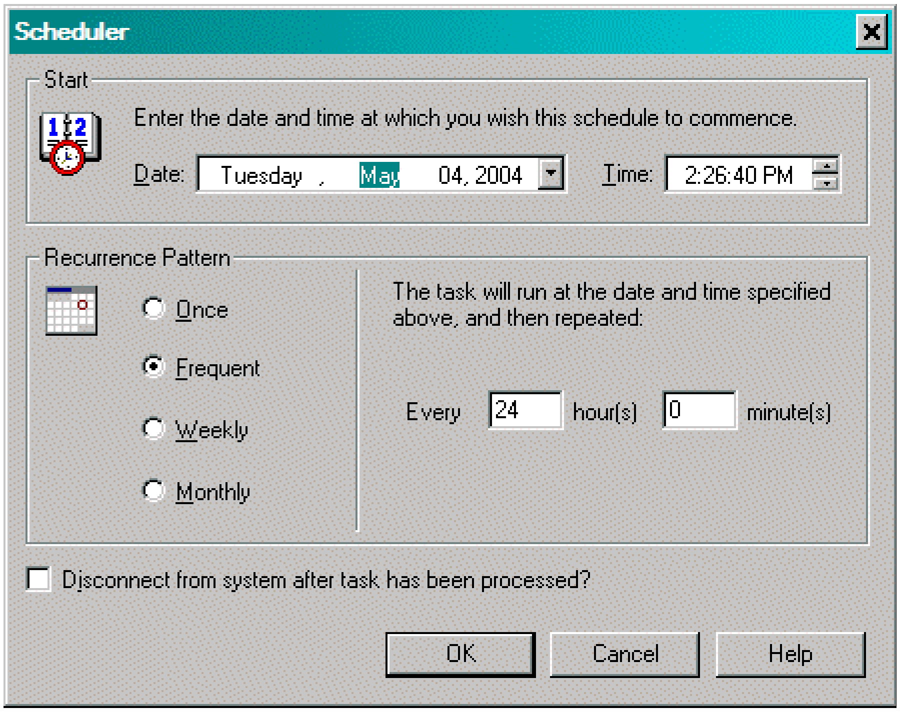
Figure 6: Time Synchronization - Schedule screen



9. Select either:
   - **Run Now** to run this program immediately and click **Next**.
   - **Schedule this task to run** and check the field below to determine if the default setting is satisfactory. If this setting is not satisfactory, click **Schedule**.

   The **Scheduler** screen displays.

Figure 7: Scheduler screen

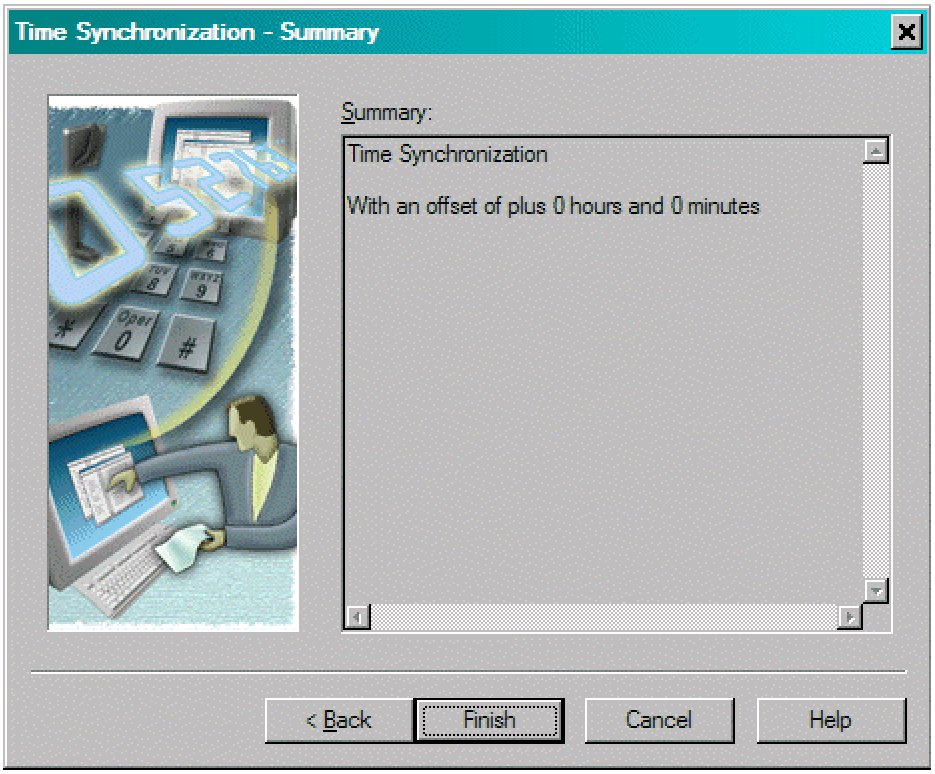


10. In the **Date** field, highlight each object and use the pull-down menu to select the desired setting.
11. In the **Time** field, highlight each item of time and use the pull-down menu to select the desired time.
12. In the **Recurrent Pattern** area, select one radio button for the desired frequency. Notice that each radio button is explained in the area to the immediate right.
13. If there are times to be included, set to the desired frequency.
14. Click **OK**.

    The **Time Synchronization - Schedule** displays.
15. Click **Next**.

    The **Time Synchronization - Summary** screen displays.

Figure 8: Time Synchronization - Summary screen



16. If the time synchronization description is satisfactory, click **Finish**.

    If the time synchronization is not satisfactory, click **Back** and revise the necessary information.

    Note:

    Whenever you change the time of day, the time zone, or daylight savings rules, you must reboot the server for the changes to take effect. See the documentation for your system for information on rebooting the server.

For more information about setting time synchronization, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Setting the System Date and Time

Update the date and time for events such as a leap year, the change to or from daylight savings time, or a system restart after a power failure. The correct date and time assure that Call Detail Recording (CDR) records are correct. CDR does not work until the date and time have been entered.

Note:

Changing the date and time can modify CDR data by 9 hours and 59 minutes. Therefore, you should change the date and time after normal business hours. After you change the date and time, review the time settings for any adjunct (other than AUDIX) linked to your system that uses the system time.

Before you can set the date and time, you need to know whether it is currently daylight savings or standard time and know which daylight savings rule number you want to use. Daylight savings rule numbers are located on the **Daylight Savings Rules** screen.

In our example, we set the date and time to Tuesday, November 5, 2006 at 8:30 p.m. standard time.

To set the system date and time:

1. Type **set time**. Press **Enter**.

   The **Date and Time** screen displays.

Figure 9: Date and Time screen

```
set time

                              DATE AND TIME
DATE
     Day of the Week: _______           Month: ________
    Day of the Month: __                 Year: ____

TIME
    Hour: __ Minute: __       Second: __                      Type:
________________

           Daylight Savings Rule: _

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
         and MEASUREMENTS
```

2. Complete the **Date** fields.

   a. Type **Tuesday** in the **Day of the Week** field.

   b. Type **November** in the **Month** field.

c. Type **5** in the **Day of the Month** field.

d. Type **2006** in the **Year** field.

3. Complete the **Time** fields.

   Use the 24-hour clock to set the hour, so if the current time is 2:00 p.m., you enter **14:00**. You cannot update **Second** — it automatically resets to 0 when you save your changes.

   a. Type **20** in the **Hour** field.

   b. Type **30** in the **Minute** field (8:30 p.m.).

   c. Type **standard** in the **Type** field.

   d. Type **1** in the **Daylight Savings Rule** field.

4. Press **Enter** to save your changes.

   Some display telephones might not automatically refresh the display when you change the date or time. If this occurs, have each user press the date/time button on their telephone to update the display.

   Note:

   Whenever you change the time of day, the time zone, or daylight savings rules, you must reboot the server for the changes to take effect. See the documentation for your system for information on rebooting the server.

Displaying the system date and time

To display the system date and time:

1. Type **`display time`**. Press **Enter**.

   The **Date and Time** screen displays. Verify the information you entered is correct.

Related topics

See Establishing Daylight Savings Rules on page 32 for more information about setting system time.

For additional information, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Using the Bulletin Board

Avaya Communication Manager allows you to post information to a bulletin board. You can also display and print messages from other Avaya server administrators and Avaya personnel using the bulletin board. Anyone with the appropriate permissions can use the bulletin board for messages. Only one user can post or change a message at a time.

Whenever you log in, the system alerts you if you have any messages on the bulletin board and the date of the latest message. Also, if Avaya personnel post high-priority messages while you are logged in, you receive notification the next time you enter a command. This notification disappears after you enter another command and reoccurs at login until deleted by Avaya personnel.

You maintain the bulletin board by deleting messages you have already read. You cannot delete high-priority messages. If the bulletin board is at 80% or more capacity, a message appears at login indicating how much of its capacity is currently used (for example, 84%). If the bulletin board reaches maximum capacity, new messages overwrite the oldest messages.

Note:

The bulletin board does not lose information during a system reset at level 1 or level 2. If you save translations, the information can be restored if a system reset occurs at levels 3, 4, or 5.

Instructions

Displaying messages

To display the bulletin board:

1. Type **`display bulletin-board`**. Press **Enter**.

   The **Bulletin Board** screen displays.

Figure 10: Bulletin Board screen

```
display bulletin-board                                          Page 1 of 3

              Message (* indicates high-priority)                     Date
          *Avaya is in the process of                                03/02/93
          *investigating your trunk lockup problem.                  03/02/93
          *The Bulletin Board will be updated as                     03/02/93
          *we find information.                                      03/02/93
          *  We have identified the problem.                         03/04/93
          *The trunk you added does not provide                      03/04/93
          *disconnect supervision. However, the                      03/04/93
          *trunk group was administered as such.                     03/04/93
          *Please call Pat J. for details.                           03/04/93
          We recently added a new trunk group (14)                   03/02/93
          and have had many of the members getting                   03/02/93
          locked up.                                                 03/02/93
          We see the error - thanks for checking.                    03/05/93
```

Posting a message

In our example, we post a message to the bulletin board about a problem with a new trunk group, and an Avaya representative replies to our message.

To post a message to the bulletin board:

1. Type **`change bulletin-board`**. Press **Enter**.

   The **Bulletin Board** screen displays. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11-19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

2. Type your message.

   You can enter up to 40 characters of text per line. You also can enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The **Tab** key moves the cursor to the next line.

3. Press **Enter** to save your changes.

Deleting messages

To delete a message from the bulletin board:

1. Type **`change bulletin-board`**. Press **Enter**.

   The **Bulletin Board** screen appears.

2. Enter a space as the first character on each line of the message you want to delete. Press **Enter**.

3. Press **Enter** to save your changes.

Saving Translations

Communication Manager retains all translation data in memory while the system is operating. If it goes down, you lose all this translation data. You must save in-memory translation data to the memory card (flash ROM), disk, or tape. Saving translation data to memory card or tape is the same as backing up your system.

Note:

Save translations on a daily basis. You might want to save translations after business hours to prevent dial tone delays or during slow business hours if your business is open 24 hours.

The **`save translation`** command writes two time-stamped identical copies of the translation data to the selected memory card, disk, or tape. The save writes one complete copy first, then writes the second copy in a different area of the device — both with the same time-stamp. Failure during a save, including a system crash, usually affects only one copy. The affected copy is marked "bad" and should not be used for backup.

You can set save translation to be completed automatically as part of regularly scheduled maintenance or manually, as needed. For more information about saving translations automatically, see the maintenance book for your system.

Tip:

To determine if your system saves translations automatically, type **`display system-parameters maintenance`** to see if you have scheduled maintenance.

Translation copy protection assigns a number to a specific telephone system and to the flash card or set of flash cards that belong to that system. For Avaya Communication Manager on a DEFINITY Server CSI, this number is the same on both the translation storage device (flash card) and the Flash PROM (Programmable Read Only Memory) of the processor circuit pack. In a duplicated system, the Flash PROM of each processor circuit pack has a translation ID and both ID's are stored on the memory card.

An attempt to initialize (boot) the system with translations that do not contain the same identification number as stored in the processor circuit pack raises a major alarm and disables access to the **`save translations`** command for all non-Avaya logins. You also receive a warning message on the copyright screen notifying you of the mismatch. Contact your Avaya representative to correct this mismatch and reset the **`save translations`** command. You must correct the mismatch before the end of the specified grace period, otherwise you cannot access system management commands (such as: add, change, remove, and duplicate) that modify translation data. Avaya specifies the grace period during a system installation or following an upgrade.

Before you start

If you are saving translations to a memory card or tape, you must verify the memory card or tape is in place and clear any active alarms from the alarms panel.

If you have Avaya Communication Manager on a DEFINITY Server CSI, verify the memory card translation ID matches the translation ID of your server's Flash PROM.

Instructions

In our example, we save translations to the tapes on both processor A and B.

To save translations manually:

1. Type **`save translation both tape`**. Press **Enter**.

   The save process can take up to 10 minutes. You cannot administer your system while the save is in process. The **Save Translation** screen displays.

Figure 11: Save Translation screen

```
SAVE TRANSLATION


Processor          Command Completion Status               Error Code
SPE_A                                   Success            0
SPE_B                                   Success            0
```

2. If there is an error message in the **Command Completion Status** field and an error code in the **Error Code** field, clear the error and repeat the save process.

More information

When mass storage system (MSS) devices on both processors in a duplex system are specified, translation data is saved from the active processor to the active and standby MSS devices at the same time. If the save to one device fails or one device is out of service, the other save continues. You receive the status of each save separately.

Note:
If you have a duplex system and you save translation data to both MSS devices one at a time, translation data inconsistencies between the two devices can occur.

Fixing problems

Note:
You cannot see whether the translation ID on the flash card corresponds to the number on the Processor circuit packs. However, if the numbers do not match, the system issues an error message when you attempt a save translation operation.

When failures occur, the system responds with the following error codes.

Problem	Possible causes	Solution
1	Save translation cannot write to the active drive.	Repeat the save translation process for the active drive.
2	Save translation cannot write to the standby drive.	Repeat the save translation process for the standby drive.

For more information about error messages and how to correct them, or backing up or restoring your system, see the maintenance book for your system.

Performing Backups

Information on performing backups to your system can be found in the *Maintenance Procedures for Avaya Communication Manager, Media Gateways and Servers*, 03-300432.

Chapter 2: Planning the System

Understanding Your Configuration

At a very basic level, Avaya Communication Manager consists of hardware to perform call processing, and the software to make it run. You use the administration interface to let the system know what hardware you have, where it is located, and what you want the software to do with it.

You can find out which circuit packs are in the system and which ports are available by entering the command **`list configuration all`**. There are variations on this command that display different types of configuration information. Use the help function to experiment, and see which command works for you.

To view a list of port boards on your system:

1. Type **list configuration port-network**. Press **Enter**.

   The **System Configuration** screen appears.

Figure 12: System Configuration screen

```
 SYSTEM CONFIGURATION
Board                                                      Assigned Ports
Number     Board Type Code     Vintage     u=unassigned t=tti p=psa

01A05      DIGITAL LINE          TN754B 000002      01 u  03 u  05 u  07 08
01A06      ANALOG LINE           TN742  000010      01 02 03 04 u  u  u  u
01B05      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C04      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C05      DIGITAL LINE          TN2224 000004      01 u  u  04 u  u  07 08
                                                    u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C06      HYBRID LINE           TN762B 000004      01 02 u  u  u  u  u  u
01C10      DIGITAL LINE          TN754  000004      u  u  u  u  u  u  u  u
```

The **System Configuration** screen shows all the boards on your system that are available for connecting telephones, trunks, data modules and other equipment. You can see the board number, board type, circuit-pack type, and status of each board's ports. The **u** entries on this screen indicate unused ports that are available for you to administer. These might also appear as **p** or **t**, depending on settings in your system.

You will find many places in the administration interface where you are asked to enter a port or slot. The port or slot is actually an address that describes the physical location of the equipment you are using.

A port address is made up of four parts:

- cabinet — the main housing for all the server equipment. Cabinets are numbered starting with 01.
- carrier — the rack within the cabinet that holds a row of circuit packs. Each carrier within a cabinet has a letter, A to E.
- slot — the space in the carrier that holds an individual circuit pack. Slots are numbered 01-16.
- port — the wire that is connected to an individual piece of equipment (such as a telephone or data module). The number of ports on a circuit pack varies depending on the type.

So, if you have a single-carrier cabinet, the circuit pack in slot 06 would have the address 01A06. If you want to attach a telephone to the 3rd port on this board, the port address is 01A0603 (01=cabinet, A=carrier, 06=slot, 03=port).

Understanding the Dial Plan

Your dial plan tells your system how to interpret dialed digits. For example, if you dial 9 on your system to access an outside line, it is actually the dial plan that tells the system to find an external trunk when a dialed string begins with a 9.

The dial plan also tells the system how many digits to expect for certain calls. For example, the dial plan might indicate that all internal extensions are 4-digit numbers that start with 1 or 2.

Let us take a look at an example dial plan so you'll know how to read your system's dial plan. The following figure shows an example of a simple dial plan.

Figure 13: Dial Plan Analysis Table screen

```
display dialplan analysis                                         Page 1 of x
                              DIAL PLAN ANALYSIS TABLE
                                 Location: All                 Percent Full: 7

       Dialed  Total  Call        Dialed  Total  Call         Dialed  Total  Call
       String  Length Type        String  Length Type         String  Length Type
         00       2    attd
         1        3    dac
         2        4    ext
         3        5    ext
         3        1    aar
         4        1    ars
         4        5    ext
         5        5    ext
         5        7    ext
         6        5    ext
         7210     7    ext
         8        7    ext
         9        1    fac
         *        3    fac
         #        3    fac
```

The **Dial Plan Analysis Table** defines the dialing plan for your system.

Note:

In Communication Manager 5.0 and later, you can administer dial plans per-location. Typing the command **`change dialplan analysis`** displays the all-locations **Dial Plan Analysis** screen. To access a per-location screen, type **`change dialplan analysis location`** ***`n`***, where ***`n`*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

The **Call Type** column in the **Dial Plan Analysis Table** indicates what the system does when a user dials the digit or digits indicated in the **Dialed String** column. The **Total Length** column indicates how long the dialed string will be for each type of call. For example, this dial plan shows that when users dial a 5-digit number that starts with 3, they are dialing an extension.

The **Dial Plan Analysis Table** in our example contains the following call types:

- Attendant (attd) — Defines how users call an attendant. Attendant access numbers can be any number from 0 to 9 and contain 1 or 2 digits.

  In our example figure, the system calls an attendant when users dial 00.
- Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. For example, you could define the group 100 to 199, which would allow both FAC and TAC in that range. Dial access codes can start with any number from 1 to 9, * and #, and contain up to 4 digits.

  In our example figure, dial access codes begin with 1 and must be 3 digits long.

  Note:

  The **Dial Plan Analysis Table** does not allow you to enter a range specifically for trunk access codes. However, the **Trunk Group** screen still allows you to assign a TAC to a trunk group. The TAC you enter on the **Trunk Group** screen must match the format you have administered for a DAC on the **Dial Plan Analysis Table**.
- Extensions (ext) — Defines extension ranges that can be used on your system. In our figure, extensions must be in the ranges 30000 to 39999, 40000 to 49999 and 50000 to 59999.
- Feature access codes (fac) only — FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. In our example, feature access codes can begin with * or # and are 3-digits long.

The **Dial Plan Analysis Table** works with the **Dial Plan Parameters Table** for fully defining your dial plan. The **Dial Plan Parameters Table** allows you to set system-wide parameters for your dial plan, or to define a **Dial Plan Parameters Table** per-location.

Displaying your dial plan

You might want to take this opportunity to look at and interpret your own dial plan. To display your system's dial plan:

1. Type **`display dialplan analysis`** or **`display dialplan analysis location`** ***n***, where ***n*** represents the number of a specific location. Press **Enter**.

Modifying your dial plan

It is easy to make changes to your dial plan. For example, we will add a new range of dial access codes to the dial plan. We want to be able to assign both FAC and TAC in the 700 to 799 range.

1. Type **`change dialplan analysis`** or **`change dialplan analysis location`** ***n***, where ***n*** represents the number of a specific location. Press **Enter**.

   The Dial Plan Analysis Table screen appears.
2. Move the cursor to an empty row.
3. Type **7** in the **Dialed String** column. Press **Tab** to move to the next field.
4. Type **3** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **dac** in the **Call Type** column.
6. Press **Enter** to save your changes.

Adding extension ranges

You might find that as your needs grow you want a new set of extensions. Before you can assign a station to an extension, the extension must belong to a range that is defined in the dial plan. We will add a new set of extensions that start with 3 and are 4 digits long (3000 to 3999).

To add this set of extensions to the dial plan:

1. Type **`change dialplan analysis`** or **`change dialplan analysis location`** ***n***, where ***n*** represents the number of a specific location. Press **Enter**.

   The Dial Plan Analysis Table screen appears.
2. Move the cursor to an empty row.
3. Type **3** in the **Dialed String** column. Press **Tab** to move to the next field.
4. Type **4** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **ext** in the **Call Type** column.
6. Press **Enter** to save your changes.

Administering a Uniform Dial Plan

You can set up a Uniform Dialing Plan that can be shared among a group of servers. For more information, see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Administering a Multi-Location Dial Plan

When a customer migrates from a multiple independent node network to a single distributed server whose gateways are distributed across a data network, it might initially appear as if some dial plan functions are no longer available.

The multi-location dial plan feature preserves dial plan uniqueness for extensions and attendants that were provided in a multiple independent node network, but appear to be unavailable when customers migrate to a single distributed server. This feature is available beginning with Communication Manager, release 2.0.

For example, in a department store with many locations, each location might have had its own switch with a multiple independent node network. The same extension could be used to represent a unique department in all stores (extension 123 might be the luggage department). If the customer migrates to a single distributed server, a user could no longer dial 123 to get the luggage department in their store. The user would have to dial the complete extension to connect to the proper department.

Instead of having to dial a complete extension, the multi-location dial plan feature allows a user to dial a shorter version of the extension. For example, a customer can continue to dial 123 instead of having to dial 222-123.

Communication Manager takes leading digits of the location prefix and adds some or all of its leading digits (specified on the **Uniform Dial Plan** screen) to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the **Dial Plan Parameters** and **Dial Plan Analysis** screens.

Note:

Before you can administer the multi-location dial plan feature, the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen must be enabled. To check if this is enabled, use the `display system-parameters customer-options` command. The **Multiple Locations** field is on page 3 of the **System Parameters Customer-Options (Optional Features)** screen. Ensure that the field is set to **y**.

Prepending the location prefix to dialed numbers

Use the **Insert Digits** field on the **Uniform Dial Plan Table** screen to assign the location prefix from the caller's location on the **Locations** screen. The system adds some or all of its leading digits (specified on the **Uniform Dial Plan** screen) to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the **Dial Plan Parameters** screen.

- Non-IP telephones and trunks inherit the location number of the hardware they are connected to (for example, the cabinet, remote office, or media gateway).
- IP telephones indirectly obtain their location number.

- A location number is administered on the **IP Network Region** screen that applies to all telephones in that IP region.
- If a **Location** field is left blank on an **IP Network Region** screen, an IP telephone derives its location from the cabinet where the CLAN board is that the telephone registered through.

- IP trunks obtain their location from the location of its associated signaling group. Either direct administration (only possible for signaling groups for remote offices), or the ways described for IP telephones, determines the location.

Other options for the dial plan

You can establish a dial plan so that users only need to dial one digit to reach another extension. You can also establish a dial plan that allows users to dial, for example, two digits to reach one extension, and three digits to reach another. This is particularly useful in the hospitality industry, where you want users to be able to simply dial a room number to reach another guest.

If you have Communication Manager 5.0 or later, you can administer dial plans per-location. To access a per-location screen, type **`change dialplan analysis location n`**, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Adding feature access codes

As your needs change, you might want to add a new set of FAC for your system. Before you can assign a FAC on the **Feature Access Code (FAC)** screen, it must conform to your dial plan.

In our example, if you want to assign a feature access code of 33 to **Last Number Dialed**, first you need to add a new FAC range to the dial plan.

To add a FAC range from 30 to 39:

1. Type **`change dialplan analysis`** or **`change dialplan analysis location n`**, where ***n*** represents the number of a specific location. Press **Enter**.

   The **Dial Plan Analysis Table** screen appears.
2. Move the cursor to an empty row.
3. Type **3** in the **Dialed String** column. Press **Tab** to move to the next field.
4. Type **2** in the **Total Length** column. Press **Tab** to move to the next field.
5. Type **fac** in the **Call Type** column.
6. Press **Enter** to save your changes.

Changing feature access codes

Feature access codes (FAC) allow users to activate and deactivate features from their telephones. A user who knows the FAC for a feature does not need a programmed button to use the feature. For example, if you tell your users that the FAC for the Last Number Dialed is *33, then users can redial a telephone number by entering the FAC, rather than requiring a Last Number Dialed button.

Many features already have factory-set feature access codes. You can use these default codes or you can change them to codes that make more sense to you. However, every FAC must conform to your dial plan and must be unique. For more information about the dial plan, refer to Understanding the Dial Plan on page 51.

Let us try an example. If you want to change the feature access code for Call Park to *72:

1. Type `change feature-access-codes`. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.
2. Move the cursor to the **Call Park Access Code** field.
3. Type ***72** in the access code field over the old code.
4. Press **Enter** to save your changes.

   If you try to enter a code that is assigned to a feature, the system warns you of the duplicate code and does not allow you to proceed until you change one of them.

Tip:

To remove any feature access code, merely delete the existing FAC and leave the field blank.

Administering Dial Plan Transparency (DPT)

The Dial Plan Transparency (DTP) feature preserves users' dialing patterns when a media gateway registers with a local survivable processor (LSP), or when a Port Network requests service from an Enterprise Survivable Server (ESS). Note that this feature does not provide alternate routing for calls made between Port Networks connected via networks other than IP (e.g., ATM or DS1C), and that register to different ESS servers during a network outage.

Administration of Dial Plan Transparency (DPT) is similar to setting up Inter-Gateway Alternate Routing (IGAR). You must first enable the DPT feature, then set up Network Regions and trunk resources for handling the DPT calls. For ESS servers, you must also assign Port Networks to communities. The following table show the screens and field used in setting up Dial Plan Transparency:

Screen name	Purpose	Fields
Feature-Related System Parameters	• Enable the DPT feature for your system. • Indicate the Class of Restriction to use for the Dial Plan Transparency feature.	• **Enable Dial Plan Transparency in Survivable Mode?** • **COR to use for DPT**
IP Network Region	Administer the DPT feature for Network Regions.	• **Incoming LDN Extension** • **Dial Plan Transparency in Survivable Mode?**
System Parameters-ESS	Enter the community assignments for each Port Network.	**Community**

For more information on the Dial Plan Transparency feature, see "Dial Plan Transparency" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Controlling the features your users can access

Avaya Communication Manager offers a wide range of features and functions. Some of these you can administer differently from one user to the next. For example, you can give one user a certain set of telephone buttons, and the next user a completely different set, depending on what each person needs to get his/her job done. You decide on these things as you administer the telephones for these individuals. Refer to Telephone Feature Buttons Table on page 134 for more information.

You can also establish classes of service (COS) to control the Communication Manager features that users can access. For example, you can permit users to forward their calls, or restrict them from placing priority calls. Once you have set permissions for a class of service, you assign this COS to a user's telephone or other device.

Classes of service are very similar to classes of restriction. COR and COS do not overlap in the access or restrictions they control.

Class of service and class of restriction give you great flexibility with what you allow users to do. If you are in doubt about the potential security risks associated with a particular permission, contact your Avaya technical support representative.

System-wide settings

There are some settings that you enable or disable for the entire system, and these settings effect every user. You might want to look over the various System Parameters screens and decide which settings best meet the needs of your users.

To see a list of the different types of parameters that control your system, type **`display system-parameters`**. Press **Help**. You can change some of these parameters yourself. Type **`change system-parameters`**. Press **Help** to see which types of parameters you can change. In some cases, an Avaya technical support representative is the only person who can make changes, such as to the **System Parameters Customer-Options (Optional Features)** screen.

Type **`list usage`** to see all the instances of an object, such as an extension or ip address, in your system. This is useful when you attempt to change administration and receive an "in use" error. See *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431, for more information.

This chapter offers a few examples of how you establish these system-wide settings. The Screen Reference contains explanations of each of the system parameters screens, and how to use them.

Changing system parameters

You can modify the system parameters that are associated with some of the system features. For example, you can use the system parameters to allow music to play if callers are on hold or to allow trunk-to-trunk transfers on the system.

Generally, Avaya sets your system parameters when your system is installed. However, you can change these parameters as your organization's needs change. For example, let us say that you are told that the number of rings between each point for new coverage paths should change from 4 to 2 rings.

To change the number of rings:

1. Type **`change system-parameters coverage/forwarding`**. Press **Enter**.

   The **System Parameters Call Coverage/Call Forwarding** screen appears.

Figure 14: System-Parameters Call Coverage/Call Forwarding screen

```
 change system-parameters coverage-forwarding                    page 1 of x

         SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
           Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
         Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
                           Coverage - Caller Response Interval (seconds): _
     Threshold for Blocking Off-Net Redirection of Incoming Trunks Calls: 1

COVERAGE
                                         Keep Held SBA at Coverage Point? _
        External Coverage Treatment for Transferred Incoming Trunk Calls? _
         Immediate Redirection on Receipt of PROGRESS Inband Information? _
                                               Maintain SBA At Principal? _
              QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? _
                                            Station Hunt Before Coverage? n

FORWARDING
                                                   Call Forward Override? _
                                               Coverage After Forwarding? _
```

2. In the **Local Coverage Subsequent Redirection/CFWD No Answer Interval** field, type **2.**
3. Press **Enter** to save your changes.

   Each telephone in a Call Coverage path now rings twice before the call routes to the next coverage point. The **Local Cvg Subsequent Redirection/CFWD No Ans Interval** field also controls the number of rings before the call is forwarded when you use Call Forwarding for busy/don't answer calls. This applies only to calls covered or forwarded to local extensions. Use Off-Net to set the number of rings for calls forwarded to public network extensions.

Setting WAN bandwidth limits between network regions

Using the Communication Manager Call Admission Control: Bandwidth Limitation (CAC-BL) feature, you can specify a VOIP bandwidth limit between any pair of IP network regions, and then deny calls that need to be carried over the WAN link that exceed that bandwidth limit. Bandwidth limits can be administered in terms of:

- Kbit/sec WAN facilities
- Mbit/sec WAN facilities
- Explicit number of connections

- No limit

It is highly recommended that you have the following design information before setting the bandwidth limits and mapping the connections:

1. Network topology and WAN link infrastructure.
2. An understanding of the Committed Information Rate (CIR) for the WAN infrastructure.
3. Overlay/design of the Network Regions mapped to the existing topology.
4. Codec sets administered in the system.
5. Bandwidth is assumed to be full duplex.

The following table can be used to help assess how much bandwidth (in Kbits/sec) is used for various types of codecs and packet sizes. The values shown assume a 7 byte L2 WAN header (and are rounded up).

Table 1: Bandwidth usage (in Kbits/sec) based on packet size and codec selection

Packet Size	10 ms	20 ms	30 ms	40 ms	50 ms	60 ms
G.711	102	83	77	74	72	71
G.729	46	27	21	18	16	15
G.723-6.3	NA	NA	19	NA	NA	13
G.723-5.3	NA	NA	18	NA	NA	12

These values, when compared to the actual bandwidth used for 8 byte as well as 10 byte L2 WAN headers are not significantly different. In some cases, the rounded up values shown above are greater than values used for 10 bytes.

The bandwidth usage numbers shown above assume 6 bytes for Multilink Point-to-Point Protocol (MP) or Frame Relay Forum (FRF), 12 Layer 2 (L2) header, and 1 byte for the end-of-frame flag on MP and Frame Relay frames for a total of 7 byte headers only. They do not account for silence suppression or header compression techniques, which might reduce the actual bandwidth. For other types of networks (such as Ethernet or ATM) or for cases where there is a lot of silence suppression or header compression being used, the network might be better modeled by administering the CAC-BL limits in terms of number of connections rather than bandwidth used.

Instructions

Note:

All DIRECT links must be administered first, and INDIRECT links administered last.

To set bandwidth limitations between directly-connected network regions:

1. Type **`change ip-network region <n>`**, where ***n*** is the region number you want to administer. Press **Enter**.
2. The **IP Network Region** screen appears.
3. Scroll to page 3 of the form, **Inter Network Region Connection Management**.
4. In the **codec-set** field, enter the number (1-7) of the codec set to be used between the two regions.
5. In the **Direct WAN** field, enter **y**.
6. In the **WAN-BW-limits** field, enter the number and unit of measure (Calls, Kbits, Mbits, No Limit) that you want to use for bandwidth limitation.
7. Press **Enter** to save your changes.

To set bandwidth limitations between indirectly-connected network regions:

1. Type **`change ip-network region <n>`**, where ***n*** is the region number you want to administer. Press **Enter**.

   The **IP Network Region** screen appears.
2. Scroll to page 3 of the screen, **Inter Network Region Connection Management**.
3. In the **codec-set** field, enter the number (1-7) of the codec set to be used between the two regions.
4. In the **Direct WAN** field, enter **n**.
5. In the **Intervening-regions** fields, enter up to four intervening region numbers between the two indirectly connected regions.
6. In the **Dynamic CAC Gateway** field, set the gateway that reports the bandwidth limit for this link. The gateway must be configured to be a CAC gateway.
7. Press **Enter** to save your changes.

   Note:

   Type **`display ip-network region <n>`** to view the current settings of inter-network region connections. Type **`status ip-network region <n>`** to view the current status (i.e., bandwidth and number of connections being used) of network-region connections.

Administering Treatment for Denied or Invalid Calls

You can administer your system to reroute denied or invalid calls to an announcement, the attendant, or to another extension.

Instructions

In this example, we want:

- all outward restricted call attempts to route to an announcement at extension 2040
- all incoming calls that are denied to route to the attendant
- all invalid dialed numbers to route to an announcement at extension 2045

1. Type **`change system-parameters features`**. Press **Enter**.

   The **Feature-Related System Parameters** screen appears.
2. In the **Controlled Outward Restriction Intercept Treatment** field, type **announcement**.
3. Another blank field appears.
4. In this blank field, type **2040**.
5. This is the extension of an announcement you recorded earlier.
6. In the **DID/Tie/ISDN Intercept Treatment** field, type **attd**.
7. This allows the attendant to handle incoming calls that have been denied.
8. In the **Invalid Number Dialed Intercept** field, type **announcement**.
9. Another blank field appears.
10. In this blank field, type **2045**.
11. This is the extension of an announcement you recorded earlier.
12. Press **Enter** to save your changes.

Setting up Music-on-Hold

Music-on-Hold automatically provides music to a caller placed on hold. Music lets the caller know that the connection is still active. The system does not provide music to callers in a multiple-party connection who are in queue, on hold, or parked.

For more information on locally-sourced Music-on-Hold, see the "Locally Sourced Announcements and Music" feature in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Locally sourced announcements and music

The Locally Sourced Announcements and Music feature is based on the concept of audio source groups. This feature allows announcement and music sources to be located on any or all of the Voice Announcement with LAN (VAL) boards or on virtual VALs (vVAL) in a media gateway. The VAL or vVAL boards are assigned to an audio group. The audio group is then assigned to an announcement or audio extension as a group sourced location. When an incoming call requires an announcement or Music-on-Hold, the audio source that is closest to the incoming call trunk plays.

Storing audio locally minimizes audio distortion because the audio is located within the same port network or gateway as the caller. Therefore, this feature improves the quality of announcements and music on hold. This feature also reduces resource usage, such as VoIP resources, because the nearest available audio source of an announcement or music is played. Locally Sourced Announcements and Music also provides a backup for audio sources because multiple copies of the audio files are stored in multiple locations. Audio sources are assigned either to an audio group or a Music-on-Hold group.

An *audio group* is a collection of identical announcement or music recordings stored on one or more VAL or vVAL boards. The audio group can contain announcements and music. The nearest recording to a call plays for that call.

A *Music-on-Hold (MOH) group* is a collection of externally connected and continuously playing identical music sources. An example of a Music-on-Hold source is a radio station connected to a media gateway using an analog station port. Multiple Music-on-Hold sources can be used in the same system. Like the audio group, the nearest music source to a call plays for that call.

As with the Music-on-Hold feature, only one music source is defined for a system or for a tenant partition. However, you can define a music source as a group of Music-on-Hold sources. Therefore, both non-tenant and tenant systems can use the group concept to distribute Music-on-Hold sources throughout a system.

Adding an audio group

To add an audio group:

1. Type **`add audio-group`** ***`n`***, where ***`n`*** is the group number you want to assign to this audio group, or **`next`** to assign the next available audio group number in the system. Press **Enter**.

The system displays the **Audio Group** screen.

Figure 15: Audio Group screen

```
add audio-group next                                          Page 1 of x
                              Audio Group 2

                            Group Name:

AUDIO SOURCE LOCATION
    1:          16:          31:          46:          61:
    2:          17:          32:          47:          62:
    3:          18:          33:          48:          63:
    4:          19:          34:          49:          64:
    5:          29:          35:          50:          65:
    6:          21:          36:          51:          66:
    7:          22:          37:          52:          67:
    8:          23:          38:          53:          68:
    9:          24:          39:          54:          69:
   10:          25:          40:          55:          70:
   11:          26:          41:          56:          71:
   12:          27:          42:          57:          72:
   13:          28:          43:          58:          73:
   14:          29:          44:          59:          74:
   15:          30:          45:          60:          75:
```

2. In the **Group Name** field, type an identifier name for the group.
3. In the **Audio Source Location** fields, type in the VAL boards or vVAL location designators for each audio source in the audio group.
4. Press **Enter** to save your changes.

Adding a Music-on-Hold group

To add a Music-on-Hold group:

1. Type **`add moh-analog-group n`**, where ***n*** is the Music-on-Hold group number. Press **Enter**.

   The system displays the [MOH Group](#) screen.

Figure 16: MOH Group screen

```
change moh-analog-group 2                                          Page 1 of x
                              MOH Group 2

                            Group Name:

MOH SOURCE LOCATION
     1:         16:         31:         46:         61:
     2:         17:         32:         47:         62:
     3:         18:         33:         48:         63:
     4:         19:         34:         49:         64:
     5:         29:         35:         50:         65:
     6:         21:         36:         51:         66:
     7:         22:         37:         52:         67:
     8:         23:         38:         53:         68:
     9:         24:         39:         54:         69:
    10:         25:         40:         55:         70:
    11:         26:         41:         56:         71:
    12:         27:         42:         57:         72:
    13:         28:         43:         58:         73:
    14:         29:         44:         59:         74:
    15:         30:         45:         60:         75:
```

2. In the **Group Name** field, type in an identifier name for the Music-on-Hold group.
3. In the **MOH Source Location** numbered fields, type in the Music-on-Hold VAL or vVAL source locations.
4. Press **Enter** to save your changes.

Setting system parameters for Music-on-Hold

You must administer the Music-on-Hold (MOH) feature at the system level to allow local callers and incoming trunk callers to hear music while on hold. Note that if your system uses Tenant Partitioning, you cannot set up Music on Hold this way. See Providing MOH Service for Multiple Tenants on page 66 for more information.

To set system parameters for MOH:

1. Type **`change system-parameters features`**. Press **Enter**.

   The **Feature-Related System Parameters** screen appears.

Figure 17: Feature-Related System Parameters screen

```
change system-parameters features                                    page 1 of x

                         FEATURE-RELATED SYSTEM PARAMETERS
                            Self Station Display Enabled? n
                                 Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                   Call Park Timeout Interval (minutes): 10
     Off-Premises Tone Detect Timeout Interval (seconds): 20_
                              AAR/ARS Dial Tone Required? y
                        Music/Tone On Hold: music    Type: _______
           Music (or Silence) On Transferred Trunk Calls: all
                  DID/Tie/ISDN/SIP Intercept Treatment: attd

 Internal Auto-Answer of Attd-Extended/Transferred Calls? y
              Automatic Circuit Assurance (ACA) Enabled? y
                                     ACA Referral Calls: local
                               ACA Referral Destination: ________
           ACA Short Holding Time Originating Extension: ______
            ACA Long Holding Time Originating Extension: ______
          Abbreviated Dial Programming by Assigned Lists:
     Auto Abbreviated/Delayed Transition Interval(rings):
                Protocol for Caller ID Analog Terminals: Bellcore
 Display Calling Number for Room to Room Caller ID Calls?
```

2. In the **Music/Tone On Hold** field, type **music**.

   The **Type** field appears.

3. In the **Type** field, enter the type of music source you want to utilize for MOH: an extension (ext), an audio group (group), or a port on a circuit pack (port).
4. In the text field that appears to the right of your **Type** selection, type the extension number, the audio group, or the port address of the music source.
5. In the **Music (or Silence) on Transferred Trunk Calls**, type **all**.
6. Press **Enter** to save your changes.
7. Now administer a class of restriction with **Hear System Music on Hold** set to **y**, to allow your local users to hear Music-on-Hold.

Providing MOH Service for Multiple Tenants

If you manage the switching system for an entire office building, you might need to provide individualized telephone service for each of the firms who are tenants. You can set up your system so that each tenant can have its own attendant, and can chose to have music or play special announcements while callers are on hold.

Before you can administer tenants in your system, **Tenant Partitioning** must be set to **y** on the System Parameters Customer-Options (Optional Features) screen. This setting is controlled by your license file.

The following example illustrates how to administer the system to allow one tenant to play Country music for callers on hold, and another to play Classical music.

1. Type **`change music-sources`**. Press **Enter**.

   The Music Sources screen appears.

Figure 18: Music Sources screen

```
change music-source           Music Sources                        Page 1 of X

         Source No      Type       Source               Description

                1       music      Type: ext 30002    music-on-extension
                2       music      Type: group 10     music-on-group
                3       music      Type: a0904        music-on-part
                4       tone                          tone-on-hold
                5       none
                6       none
                7       none
                8       none
                9       none
               10       none
               11       none
               12       none
               13       none
               14       none
               15       none
```

2. For **Source No 1**, enter **music** in the **Type** column. A **Type** field appears under the **Source** column.
3. In the **Type** field, enter **port**. A blank text field appears.
4. Enter the port number, **01A1001** in this case, in the text field.
5. In the description field, enter **Country**.
6. Move to **Source 3**, and enter **music** in the **Type** column, port in the **Type** field, **01A1003** for the port number, and **Classical** for the **Description**.
7. Press **Enter** to save your changes.
8. Type **`change tenant 1`**. Press **Enter**.

   The Tenant screen appears.

Figure 19: Tenant screen

```
change tenant 1
                                Tenant 1

                 Tenant Description: ________________________________________
                    Attendant Group: 1
              Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____
                  Night Destination: _____
                       Music Source: 1
            Attendant Vectoring VDN:
```

9. In the **Tenant Description** field, type **Dentist**.

   This identifies the client in this partition.

10. In the **Attendant Group** field, type **1**.

    Note:
    The attendant group number must also appear in the **Group** field of the **Attendant Console** screen for this tenant.

11. In the **Music Source** field, type **1**.

    Callers to this tenant will now hear country music while on hold.

12. Press **Enter** to save your changes.
13. To administer the next partition, type **change tenant 2**. Press **Enter**.
14. Administer this tenant, Insurance Agent, to use Attendant Group 2 and Music Source 3. Be sure to change the **Attendant Console** screen so that this attendant is in group 2.

    This tenant's callers will hear classical music on hold.

More MOH information

If a call with either Data Privacy or Data Restriction activated is placed on hold, the Music/Tone on Hold is withheld. This is to prevent transmission of a musical tone that a connected data service might falsely interpret as a data transmission.

If you administer the **Music/Tone on Hold** field to provide music, the system provides the music after a hunt group or Automatic Call Distribution (ACD) split delayed announcement.

Music on Hold might sound distorted when played to IP trunks or to IP telephones through certain codecs, particularly the G.723 codec. You can provide different on-hold materials for these endpoints. Using the instructions for Providing MOH Service for Multiple Tenants, create one tenant partition for all endpoints that do not use the G.723 codec and administer Music on Hold for this tenant. Create another tenant partition for endpoints that use the G.723 codec and administer silence, news, or other material that does not sound distorted for these endpoints.

Receiving Notification in an Emergency

If one of your users calls an emergency service such as the police or ambulance, someone, perhaps the receptionist, security or the front desk, needs to know who made the call. Thus, when the emergency personnel arrive, they can be directed to the right place.

You can set up Communication Manager to alert the attendant and up to ten other extensions whenever an end-user dials an emergency number. The display on the notified user's telephone shows the name and number of the person who placed the emergency call. The telephones also ring with a siren-type alarm, which users must acknowledge to cancel.

Before you start

Decide if you want one user to be able to acknowledge an alert, or if all users must respond before an alert is cancelled.

Verify that the **ARS** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Make sure that the extensions you notify belong to physical digital display telephones. Refer to Telephone Reference on page 653 for a list of telephone types. When you assign crisis alert buttons to the telephones, check the **Type** field on the **Station** screen to be sure you are not using a virtual extension.

Instructions

In this example, we will set up the system to notify the attendant and the security guards at all 3 entrances when someone dials the emergency number 5555. All three guards must acknowledge the alert before it is silent.

1. Type `change ars analysis n`. Press **Enter**.

   The **ARS Digit Analysis Table** screen appears.

Figure 20: ARS Digit Analysis Table screen

```
change ars analysis                                          Page 1 of X

                          ARS DIGIT ANALYSIS TABLE
                                                    Percent Full:  ___

    Dialed          Total    Route     Call    Node   ANI
    String         Min Max   Pattern   Type    Num    Reqd
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
_________________  __  __    _____     ____    ___     n
```

2. In the **Dialed String** field, type **5555**.

   This is the number that end-users dial to reach emergency services.

3. In the **Total Min** and **Max** fields, type **4**.

   In this example, the user must dial all 4 digits for the call to be treated as an emergency call.

4. In the **Route Pattern** field, type **1**.

   In this example, we use route pattern 1 for local calls.

5. In the **Call Type** field, type **alrt**.

   This identifies the dialed string 5555 as one that activates emergency notification.

6. Press **Enter** to save your changes.

Now set up the attendant console to receive emergency notification.

7. Type **change attendant 1**. Press **Enter**.

   The **Attendant Console** screen appears.

8. In the feature button area, assign a **crss-alert** button.

9. Press **Enter** to save your changes.

10. Assign a **crss-alert** button to each security guard's telephone.

    You cannot assign this button to a soft key. See Adding Feature Buttons on page 129 for more information.

Finally, we make sure that all security personnel and the attendant will have to acknowledge the alert.

11. Type **`change system-parameters crisis-alert`**. Press **Enter**.

    The **Crisis Alert System Parameters** screen appears.

12. Go to the **Every User Responds** field and type **y**.
13. Press **Enter** to save your changes.

More information

Attendants cancel an alert by pressing the crisis alert button three times. The first button push turns off the siren, the second stops the lamp from flashing, and the third clears the display.

Digital telephone users cancel the siren by pushing the crisis alert button. If you have set the system so that only one user needs to respond, this stops the alerting at all telephones. If all users must respond, each telephone continues to alert until that user presses the crisis alert button to acknowledge the alarm. The emergency caller's name and extension remain on the display at this point. To completely cancel an alert and clear their displays, users press the Normal button.

Once you administer Crisis Alert, Communication Manager still records each emergency call and sends a record to the journal printer, if available. If not, you can view the emergency log with the command **`list emergency`**.

Related topics

To determine what types of digital telephones have displays, see Telephone Reference on page 653.

For information on setting up Centralized Automatic Message Accounting (CAMA) trunks and providing Caller's Emergency Service Identification (CESID) information to your local community's Enhanced 911 system, see CAMA Numbering Format.

For information about updating station location information without having to change the USA 911 system's Automatic Location Identification database, see the **Emergency Location extension** field in Station on page 1491.

For information on how to administer IP telephones to make emergency calls, see Setting up emergency calls on IP telephones on page 114.

For more information on individual features, see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Notifying a Digital Pager of an Emergency

You also have the option of having your emergency calls go to a digital pager. When someone dials an emergency number (for example, 911), the system sends the extension and location (that originated the emergency call) to the administered pager.

Before you start

- You need to administer a **crss-alert** button on at least one of the following.
  - Attendant Console (use the `change attendant` command)
  - Digital telephone set (use the `change station` command)
- The **ARS Digit Analysis Table** must have emergency numbers in the **Call Type** column set to **alrt** (crisis alert).
- You need a digital numeric pager.

Instructions

To set up crisis alert to a digital pager:

1. Type `change system-parameters crisis-alert`. Press **Enter**.

   The Crisis Alert System Parameters screen appears.
2. In the **Alert Pager** field, type **y**.

   This allows you to use the Crisis Alert to a Digital Pager feature and causes additional crisis alert administration fields to appear.
3. In the **Originating Extension** field, type a valid unused extension to send the crisis alert message.

   We'll type **7768**.
4. In the **Crisis Alert Code** field, type **911**.

   This is the number used to call the crisis alert pager.
5. In the **Retries** field, type **5**.

   This is the number of additional times the system tries to send out the alert message in case of an unsuccessful attempt.
6. In the **Retry Interval (sec)** field, type **30**.

   This is length of time between retries.

7. In the **Main Number** field, type the number that is to be displayed at the end of the pager message.

   We'll type **303-555-0800**.

8. In the **Pager Number** field, type the number for the pager.

   We'll type **303-555-9001**.

9. In the **Pin Number** field, type **pp77614567890**.

   This is the PIN number, if required, for the pager. Insert any pause digits (pp) as needed to wait for announcements from the pager service to complete before sending the PIN.

10. In the **DTMF Duration - Tone (msec)** field, type **100**.

    This is the length of time the DTMF tone is heard for each digit.

11. In the **Pause (msec)** field, type **100**.

    This is the length of time between DTMF tones for each digit.

12. Press **Enter** to save your changes.

Related topics

Refer to the Crisis Alert feature in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more detailed information.

Other Useful Settings

There are many settings that control how your system operates and how your users telephones work. Most of these you administer through one of the System Parameters screens. This section describes a few of the items you can enable in your system to help your users work more efficiently. See Feature-Related System Parameters for a more detailed description of the available system settings.

Automatic callback if an extension is busy

You can allow users to request that the system call them back if they call a user whose telephone is busy. For more information, see "Automatic Callback" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Automatic hold

You can set a system-wide parameter that allows your users to initiate a call on a second line without putting the first call on Hold. This is called Automatic Hold, and you enable it on the **Feature-Related System Parameters** screen. If you do not turn this on, the active call drops when a the user presses the second line button.

Bridging onto a call that has gone to coverage

You can allow users to join (bridge) on to a call that rang at their extension and then went to coverage before they could answer. For more information, see "Temporary Bridged Appearance" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Distinctive ringing

You can establish different ringing patterns for different types of calls. For example, you can administer your system so that internal calls ring differently from external calls or priority calls. For more information, see "Distinctive Ringing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Warning when telephones are off-hook

You can administer the system so that if a telephone remains off-hook for a given length of time, Communication Manager sends out a warning. This is particularly useful in hospitals, where the telephone being off-hook might be an indication of trouble with a patient. See **Class of Service** for more information.

Warning users if their calls are redirected

You can warn analog telephone users if they have features active that might redirect calls. For example, if the user has activated send all calls or call forwarding, you can administer the system to play a special dial tone when the user goes off-hook. See "Distinctive Ringing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Controlling the Calls Your Users Can Make and Receive

The Avaya Communication Manager provides several ways for you to restrict the types of calls your users can make, and the features that they can access.

You use class of restriction (COR) to define the types of calls your users can place and receive. Your system might have only a single COR, a COR with no restrictions, or as many CORs as necessary to effect the desired restrictions.

You will see the **COR** field in many different places throughout Communication Manager when administering telephones, trunks, agent logins, and data modules, to name a few. You must enter a COR on these screens, although you control the level of restriction the COR provides.

Strategies for assigning CORs

The best strategy is to make it as simple as possible for you and your staff to know which COR to assign when administering your system. You can create a unique COR for each type of user or facility, for example, call center agents, account executives, administrative assistants, Wide Area Telecommunications Service (WATS) trunks, paging zones or data modules.

You can also create a unique COR for each type of restriction, for example, toll restriction, or outward restriction. If you have a number of people who help you administer your system, using this method would also require the additional step of explaining where you wanted to use each type of restriction.

Note:

COR-to-COR calling restrictions from a station to a trunk do not apply when Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), or Uniform Dial Plan (UDP) is used to place the call. In these cases, use Facility Restriction Levels to block groups of users from accessing specific trunk groups.See "Class of Restriction" and "Facility Restriction Levels" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

To find out what CORs are administered in your system already, type **`list cor`**. You can also display information for a single COR by typing **`list cor #`**.

Allowing users to change CORs

You can allow specific users to change their Class of Restriction (COR) from their telephones using a Change COR feature access code. You can also limit this feature by insisting that the user enter a password as well as a feature access code before they can change their COR. The Station Lock feature also allows users to change their own COR. For more information on the Station Lock feature, see Station Lock on page 78.

Before you start:

- Be sure that **Change COR by FAC** field is set to **y** on the **System-Parameters Customer-Options (Optional Features)** screen. Note that you cannot have both **Change COR by FAC** and **Tenant Partitioning** enabled.
- Be sure that each user (who you want to allow to change a COR) has a class of service with console permissions. For more information about console permissions, refer to Class of Service on page 852.

To allow users to change their own class of restriction, you must define a feature access code and can, optionally, create a password. For example, we will create a change COR feature access code of *55 and a password of 12344321.

1. Type `change feature-access-codes`. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.
2. Move the cursor to the **Change COR Access Code** field.
3. Type ***55** in the access code field.
4. Press **Enter** to save your changes.

   Now we have to define the password.
5. Type `change system-parameters features`. Press **Enter**.

   The **Feature-Related System Parameters** screen appears. Press **Next Page** to find the **Automatic Exclusion Parameters** section.

Figure 21: Feature-Related System Parameters screen

```
change system-parameters features                              Page 16 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                Automatic Exclusion Coverage/Hold? y
            Automatic Exclusion with Whisper Page? y
                              Recall Rotary Digit: 2
                    Password to Change COR by FAC: *
         Duration of Call Timer Display (seconds): 3

WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

IP PARAMETERS
                   Direct IP-IP Audio Connections? n
                             IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                           Re-try?
     T2 (Backward Signal) Activation Timer (secs):
```

6. Move to the **Password to Change COR by FAC** field and enter **12344321**.

   This field determines whether or not Communication Manager requires the user to enter a password when they try to change their COR. Avaya recommends that you require a password.

7. Press **Enter** to save your changes.

Station Lock

Station Lock provides users with the capability to manually lock their stations, using a button or feature access code, in order to prevent unauthorized external calls from being placed.

Station Lock can prevent unauthorized external calls. Telephones can be remotely locked and unlocked. Station Lock allows users to:

- change their Class of Restriction (COR); usually the lock COR is set to fewer calling permissions than the station's usual COR
- lock their telephones to prevent unauthorized outgoing calls.
- block outgoing calls and still receive incoming calls.
- block all outgoing calls except for emergency calls.

Station Lock is activated by pressing a telephone button, which lights the button indicator, or dialing a FAC.

Analog and XMOBILE stations must dial a FAC to activate the feature. The user hears a special dial tone on subsequent origination attempts from the telephone to indicate that the lock feature is active.

Digital stations (including DCP, BRI, IP hardphones and softphones) access Station Lock with a feature button or via a FAC. If a digital station has a Station Lock button but activates the feature with the FAC, the LED for the button lights and no special dial tone is provided. However, if a digital station does not have a Station Lock button and activates the feature with the FAC, a special dial tone is provided.

A station can be locked or unlocked from any other station if the FAC is used and the Station Security Code is known. The attendant console can never be locked but can be used to lock or unlock other stations. A station also can be locked or unlocked via a remote access trunk.

Interactions

- Attendant Console

  Station Lock cannot be used for attendant consoles but it can be assigned to regular digital stations that might also have console permissions. The FAC cannot be used to activate Station Lock for the attendant console, but the FAC can be dialed from the attendant console in an attempt to remotely activate or deactivate Station Lock for another station.

- Personal Station Access (PSA)

  Station Lock can be used for PSA stations as long as they are associated with an extension. When stations are disassociated, Station Lock cannot be activated.

- Remote Access

  After a remote user dials a valid barrier code, the user receives system dial tone. To activate/deactivate Station Lock, the user must dial the FAC, then the extension number, then the security code number.

Station Lock by time of day

Beginning with Communication Manager 4.0 or later, you can you can also lock stations using a Time of Day (TOD) schedule.

To engage the TOD station lock/unlock you do not have to dial the station lock/unlock FAC, or use **stn-lock** button push.

When the TOD feature activates the automatic station lock, the station uses the Class of Restriction (COR) assigned to the station lock feature for call processing. The COR used is the same as it is for manual station locks.

The TOD lock/unlock feature does not update displays automatically, because the system would have to scan through all stations to find the ones to update.

The TOD Station Lock feature works as follows:

- If the station is equipped with a display, the display will show “Time of Day Station Locked”, if the station invokes a transaction which is denied by the Station Lock COR.
  Whenever the station is within a TOD Lock interval, the user will hear a special dial tone instead of the normal dial tone, if the special dial tone is administered.
- For analog stations or without a display, the user hears a special dial tone. The special dial tone has to be administered and the user hears it when the station is off hook.

After a station is locked by TOD, it can be unlocked from any other station if the Feature Access Code (FAC) or button is used. You have to also know the Station Security Code, and that the **Manual-unlock allowed?** field on the **Time of Day Station Lock Table** screen is set to **y**.

Once a station has been unlocked during a TOD lock interval, the station remains unlocked until next station lock interval becomes effective.

If the station was locked by TOD and by Manual Lock, an unlock procedure will unlock the Manual Lock as well as the TOD Lock (“Manual-unlock allowed?” field on the **Time of Day Station Lock Table** screen is set to **y**).

The TOD feature does not unlock a manually locked station.

Note:

The attendant console cannot be locked by TOD or manual station lock.

Screens for administering Station Lock

Screen name	Purpose	Fields
COR	Administer a Class of Restriction (COR) that allows the user to activate Station Lock with a feature access code (FAC).	**Station Lock COR**
Feature Access Code (FAC)	Assign one FAC for Station Lock activation, and another FAC for Station Lock Deactivation.	**Station Lock Activation** **Station Lock Deactivation**
Station	Assign the user a COR that allows the user to activate Station Lock with an FAC.	**COR** **Time of Day Lock Table**
	Assign a **sta-lock** feature button for a user.	Any available button field in the **BUTTON ASSIGNMENTS** area
	Assign a Station Security Code (SSC) for a user.	**Security Code**
Time of Day Station Lock Table	Administer station lock by time of day.	**Table Active?** **Manual Unlock Allowed?** **Time Intervals**

Chapter 3: Managing Telephones

Installing New Telephones

Simple administration allows you to plug a telephone into a jack and dial a sequence to start up service to the telephone. The dialing sequence sets up an association between the telephone and the corresponding station administration.

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use might cause you security problems. Consult the Avaya Products Security Handbook for suggestions on how to secure your system and find out about obtaining additional security information.

For traditional instructions, see Installing New Telephones.

Before you start

On the **Feature-Related System Parameters** screen, be sure the **Customer Telephone Activation (CTA) Enabled** field is **y** and the **TTI Enabled** field is **y**.

Complete the **Station** screen for the new telephone and type x in the port field. Note that the telephone type must match the board type. For example, match a two-wire digital telephone with a port on a two-wire digital circuit pack. Use this procedure with all circuit-switched telephones except BRI (ISDN) and model 7103A.

Tip:

See Completing the Station screens for more information. See Duplicating telephones if you want to add a number of telephones with similar settings.

Instructions

CAUTION:

You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To associate a telephone with existing x-port station administration, complete the following steps from the telephone you want to install:

1. Plug the telephone into the wall jack.
2. Lift the receiver and continue if you hear dial tone.
3. Dial **#*`nnnn`**, where ***`nnnn`*** is the extension number of the telephone you are installing.
4. Hang up after you receive confirmation tone.
5. Dial a test call to confirm that the telephone is in service.

   If possible, call a telephone with a display so the person answering can confirm that you entered the correct extension number.
6. Repeat the process until all new telephones have been installed.
7. For security reasons, you should disable this feature when you are done. At the system administration terminal type **`change system-parameters features`** to access the **Feature-Related System Parameters** screen.
8. Type **n** in the **Customer Telephone Activation (CTA) Enabled** field.
9. Press **Enter** to save your changes.
10. Type **`save translations`**. Press **Enter** to permanently save the changes.
    a. Fixing problems

       If you misdial and the wrong extension is activated for the telephone you are using, use the terminal translation initialization (TTI) unmerge feature access code to "uninstall" the telephone before you try again.

Adding new telephones

When you are asked to add a new telephone to the system, what do you do first? To connect a new telephone you need to do three things:

- find an available port
- wire the port to the cross-connect field or termination closet
- tell the telephone system what you're doing

Before you can determine which port to use for the new telephone, you need to determine what type of telephone you are installing, what ports are available, and where you want to install the telephone.

To add an IP telephone, see Adding an IP telephone on page 110.

Gathering necessary information

Gather the following information:

1. Determine whether the telephone is an analog, digital, ISDN, or hybrid set. You can also administer a virtual telephone, one without hardware at the time of administration.

   You need this information to determine the type of port you need, because the port type and telephone type must match. If you do not know what type of telephone you have, see the **Type** field on the **Station** screen for a list of telephones by model number.

2. Record the room location, jack number, and wire number.

   You might find this information on the jack where you want to install the telephone, recorded in your system records, or from the technician responsible for the physical installation.

3. Display the available boards (cards) and ports.

   To view a list of boards on your system, type `list configuration station`. Press **Enter**.

   The **System Configuration** screen appears.

Figure 22: System Configuration screen

```
                           SYSTEM CONFIGURATION
Board                                                Assigned Ports
Number     Board Type       Code    Vintage u=unassigned t=tti p=psa

01A05     DIGITAL LINE       TN754B 000002  01 u  03 u  05 u  07 08
01A06     ANALOG LINE        TN742  000010  01 02 03 04 u  u  u  u
01B05     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C04     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C05     DIGITAL LINE       TN2224 000004  01 u  u  04 u  u  07 08
                                            u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C06     HYBRID LINE        TN762B 000004  01 02 u  u  u  u  u  u
01C10     DIGITAL LINE       TN754  000004  u  u  u  u  u  u  u  u
```

The **System Configuration** screen shows all the boards on your system that are available for connecting telephones. You can see the board number, board type, circuit-pack type, and status of each board's ports.

4. Choose an available port and record its port address.

   Each port that is available or unassigned is indicated by a '`u`.' Choose an available port from a board type that matches your telephone type (such as a port on an analog board for an analog telephone).

   Every telephone must have a valid port assignment, also called a port address. The combined board number and port number is the port address. So, if you want to attach a telephone to the 3rd port on the 01C05 board, the port address is 01C0503 (01=cabinet, C=carrier, 05=slot, 03=port).

   Tip:

   If you add several telephones at one time, you might want to print a paper copy of the **System Configuration** screen. To print the screen to a printer attached to the system terminal, type **`list configuration station print`**. Press **Enter**. To print to the system printer that you use for scheduled reports, type **`list configuration station schedule immediate`**. Press **Enter**.

5. Choose an extension number for the new telephone.

   The extension you choose must not be assigned and must conform to your dial plan. You should also determine whether this user needs an extension that can be directly dialed (DID) or reached via a central telephone number.

   Be sure to note your port and extension selections on your system's paper records.

Physically connecting the telephone

Once you have collected all the information, you are ready to physically wire the port to the cross-connect field.

If you have an Avaya technical support representative or on-site technician who completes the physical connections, you need to notify them that you are ready to add the telephone to the system. To request that Avaya install the new connections, call your Avaya technical support representative to place an order.

If you are responsible for making the connections yourself and if you have any questions about connecting the port to the cross-connect field, see your system installation guide.

Now you are ready to configure the system so that it recognizes the new telephone.

Before you start

To download language display labels for telephones, set the **Display Language** field on the **Station** screen to **english**, **spanish**, **italian**, **french**, **user-defined**, or **unicode**.

Note:

Unicode display is only available for Unicode-supported telephones. Currently, the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones support Unicode display. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

For a Eurofont character display for the 2420/2410 telephone, set the **Display Character Set** field on the **System-Parameters Country-Options** screen to **Eurofont**.

For a Katakana character display for the 2420/2410 telephone, set the **Display Character Set** field on the **System-Parameters Country-Options** screen to **Katakana**.

Completing the Station screens

The information that you enter on the **Station** screen advises the system that the telephone exists and indicates which features you want to enable on the telephone.

Communication Manager allows customers enter extensions with punctuation on the command line. Punctuation is limited to dashes (hyphens) and dots (periods). Communication Manager cannot process a command like **`add station`** ***`431 4875`***. You must format a command in one of these ways:

- **`add station`** ***`431-4875`***
- **`add station`** ***`431.4875`***
- **`add station`** ***`4314875`***

Make sure the extension conforms to your dial plan. You can also use the **`add station next`** command to add a telephone to the **`next`** available extension.

To access the **Station** screen for the new telephone:

1. Type **`add station`** ***`nnnn`***, where ***`nnnn`*** is the extension for the new telephone. Press **Enter**.

   Note:

   If you have Terminal Translation Initialization (TTI) enabled, you might receive the following error message when attempting to add a new station: "**No station/TTI port records available; ‘display capacity’ for their usage.**" If this occurs, try one or more of the following:

   - Remove any DCP or Analog circuit packs that have no ports administered on them.
   - If you are not using TTI or any related feature (such as PSA or ACTR), set the **Terminal Translation Initialization (TTI) Enabled?** field on the **Feature-Related System Parameters** screen to **n**.
   - Contact your Avaya technical support representative.

   For more information on TTI, see "Terminal Translation Initialization" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

   For more information on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

   When the **Station** screen appears, you see the extension number and some default field values. For example, the following screen is for a new telephone, displayed by typing **`add station next`**.

Figure 23: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
                                              Time of Day Lock Table:
              Loss Group: 2           Personalized Ringing Pattern: 3
             Data Module? n                        Message Lamp Ext: 1014
            Speakerphone: 2-way                 Mute button enabled? y
        Display Language? English
                   Model:                             Expansion Module?

 Survivable GK Node Name:                            Media Complex Ext:
          Survivable COR:                                 IP Softphone? y
   Survivable Trunk Dest?                           Remote Office Phone? y
                                                     IP Video Softphone?
                                                               IP Video?

                                                    Customizable Labels?
```

2. Type the model number of the telephone into the **Type** field.

   For example, to install a 6508D+ telephone, type **6480D+** in the **Type** field. Note that the displayed fields might change depending on the model you add.

3. Type the port address in the **Port** field.

   Note:

   Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

4. Type a name to associate with this telephone in the **Name** field. The name you enter displays on called telephones that have display capabilities. Some messaging applications, such as INTUITY, recommend that you enter the user's name (last name first) and their extension to identify the telephone. The name entered is also used for the integrated directory.

Note:

To hide a name in the integrated directory, enter two tildes (~~} before the name when you assign it to the telephone, and set **Display Character Set** on the **System Parameters Country-Options** screen to **Roman**. This hides the name in the integrated directory. The tildes are not displayed with Caller ID name. Note that this is the only method to hide a name in the integrated directory. Also, if a name is entered with only one tilde (~), the name is converted to Eurofont characters.

Note:

For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field is supported by Unicode language display. You must be using ASA or MSA. For more information on Unicode language display, see Administering Unicode display on page 203. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

5. Press **Enter** to save your changes.

To make changes to this new telephone, such as assigning coverage paths or feature buttons, type **`change station`** ***`nnnn`***, where ***`nnnn`*** is the extension of the new telephone. Press **Enter**. See Adding Feature Buttons on page 129 for more information.

Duplicating telephones

A quick way to add telephones is to copy the information from an existing telephone and modify it for each new telephone. For example, you can configure one telephone as a template for an entire work group. Then, you duplicate the template **Station** screen to add all the other extensions in the group.

Note that only telephones of the same model can be duplicated. The **`duplicate`** command copies all the feature settings from the template telephone to the new telephones.

To duplicate an existing telephone:

1. Type **`display station`** ***`nnnn`***, where ***`nnnn`*** is the extension of the **Station** screen you want to duplicate to use as a template. Press **Enter**. Verify that this extension is the one you want to duplicate.

   Press **Cancel** to return to the command prompt.

2. Type **`duplicate station`** ***`nnnn`***, where ***`nnnn`*** is the extension you want to duplicate. Press **Enter**.

   The system displays a blank duplicate **Station** screen.

   Alternately, you can duplicate a range of stations by typing **`duplicate station`** ***`<extension>`*** **`start`** ***`nnnn`*** **`count`** *`<1-16>`*, where ***`<extension>`*** represents the station you want to duplicate, ***`nnnn`*** represents the first extension number in a series, and **`count`** *`<1-16>`* represents the number of consecutive extensions after the **`start`** extension to create as duplicates.

   Note:

   If you want to duplicate the settings of another station, but need to change the port or station type, you must individually administer each station after creating the duplicates.

Figure 24: Duplicate Station screen - page 1

```
 duplicate station 1234567890123                                    Page 1 of 2
                                    STATION


                                                    Security            Endpt
Extension       Port    Name                        Code                ID
1234567890123   1234567 123456789012345678901234567 12345678            12
```

Figure 25: Duplicate Station screen - page 2

```
 duplicate station 1234567890123                                    Page 2 of 2
                                    STATION

Extension       Room       Jack  Cable
1234567890123   1234567890 12345 12345
```

3. Type the extension, port address, and telephone name for each new telephone you want to add.

   The rest of the fields on the **Station** screen are optional. You can complete them at any time.

4. Press **Enter** to save your changes to system memory.

To make changes to these telephones, such as assigning coverage paths or feature buttons, type **`change station`** ***`nnnn`***, where ***`nnnn`*** is the extension of the telephone that you want to modify. Press **Enter**

Related topics

You can also add multiple call center agents, all with the same settings based on an agent that is already administered. Enter **`command duplicate agent-loginID`** and the extension of the agent you want to duplicate, then **`start`** and the extension you want to use for the first new agent, then **`count`** and the number of agents you want to add. Fill in the information on the **Agent LoginID** screen. For more information, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Using an alias

Not every telephone model or device has a unique **Station** screen in the system. You might have to use an available model as an "alias" for another. If you need to enter a telephone type that the system does not recognize or support, use an alias. Defining aliases is also a useful method to identify items that act as analog stations on Communication Manager, such as fax machines, modems, or other analog device.

If you purchase a telephone model that is newer than your system, you can alias this telephone to an available model type that best matches the features of your new telephone. See your telephone's manual to determine which alias to use. If your manual does not have this information, you can contact the DEFINITY helpline for an appropriate alias.

For example, we will create two aliases: one to add a new 6220 telephone and one to add modems to our system.

1. See your new telephone's manual to find the correct alias.

   In our example, we find that the 6220 should be administered on an older system as a 2500 telephone.

2. Type **`change alias station`**. Press **Enter**.

   The **Alias Station** screen appears.

Figure 26: Alias Station screen

```
change alias station                                                Page 1 of 1
                                    ALIAS STATION
                        Alias Set Type    Supported Set Type
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
               '#' indicates previously aliased set type is now native
```

3. Enter **6220** in the **Alias Set Type** field.

   This is the name or model of the unsupported telephone.

4. Enter **2500** in the **Supported Set Type** field.

   Enter the supported model in this field.

5. Enter **modem** in the second **Alias Set Type** field.

   You can call the alias set anything you like. Once you define the alias, you can use the alias set in the **Type** field on the **Station** screen.

6. Enter **2500** in the second **Supported Set Type** field.

   Entering 2500 indicates to the system that these models are basic analog devices.

7. Press **Enter** to save your changes.

Now you can follow the instructions for adding a new telephone (or adding a fax or modem). Avaya Communication Manager now recognizes the new type (6220 or modem) that you enter in the **Type** field.

Be sure to see your telephone's manual for instructions on how to set feature buttons and call appearance buttons. Note that if you need to use an alias for a telephone, you might not be able to take advantage of all the features of the new telephone.

Customizing your telephone

This section provides recommendations for setting up or enhancing your personal telephone. You need a telephone that is powerful enough to allow you to use all the features you might give to other employees. You might want to add feature buttons that allow you to monitor or test the system, so that you can troubleshoot the system from your telephone.

It will be much easier to monitor and test your system if you have a telephone with:

- a large multi-button display (such as 8434D or 8410D)
- a class of service (cos) that has console permissions
- the following feature buttons
  - ACA and Security Violations (assign to lamp buttons)
  - Busy verify
  - Cover message retrieval button
  - Major/minor alarm buttons
  - Trunk ID buttons
  - Verify button

Once you select a telephone, you'll want to determine if you want to place this telephone at your desk or in the server room. If the telephone is in the server room (near the system administration terminal), you can quickly add or remove feature buttons to test features and facilities. You might decide that you want a telephone at both your desk and in the server room — it's up to you.

You might also find it handy to set up multiple telephones for testing applications and features before you provide them to users. You might want to have a telephone that mimics each type of user telephone in your organization. For example, if you have four basic telephone templates, one for executives, one for marketing, one for technicians, and one for other employees, you might want to have examples of each of these telephones so you can test new features or options. Once you are satisfied that a change works on the test telephone, you can make the change for all the users in that group.

Upgrading telephones

If you want to change telephone types for a user and do not need to change locations, you can just access the **Station** screen for that extension and enter the new model number.

> **Tip:**
>
> This method can be used only if the new telephone type matches the existing port type (such as digital telephone with a digital port).

For example, if a user at extension 4556 currently has a 7410+ telephone and you want to replace it with a new 8411D telephone:

1. Type **`change station 4556`**. Press **Enter**.

   The **Station** screen for 4556 appears.

2. Overwrite **7410+** with **8411D** in the **Type** field.

   Now you can access the functions and feature buttons that correspond to an 8411D telephone.

Swapping telephones

You will often find that you need to move or swap telephones. For example, employees moving from one office to another might want to bring their telephones. In this case, you can use X ports to easily swap the telephones.

In general, to swap one telephone (telephone A) with another telephone (B), you change telephone A's port assignment to **x**, change telephone B's port assignment to A's old port, and, finally, change the **x** for telephone A to B's old port. Note that these swapping instructions work only if the two telephones are the same type (both digital or both analog, etc.).

For example, to swap telephones for extension 4567 (port 01C0505) and extension 4575 (port 01C0516), complete the following steps:

1. Type **`change station 4567`**. Press **Enter**.
2. Record the current port address (`01C0505`) and type **x** in the **Port** field.
3. Press **Enter** to save your changes.
4. Type **`change station 4575`**. Press **Enter**.
5. Record the current port address (`01C0516`).
6. Type **01C0505** in the **Port** field.
7. Update the **Room** and **Jack** fields.
8. Press **Enter** to save your changes.
9. Type **`change station 4567`** again. Press **Enter**.
10. Type **01C0516** in the **Port** field.

    This is the port that used to be assigned to extension 4575.
11. Update the **Room** and **Jack** fields.
12. Press **Enter** to save your changes.
13. Physically unplug the telephones and move them to their new locations.

When you swap telephones, the system keeps the old button assignments. If you are swapping to a telephone with softkeys, the telephone could have duplicate button assignments, because softkeys have default assignments. You might want to check your button assignments and modify them as necessary.

Using ACTR to move telephones

Automatic Customer Telephone Rearrangement (ACTR) allows a telephone to be unplugged from one location and moved to a new location without additional administration in Avaya Communication Manager. Communication Manager automatically associates the extension to the new port. ACTR works with 6400 Serialized telephones and with the 2420/2410 telephones. The 6400 Serialized telephone is stamped with the word "Serialized" on the faceplate for easy identification. The 6400 Serialized telephone memory electronically stores its own part ID (comcode) and serial number, as does the 2420/2410 telephone. ACTR uses the stored information and associates the telephone with new port when the telephone is moved.

ACTR is an enhancement to Terminal Translation Initialization (TTI), Personal Station Access (PSA), Customer Telephone Activation (CTA). ACTR makes it easy to identify and move telephones.

CAUTION:

When a telephone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Access Point (PSAP) could send emergency response personnel to the wrong location.

Detailed description

On the **Feature-Related System Parameters** screen, set the **Terminal Translation Initialization (TTI) Enabled** field to **y** and the **TTI State** field to **voice**.

Note:

When a telephone is moved, if there is any local auxiliary power (a power supply plugged into a local AC outlet), the telephone must be plugged into an AC outlet at the telephone’s new location. A telephone with remote auxiliary power must be supplied remote auxiliary power at its new location. If you do not supply auxiliary power in either case after a telephone is moved, some optional adjuncts (for example, an expansion module) do not operate.

When you enter **always** or **once** in the **Automatic Moves** field on the **Station** screen, Communication Manager adds the extension to its ACTR Move List database. When the telephone is plugged in, Communication Manager asks the telephone for its serial number and records the serial number on the ACTR Move List. If you change the entry in the **Automatic Moves** field from **always** or **once** to **no**, Communication Manager removes the extension from the Move List.

Call processing

When a telephone is unplugged while on a call, and a 6400 Serialized telephone or a 2420/ 2410 telephone that is administered for automatic moves is plugged into the port within 60 seconds:

- both extensions are placed in idle state
- active calls on either extension are dropped, unless the call is active on a bridged appearance at some other telephone
- held calls remain in a hold state
- any calls ringing on either extension instantly proceed to the next point in coverage or station hunting path, unless the call is ringing on a bridged appearance at some other telephone
- user actions that were pending when the new telephone was plugged in are aborted

Design considerations

You can use the **`list station movable`** command to keep track of extensions on the move list. Once you reach the maximum number, Communication Manager does not allow additional extensions.

Administration

Before you start:

- Be sure the **TTI** field on the **Feature-Related System Parameters** screen is set to **y**.
- Before you move a telephone in your system, set the **TTI State** field to voice on the **Feature-Related System Parameters** screen.

Moving telephones

You can allow a telephone to be unplugged from one location and moved to a new location without additional administration on Avaya Communication Manager.

For example, to allow moves anytime for a telephone at extension 1234:

1. Type **`change station 1234`**. Press **Enter**.
2. Move to the **Automatic Moves** field.

Figure 27: Station screen

```
change station nnnn                                            Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
             CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                      Call Waiting Indication:
 Per Button Ring Control? n                Attd. Call Waiting Indication:
   Bridged Call Alerting? n                  Idle Appearance Preference? n
       Switchhook Flash? n                 Bridged Idle Line Preference? y
    Ignore Rotary Digits? n                    Restrict Last Appearance? y
  Active Station Ringing: single        Conf/Trans On Primary Appearance? n
                                                      EMU Login Allowed?
        H.320 Conversion? n         Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed      Busy Auto Callback without Flash? y
         Multimedia Mode: basic
     MWI Served User Type: ______            Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                   Select Last Used Appearance? n
                              _               Coverage After Forwarding? _
      Recall Rotary Digit? n                     Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n       IP Audio Hairpinning? n
Precedence Call Waiting? y
```

3. Type **always** in the **Automatic Moves** field.
4. Press **Enter** to save your changes.

Using TTI to move telephones

Terminal Translation Initialization (TTI) allows you to merge an x-ported station to a valid port by dialing a TTI merge code, a system-wide security code, and the x-port extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the station to revert to an x-port.

TTI can be used for implementing telephone and data module moves from office to office. That is, you can separate a telephone from its port with TTI, unplug the telephone from the jack, plug in the telephone in a jack in a different office, and merge the telephone to its new port with TTI.

If you are moving telephones and concerned about security, you might also want to see Setting up Personal Station Access on page 431 for more information about setting the security code for each extension.

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use might cause you security problems. For example, someone who knows the TTI security code could disrupt normal business functions by separating telephones or data terminals. You can help protect against this action by frequently changing the TTI security code. You can further enhance system security by removing the feature access code (FAC) from the system when it does not need to be used (for example, there are no moves going on at present). Consult the *Avaya Products Security Handbook* for additional steps to secure your system and find out about obtaining information regularly about security developments.

Before you start

Before you can merge a telephone, you must set the **TTI State** field to **voice** on the **Feature-Related System-Parameters** screen. You also must set the extension to match the port type of the TTI port making the merge request. For example, a digital telephone type can merge only to a port on a digital board.

CAUTION:

When a telephone is unplugged and moved to another physical location, the **Emergency Location Extension** field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Note:

You cannot use TTI to change a virtual extension.

Instructions

Merging an extension with a TTI telephone

CAUTION:
You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To merge an extension to a telephone with TTI, complete the following steps from the telephone you want to merge:

1. Dial the TTI merge FAC.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.
2. Dial the TTI security code from the telephone you want to merge.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.
3. Dial the extension of the telephone you want to merge.
   - If the extension is valid, you receive confirmation tone, which might be followed by dial tone. (It is possible to receive intercept tone immediately following the confirmation tone. If this happens, you need to attempt the merge again.)
   - If the extension is valid, but the extension is being administered, you receive reorder tone. Try the merge again later.
   - If the extension is invalid, you receive intercept tone.
   - If the system is busy and cannot complete the merge, you receive reorder tone. Try the merge again later.
   - If the telephone has a download status of pending, you receive reorder tone. You need to change the download status to complete to successfully complete the TTI merge.

Separating TTI from a telephone

To complete a TTI separation, complete the following steps from the telephone that needs to be separated:

1. Dial the TTI separate FAC.
2. Dial the TTI security code.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.

3. Dial the extension of the telephone to be separated.
   - If you have dialed the extension of the telephone currently merged with this telephone, you receive confirmation tone.
   - If you have dialed the extension of the telephone currently merged with this telephone, but the extension is being administered, you receive reorder tone. Try the separation again later.
   - If you have not dialed the extension of the telephone currently merged with this telephone, you receive intercept tone.
   - If the system is busy and cannot complete the separation, you receive reorder tone. Try the separation again later.

Fixing problems

If you are having difficulty using TTI, you might want to review the following system restrictions:

- The **TTI Ports** field on the **System Capacity** screen (type **`display capacity`**) shows the number of TTI ports used in a server running Communication Manager. This field shows only the number of TTI ports being administered. If a TTI exceeds the maximum number of ports, the port is not administered and cannot be added. In that case, a telephone cannot be added. For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

  BRI endpoints are only counted as one TTI port. For example, for every two BRI endpoints, one TTI port is counted. As such, you can have two telephones assigned to one port. If either endpoint is administered, the TTI port count is reduced by 1.
- The total number of translated telephones and Voice TTI ports in a system is limited to the maximum number of administered telephones supported in the system. The total number of translated data terminals and Data TTI ports in a system is limited to the maximum number of administered data modules allowed in the system.
- Set the **TTI State** field to **voice** and then set the **TTI State** field to **data**. When you use this order, voice and then data, you reduce the chance of a user trying to use TTI on a data-only terminal that does not have TTI port translation. This can happen when the number of telephones allowed by the system is twice the number of data terminals. For example, if the system limit for telephones is 15,000 and 7,500 for data, then when TTI was turned on for data first, only the first 7,500 unadministered ports would get TTI port translations.

- When TTI is activated for the system, the following actions take place:
  - If the **TTI State** field was previously activated but in a different state (such as, a voice to data state), the old TTI translations are removed and the new ones added on a board by board basis.
  - If the **TTI State** field is set to **voice**, then default TTI translations are generated for every unadministered port on all digital, hybrid, and analog boards.
  - If the **TTI State** field is set to **data**, then default TTI translations are generated for every unadministered port on all digital and data line boards in the system.
  - Whenever a new digital board is inserted when the system is in TTI Data mode, or when a digital, hybrid, or analog board is inserted when the system is in TTI Voice mode, the unadministered ports on the board become TTI ports.
  - When TTI is deactivated, all translation for the TTI ports are removed in the system; the ports return to an unadministered state.

Removing telephones

Before you physically remove a telephone from your system, check the telephone's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove a telephone at extension 1234:

1. Type **`status station 1234`**. Press **Enter**.

   The **General Status** screen appears.
2. Make sure that the telephone:
   - is plugged into the jack
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)
3. Type **`list groups-of-extension 1234`**. Press **Enter**.

   The **Extension Group Membership** screen shows whether the extension is a member of any groups on the system.
4. Press **Cancel**.
5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **`change pickup group 2`** and delete the extension from the list.

6. Type **`list usage extension 1234`**. Press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

7. Press **Cancel**.
8. If the extension appears on the **Usage** screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 is bridged onto extension 1235, type **`change station 1235`** and remove the appearances of 1234.

9. Type **`change station 1234`**. Press **Enter**.
10. Delete any bridged appearances or personal abbreviated dialing entries. Press **Enter**.
11. Type **`remove station 1234`**. Press **Enter**.

    The system displays the **Station** screen for this telephone so you can verify that you are removing the correct telephone.

    Tip:

    Be sure to record the port assignment for this jack in case you want to use it again later.

12. If this is the correct telephone, press **Enter**.

    If the system responds with an error message, the telephone is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and enter **`remove station 1234`** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.
14. Type **`save translations`**. Press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the set from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Once you successfully remove a set, that set is permanently erased from system memory. If you want to reactivate the set, you have to add it again as though it were a new telephone.

Adding a fax or modem

Connecting a fax machine or modem to your system is similar to adding a telephone, with a few important exceptions. If you have not added a telephone, you might want to read Adding new telephones on page 83.

Because the system does recognize the concept of "fax" or "modem," you need to administer these items as basic analog stations. You can merely use the supported station type 2500 (analog, single line).

Alternatively, you can create aliases to the 2500 for fax machines and modems. If you want to be able to create reports that indicate which stations are faxes or modem, you should create aliases for these items. For more information about aliasing, see Using an alias on page 91.

For this example, let us assume that we have already defined an alias for ‘fax’ as a 2500 and that we now want to add a fax machine to extension 4444.

To add a fax machine as extension 444, complete the following steps:

1. Type **`add station 4444`**. Press **Enter**.
2. In the **Type** field, type **fax**.
3. In the **Port** field, type the port address.
4. In the **Name** field, type a name to associate with this fax.
5. Move to the **Data Restriction** field and type **y**.

   Entering **y** in this field prevents calls to and from this extension from being interrupted by tone signals. This is important for fax machines and modems as these signals can disrupt transmissions of data.
6. In the **Distinctive Audible Alert** field, type **n**.

   This eliminates the distinct 2-burst ring for external calls, which often interferes with the auto-answer function on fax machines or modems.
7. Press **Enter** to save your changes.

Enabling transmission over IP networks for modem, TTY, and fax calls

Prerequisites

The ability to transmit fax, modem, and TTY calls over IP trunks or LANs and WANs assumes that the endpoints sending and receiving the calls are connected to a private network that uses H.323 trunking or LAN connections between gateways and/or port networks. This type of transmission also assumes that calls can either be passed over the public network using ISDN-PRI trunks or passed over an H.323 private network to Communication Manager switches that are similarly enabled.

As a result, it is assumed that you have assigned, or will assign, to the network gateways the IP codec you define in this procedure. For our example, the network region 1 will be assigned codec set 1, which you are enabling to handle fax, modem, and TTY calls.

To enable transmission over IP networks for modem, TTY, and fax calls:

1. Type **`change ip-codec-set 1`**. Press **Enter**.

   The **IP Codec Set** screen appears.

Complete the fields as required for each media type you want to enable. Press **Enter**.

For more information on modem/fax/TTY over IP, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Adding an IP Softphone

Avaya IP Softphones enable the end user to control telephone calls directly from a personal computer (PC). An end user can log in remotely to your company’s server running Avaya Communication Manager and then make and receive telephone calls from the telephone extension.

Avaya IP Softphone supports the following modes:

- Road-Warrior

  You typically use this mode for laptop users who are travelling. In this mode, the PC LAN connection carries both the call control signaling and the voice path. Because the audio portion of the voice call is handled by the PC, you must have some kind of audio device (e.g., handset, headset) PC to provide the audio connection.

- Telecommuter or Avaya IP Agent

  For the telecommuter or Avaya IP Agent mode, you make two separate connections to the Avaya DEFINITY server. The signaling path is carried over an IP network and the voice path is carried over the standard circuit-switched telephone network (PSTN). Since you are using a telephone for audio, you do not need an H.323 PC audio application.

  The telecommuter mode uses the Avaya IP Softphone interface (on the user's PC) and a standard telephone. The Avaya IP Agent mode uses the Avaya IP Agent interface (on the agent's PC) and a call center telephone.

- Native H.323 (only available with Avaya IP Softphone R2)

  The stand-alone H.323 mode enables travelers to use some Communication Manager features from a remote location. This mode uses a PC running an H.323 v2-compliant audio application, such as Microsoft NetMeeting. The H.323 mode controls the call signaling and the voice path. However, since it does not use the IP Softphone interface, this configuration is capable of operating only as an analog or single-line telephone making one call at a time without any additional assigned features. You can provide stand-alone H.323 users only features that can they can activate with dial access codes.

- Control of IP Telephone (only available with IP Softphone R4 and later)

  This mode allows you to make and receive calls under the control of the IP Softphone - just like in the Telecommuter or Road Warrior mode. The big difference is that you have a real digital telephone under your control. In the Road Warrior mode, there is no telephone. In the Telecommuter mode, the telephone you are using (whether analog, digital, or IP telephone is brain dead). In this mode (if you have an IP telephone), you get the best of both worlds.

- Control of DCP Telephone (only available with IP Softphone R5 and later)

  This feature provides a registration endpoint configuration that will allow an IP softphone and a non-softphone telephone to be in service on the same extension at the same time. In this new configuration, the call control is done by both the softphone and the telephone endpoint. The audio is done by the telephone endpoint.

  Tip:

  Use **`status station`** to show the part (product) ID, serial number, and the audio connection method used by existing stations.

Note:

Beginning with the November 2003 release of Communication Manager, R1 and R2 IP Softphone and IP Agent, which use a dual connect (two extensions) architecture, are no longer supported. R3 and R4 IP Softphone and IP Agent, which use a single connect (one extension) architecture, continue to be supported. This applies to the RoadWarrior and the Telecommuter configurations for the IP Softphone. Native H.323 registrations for R1 and R2 Softphones continue to be supported.

Before you start

Be sure that your system has been enabled to use IP Softphones. Display the **System Parameters Customer-Options (Optional Features)** screen and verify the following field settings:

- **Maximum Concurrently Registered IP Stations** is greater than **0**
- **IP Stations** field is **y**
- Information has been entered in the fields on the **Maximum IP Registrations by Product ID** page

Be sure that your DEFINITY CSI has a CLAN board and an IP Media Processor board.

Once you're finished administering Avaya Communication Manager, you need to install the IP Softphone software on each user's PC.

Adding a road-warrior mode

You can use the road-warrior mode when you have only a single telephone line available to access Avaya Communication Manager over the IP network.

You also can "take over" an IP telephone. Typically you would not have a different extension for your softphone. When you log in, the softphone takes over the existing telephone extension (turn the DCP or IP telephone off). During this time, that DCP or IP telephone is out of service. This is accomplished if, on the **Station** screen, the **IP Softphone** field is **y**.

We will add a road-warrior mode at extension 3001. Except for single-connect IP telephones, you have to actually administer two extensions for each road-warrior mode. We will first add an H.323 extension at 3000.

To assign an H.323 extension, complete the following steps:

1. Type **`add station 3000`**. Press **Enter.**

   The [Station screen]() appears.

Figure 28: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2                Personalized Ringing Pattern: 3
             Data Module? n                           Message Lamp Ext: 1014
            Speakerphone: 2-way                     Mute button enabled? y
       Display Language? English
                   Model:                               Expansion Module?

 Survivable GK Node Name:                              Media Complex Ext:
          Survivable COR:                                   IP Softphone? y
   Survivable Trunk Dest?                            Remote Office Phone? y
                                                      IP Video Softphone?
                                                                IP Video?
```

2. In the **Type** field, enter **H.323**.
3. Press **Enter** to save your work.

Now, you need to administer the telephone (DCP) extension. To do so, complete the following steps:

1. Type **`add station next`**. Press **Enter**.

   The [Station screen]() appears.

Figure 29: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 3
             Data Module? n                         Message Lamp Ext: 1014
            Speakerphone: 2-way                   Mute button enabled? y
        Display Language? English
                   Model:                              Expansion Module?

 Survivable GK Node Name:                             Media Complex Ext:
          Survivable COR:                                  IP Softphone? y
   Survivable Trunk Dest?                            Remote Office Phone? y
                                                      IP Video Softphone?
                                                                IP Video?
```

Note that you choose to change an existing DCP extension by using **`change station`** ***`nnnn`*** in this step, where ***`nnnn`*** is the existing DCP extension.

2. In the **Type** field, enter the model of telephone you want to use, such as **6408D**.
3. In the **Port** field, type **x** for virtual telephone or enter the port number if there is hardware.

Note:

Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

4. In the **Security Code** field, enter the password for this remote user, such as **1234321**.

   This password can be 3-8 digits in length.
5. In the **Media Complex Ext** field, type **3000**.

   This is the H.323 extension just administered.
6. In the **IP Softphone** field, type **y**.
7. On page 2, in the **Service Link Mode** field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote telephone users, such as call center agents.

8. In the **Multimedia Mode** field, type **enhanced**.
9. Press **Enter** to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3001) and password (1234321).

Adding a telecommuter mode

Assign this configuration to remote users who have two available telephone lines. For example, to administer a telecommuter mode for a home user at extension 3010, complete the following steps:

1. Type `add station 3010`. Press **Enter.**

   The Station screen appears.

   Note:

   Use the `add station` command if this is a new DCP extension. Use the `change station` command for an existing DCP extension and ignore steps 2 and 3.)
2. In the **Port** field, type **x** for virtual telephone or enter the port number if there is hardware.
3. In the **Security Code** field, enter the password for this remote user, such as **1234321**.

   This password can be up to 7 digits in length.
4. In the **IP Softphone** field, type **y**.
5. On page 2, in the **Service Link Mode** field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote telephone users, such as call center agents.
6. In the **Multimedia Mode** field, type **enhanced**.
7. Press **Enter** to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3010) and password (1234321).

Fixing problems

Problem	Possible causes	Solutions
Display characters on the telephone can not be recognized.	Microsoft Windows is not set to use Eurofont characters.	Set the Microsoft Windows operating system to use Eurofont.

Related topics

See the online help for assistance, or, on the Avaya IP Softphone CD, refer to *Avaya IP Softphone Overview and Troubleshooting* for customer information on Avaya IP Softphone modes. This is a Portable Document Format (PDF) document that is located in the Overview Document folder on the Avaya IP Softphone CD.

Also see *Getting Started*, also located on the Avaya IP Softphone CD for more information on how to install and configure the IP Softphone software.

Adding an IP telephone

The 4600-series IP Telephones are physical sets that connect to Avaya Communication Manager via TCP/IP.

CAUTION:
An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.

Before you start

Verify the system has a:

- TN2302 IP Media Processor circuit pack for audio capability
- TN799 Control-LAN circuit pack for signaling capability (for CSI Servers only)

Be sure that your system has been enabled to use IP Telephones. Display the **System-Parameters Customer-Options (Optional Features)** screen and verify the following field settings:

- **Maximum Concurrently Registered IP Stations** is greater than 0
- **IP Stations** field is **y**
- Information has been entered in the fields on the **Maximum IP Registrations by Product ID** page

Instructions

Let us add an IP telephone at extension 4005.

To assign an extension, complete the following steps:

1. Type **`add station 4005`**. Press **Enter.**

   The **Station screen** appears.

Figure 30: Station screen

```
add station nnnn                                            Page 1 of X
                              STATION

Extension:                    Lock Messages? n                      BCC: 0
     Type:                    Security Code:                         TN: 1
     Port:                  Coverage Path 1:                        COR: 1
     Name:                  Coverage Path 2:                        COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2               Personalized Ringing Pattern: 3
             Data Module? n                         Message Lamp Ext: 1014
            Speakerphone: 2-way                   Mute button enabled? y
        Display Language? English
                   Model:                              Expansion Module?

 Survivable GK Node Name:                             Media Complex Ext:
          Survivable COR:                                  IP Softphone? y
   Survivable Trunk Dest?                           Remote Office Phone? y
                                                      IP Video Softphone?
                                                                IP Video?
```

2. In the **Type** field, enter the station type, in this case **4620**.

Note:

When adding a new 4601 or 4602 IP telephone, you must use the 4601+ or 4602+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4601 or 4602, you receive a warning message, stating that you should upgrade to the 4601+ or 4602+ station type in order to access the Automatic Callback feature.

3. The **Port** field is display-only, and **IP** appears.
4. In the **Security Code** field, enter a password for the IP telephone user.

   Note:

   Although the system accepts a null password, the IP telephone will not work unless you assign a password.
5. Press **Enter** to save your work.

Changing from dual-connect to single-connect IP telephones

When you have a dual extension telephone and you upgrade to a single extension telephone, you can remove the connection that is no longer used for that telephone.

To remove the H.323 connection that is no longer needed, first record the media complex extension number:

1. Type `change station nnnn`, where *nnnn* is the extension number of the original dual-connect telephone that you are replacing with a single-connect telephone.

   The **Station** screen appears.
2. Move to the **Media Complex Extension** field.
3. Write down the number in the **Media Complex** field, then delete the number from the field.
4. Press **Enter** to save your changes.

Now remove the extension you recorded. Before you remove an H.323 extension from your system, check the status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, if you wrote down extension 1234 before you removed it from the **Media Complex** field on the **Station** screen, then remove extension 1234 using these steps:

1. Type `status station 1234`. Press **Enter**.

   The **General Status** screen appears.
2. Make sure that the extension:
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)

3. Type **`list groups-of-extension 1234`**. Press **Enter**.

   The **Extension Group Membership** screen shows whether the extension is a member of any groups on the system.

4. Press **Cancel**.

5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **`change pickup group 2`** and delete the extension from the list.

6. Type **`list usage extension 1234`**. Press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

7. Press **Cancel**.

8. If the extension appears on the **Usage** screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type **`change hunt group 2`** and delete the extension from the list.

9. Type **`change station 1234`**. Press **Enter**.

10. Delete any bridged appearances or personal abbreviated dialing entries. Press **Enter**.

11. Type **`remove station 1234`**. Press **Enter**.

    The system displays the **Station** screen for this telephone so you can verify that you are removing the correct telephone.

12. If this is the correct telephone, press **Enter**.

    The system responds with **command successfully completed**.

    If the system responds with an error message, the telephone is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and type **`remove station 1234`** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.

14. Type **`save translations`**. Press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Once you successfully remove the extension, it is permanently erased from system memory. If you want to reactivate the extension, you have to add it again as though it were new.

Setting up emergency calls on IP telephones

Set up which "calling number" to send to the public safety access point when an emergency call is placed from an IP telephone.

Instructions

You use the **Station screen** to set up emergency call handling options for IP telephones. As an example, we'll administer the option that prevents emergency calls from an IP telephone.

To prevent an emergency call from an IP telephone:

1. Type **`change station nnnn`**, where ***`nnnn`*** is the extension of the telephone you want to modify. Press **Enter**. The **Station** screen appears.
2. Click **Next Page** to find the **Remote Softphone Emergency calls** field.

Figure 31: Station screen

```
change station nnnn                                              Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                       Call Waiting Indication:
 Per Button Ring Control? n                 Attd. Call Waiting Indication:
   Bridged Call Alerting? n                   Idle Appearance Preference? n
        Switchhook Flash? n                 Bridged Idle Line Preference? y
   Ignore Rotary Digits? n                      Restrict Last Appearance? y
  Active Station Ringing: single         Conf/Trans On Primary Appearance? n
                                                           EMU Login Allowed?
        H.320 Conversion? n          Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed        Busy Auto Callback without Flash? y
         Multimedia Mode: basic
    MWI Served User Type: ______                Display Client Redirection? n
        Automatic Moves:
              AUDIX Name:                       Select Last Used Appearance? n
                              _                   Coverage After Forwarding? _
     Recall Rotary Digit? n                         Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n        IP Audio Hairpinning? n
Precedence Call Waiting? y
```

3. Type **block** in the **Remote Softphone Emergency calls** field. Press **Enter** to save your changes.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya representative if you have questions about emergency calls from IP telephones.

Setting up Remote Office

Avaya Remote Office provides IP processing capabilities to traditional call handling for voice and data between Avaya Communication Manager and offices with Remote Office hardware. You need to add the information about Remote Office as a node in Communication Manager, add its extensions, and set up the trunk and signaling groups.

Before you start

Be sure the following fields on the **System Parameters Customer-Options (Optional Features)** screen are set to **y** or completed. If not, contact your Avaya representative.

- **Maximum Administered Remote Office Trunks**
- **Maximum Administered Remote Office Stations**
- **Product ID registration limit**
- **Remote Office**
- **IP station**
- **ISDN-PRI**

Also, be sure your Remote Office hardware is installed and administered at the remote location. You need the following information from the remote administration:

- IP address
- Password

Adding Remote Office to Avaya Communication Manager

In our example, we'll set up a remote-office location using Avaya R300 Remote Office Communicator hardware in our branch office in Santa Fe. We'll add a new node, and set up the signaling group and trunk group.

Adding a node

Instructions

To add the remote office node to Communication Manager:

1. Type **`change node-names IP`**. Press **Enter**.

   The **Node Name** screen appears.

Figure 32: Node Name screen

```
change node-names ip                                              Page 1 of X

                              IP NODE NAMES

Name                    IP Address                Name            IP Address
 1. ______________     ___.___.___.___  17. ______________     ___.___.___.___
 2. ______________     ___.___.___.___  18. ______________     ___.___.___.___
 3. ______________     ___.___.___.___  19. ______________     ___.___.___.___
 4. ______________     ___.___.___.___  20. ______________     ___.___.___.___
 5. ______________     ___.___.___.___  21. ______________     ___.___.___.___
 6. ______________     ___.___.___.___  22. ______________     ___.___.___.___
 7. ______________     ___.___.___.___  23. ______________     ___.___.___.___
 8. ______________     ___.___.___.___  24. ______________     ___.___.___.___
 9. ______________     ___.___.___.___  25. ______________     ___.___.___.___
10. ______________     ___.___.___.___  26. ______________     ___.___.___.___
11. ______________     ___.___.___.___  27. ______________     ___.___.___.___
12. ______________     ___.___.___.___  28. ______________     ___.___.___.___
13. ______________     ___.___.___.___  29. ______________     ___.___.___.___
14. ______________     ___.___.___.___  30. ______________     ___.___.___.___
15. ______________     ___.___.___.___  31. ______________     ___.___.___.___
16. ______________     ___.___.___.___  32. ______________     ___.___.___.___
```

2. In the **Name** field, type in a word to identify the node.

   In our example, type **Remote 6**.

3. In the IP address field, type in the IP address to match the one on the Avaya R300 administration.

4. Press **Enter** to save your changes.

5. Type **`add remote office`** and the number for this remote office. Press **Enter**.

   The **Remote Office** screen appears.

Figure 33: Remote Office screen

```
add remote office n

                              REMOTE OFFICE

     Node Name: _______________
Network Region: ___
      Location: __
     Site Data: ______________________________
                ______________________________
                ______________________________
```

6. Fill in the following fields:
   - **Node Name** - match the name on the **IP Node Names** screen.
   - **Network Region** - this must match the network region on the **IP Interfaces** screen for the circuit packs that connect this remote office. Use **`display ip-interfaces`** to find this information.
   - **Location** - match the one set up on the **Location** screen for this remote office.
   - **Site Data** - identify the street address or identifier you want to use
7. Press **Enter** to save your changes.

Tip:

Use **`status remote office`** to verify that your server running Communication Manager recognizes the Remote Office information. It also displays the extensions and signaling group you administer next.

Setting up a trunk group

You can modify an existing trunk group or add a new one. In our example, we'll add trunk group 6. Before you start, perform Setting up a signaling group on page 119.

To set up the trunk group for your remote office:

1. Type **`add trunk group 6`**.

   The **Trunk Group** screen appears.

Figure 34: Trunk Group screen

```
add trunk-group next                                               Page   1 of   xx
                                TRUNK GROUP

Group Number: 1                    Group Type: isdn         CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1         TAC:
   Direction: outgoing       Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? n                Busy Threshold: 255
Queue Length: 0
Service Type:                                                TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

2. In the **Group Type** field, type **ISDN**.

   ISDN-PRI or **ISDN-BRI** must be **y** on the **System Parameters Customer-Options (Optional Features)** screen.

3. In the **TAC** field, type in the trunk access code that conforms to your dial plan.
4. In the **Carrier Medium** field, type **IP** (Medpro).
5. In the **Dial Access** field, type **y**.
6. In the **Service Type** field, type **tie**.
7. In the **Signaling Group** field, type in the signaling group you created.
8. Press **Enter** to save your changes.

Setting up a signaling group

Each Remote Office has own listen port and signaling group. Set up a new trunk group, or use an existing trunk group administered for H.323 signaling.

Instructions

Set up the signaling group for remote office:

1. Type **`add signaling-group`** and the number of the group you want to add.

   The **Signaling Group** screen appears.

Figure 35: Signaling Group screen

```
add signaling-group n                                          Page 1 of x
                               SIGNALING GROUP

Group Number  ___           Group Type:  h.323
                         Remote Office? n      Max Number of NCA TSC: ___
                                   SBS? y      Max number of CA TSC:  ___
                              IP Video?      Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
    TSC Supplementary Service Protocol: _
                       T303 Timer(sec): 10

         Near-end Node Name:              Far-end Node Name:
       Near-end Listen Port:            Far-end Listen Port:
                                     Far-end Network Region:
               LRQ Required? n     Calls Share IP Signaling Connection? n
               RRQ Required? n
           Media Encryption? y
                 Passphrase:          Bypass If IP Threshold Exceeded? y
                                                 H.235 Annex H Required? n
                      DTMF over IP:     Direct IP-IP Audio Connections? y
Link Loss Delay Timer (sec):                        IP Audio Hairpinning? y
Enable Layer 3 Test?                         Interworking Message: PROGress
                                   DCP/Analog Bearer Capability:
```

2. In the **Group Type** field, type **H.323**.
3. In the **Remote Office** field, type **y**.
4. In the **Trunk Group for Channel Selection** field, type the number of the trunk you set up for the remote office.
5. In the **Near-end Node Name** field, identify the node name assigned to the CLAN that supports the R300.
6. In the **Far-end Node Name** field, identify the node name assigned to the CLAN that supports the R300.
7. In the **Near-end Listen Port** field, type a port number in the 5000-9999 range.
8. In the **Far-end Listen Port** field, type **1720**.
9. In the **RRQ** field, type **y**.
10. Tab to the **Direct IP-IP Audio Connection** field on another page of this screen and type **y**.
11. Press **Enter** to save your changes.

Setting up Remote Office on network regions

Now we will set up a network region and show the connections between regions.

Instructions

Set up network region 1:

1. Type **`add ip-network-region 1`**. Press **Enter**.

   The **IP Network Region** screen appears.

Figure 36: IP Network Region screen

```
change ip-network-region n                                           Page 1 of x
                              IP NETWORK REGION
  Region: n
Location:                  Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                            IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value:                       Server IP Address:   .   .   .
        Audio PHB Value:                             Server Port: 5005
        Video PHB Value:
802.1P/Q PARAMETERS                    RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? y
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20     Retry upon RSVP Failure Enabled? y
     Keep-Alive Interval (sec): 6                RSVP Profile: guaranteed-service
             Keep-Alive Count: 5      RSVP unreserved (BBE) PHB Value: 40
```

2. In the **Name** field, describe the region you are setting up.
3. In the **Codec Set** field, type the codec set you want to use in this region.
4. In the **UDP Port Range** field, type the range of the UDP port number to be used for audio transport.
5. In the **Intra-region IP-IP Direct Audio** field, type **y**.
6. In the **Inter-region IP-IP Direct Audio** field, type **y**.
7. Move to page 3 to set up connections between regions and assign codecs for inter-region connections.

   Note:

   Page 2 of the **IP Network Region screen** shows a list of LSP servers for the network region, and pages 4 through 19 are duplicates of page 3 (Figure 37), providing the ability to administer up to 250 locations.

Figure 37: IP Network Region screen

```
change ip-network-region n                                           Page   3 of x

                   Inter Network Region Connection Management

 src dst    codec   direct                                   Dynamic CAC
 rgn rgn     set     WAN   WAN-BW limits Intervening-regions   Gateway    IGAR
 3   1        1       y     256:Kbits
 3   2        1       n                   1 ___  ___  ___
 3   3        1
 3   4        1       n                   1 ___  ___  ___
 3   5        1       n                   6 ___  ___  ___
 3   6        1       y     :NoLimit
 3   7        1       y     10:Calls
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

The following connections are administered in this example:

- codec-set 2 is used between region 1 and region 4
- codec-set 5 is used between region 1 and region 99
- codec-set 6 is used between region 1 and region 193.

Now you need to assign the region number to the CLAN circuit pack. All the endpoints registered with a specific CLAN circuit pack belong to the CLAN's region. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Adding telephones to Remote Office

Be sure the extensions you add fit your dialing plan.

Instructions

1. Type **`add station`** ***`nnnn`***, where ***`nnnn`*** is the extension you are adding. Press **Enter**.

   The **Station screen** appears.

Figure 38: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2               Personalized Ringing Pattern: 3
             Data Module? n                          Message Lamp Ext: 1014
            Speakerphone: 2-way                    Mute button enabled? y
        Display Language? English
                   Model:                             Expansion Module?

 Survivable GK Node Name:                            Media Complex Ext:
          Survivable COR:                                 IP Softphone? y
   Survivable Trunk Dest?                          Remote Office Phone? y
                                                    IP Video Softphone?
                                                              IP Video?
```

2. In the **Type** field, type in the model of the telephone you are adding.
3. In the **Port** field, type **x**.

   This indicates that there is no hardware associated with the port assignment.
4. In the **Name** field, identify the telephone for your records.
5. In the **Security Code** field, match the password set up on the Remote Office administration.
6. In the **Remote Office Phone** field, type **y**.
7. Press **Enter** to save your changes.

Downloading firmware to a 2420 or 2410 DCP telephone

You can copy updated application code into Communication Manager via TFTP over a TCP/IP connection. This eliminates the need to physically remove the telephone and send it to the factory for the firmware update. This feature is available on all of the servers running Avaya Communication Manager.

Before you start

1. Type **`change node-name ip`**. Press **Enter**.

   The **IP Node Names** screen appears.

2. Administer the TFTP server node name and the local node name (CLAN) and IP address.
3. Press **Enter** to save your changes.
4. Type **`change ip-interfaces`**. Press **Enter**.

   The **IP Interfaces** screen appears.
5. Administer the CLAN Ethernet interface or processor CLAN.
6. Press **Enter** to save your changes.

Downloading the firmware file to Communication Manager

1. Place the file on the TFTP server using TFTP, FTP, HTTP or another file transfer program.
2. From the Web Interface menu, click the **Set LAN Security** link.
3. Click **Advanced**. A list of settings that can be enabled or disabled through the use of check boxes appears.
4. Scroll to **tftp** and check the box enabling *inbound* tftp traffic.
5. Click **Submit**.
6. Log into SAT and enter **`change tftp-server`**. Press **Enter**.

   The **TFTP Server Configuration** screen appears.

Figure 39: TFTP Server Configuration screen

```
change tftp-server                                              Page 1 of 1
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

7. In the **Local Node Name** field, enter the valid local node name from the **IP Node Names** screen.

   The node must be assigned to a CLAN ip-interface or procr (processor CLAN).
8. In the **TFTP Server Node Name** field, enter the valid TFTP server node name from the **IP Nodes Names** screen.
9. In the **TFTP Server Port** field, enter the TFTP server port number from where the file download begins.

10. In the **File to Retrieve** field, enter the name of the file to be retrieved.
11. Press **Enter** to save your changes.

    The file transfer begins.
12. Type **`display tftp-server`**. Press **Enter** to view the status of the file transfer.

    A **File download successful** message appears when the file transfer completes. It also displays the file size and the file name in memory.

Downloading firmware to a single station

You must have console permissions to download someone else's telephones.

Note:

Steps 1 through 3 need be done only once to set up the FAC for file downloads. Thereafter, start at step 4 to download files.

To set up a FAC for file downloads:

1. Type **`change feature-access-codes`**. Press **Enter**.

   Click **Next Page** until you see the **Station Firmware Download Access Code** field on the [Feature Access Code (FAC) screen](#).

Figure 40: Feature Access Code (FAC) screen

```
change feature-access-codes                                     Page 3 of x
                             FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message: *60
           Leave Word Calling Cancel A Message: #60
   Limit Number of Concurrent Calls Activation:         Deactivation:
               Malicious Call Trace Activation:         Deactivation:
         Meet-me Conference Access Code Change:

PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:
                              Posted Messages:

                  Priority Calling Access Code:
                           Program Access Code:

       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:       Deactivation:
              Self Station Display Access Code:
                     Send All Calls Activation:       Deactivation:
         Station Firmware Download Access Code:
```

2. In the **Station Firmware Download Access Code** field, enter a valid FAC as defined in the dial plan.
3. Press **Enter** to save your changes.
4. Take the 2410 or 2420 DCP telephone off-hook.
5. Dial the Station Firmware Download FAC (for instance, *36).
6. Press **#** if you are dialing from the target station (or dial the telephone's extension to be downloaded).
7. Place the telephone on-hook within 4 seconds after the confirmation tone.

   The telephone is placed in a busy-out state (not able to make or receive calls) and displays **Firmware Download in Progress**, the amount of the file downloaded, and a timer. The telephone displays error messages and a success message before rebooting.
8. When the download completes, the telephone reboots and is released from the busy-out state.

Downloading firmware to multiple stations

You can download firmware to multiple stations of the same type, either 2410 or 2420 DCP telephones. Download firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

To download 2420 DCP station firmware to multiple stations:

1. Type `change firmware station-download`. Press **Enter**.

   The **Firmware Station Download** screen appears.

Figure 41: Firmware Station Download screen

```
 change firmware station-download

                              FIRMWARE STATION DOWNLOAD

Source File:

Schedule Download? y
     Start Date/Time:/: Stop Date/Time:/:
Continue Daily Until Completed? y
Download Set Type: 2420

Beginning Station: Ending Station:
```

2. In the **Source File** field, enter the name of the file specified in the **File to Retrieve** field on the **TFTP Server Configuration** screen.

3. In the **Schedule Download** field, type **y**. The **Start Date/Time** and **Stop Date/Time** fields appear.
4. In the **Start Date/Time** field, enter the month (mm), day (dd), year (yyyy), and time (hh:mm) that you want the download to begin.
5. In the **Stop Date/Time** field, enter the month (mm), day (dd), year (yyyy), and time (hh:mm) that you want the download to begin.
6. In the **Continue Daily Until Completed** field, enter **y** if you want the system to execute the firmware download each day at the scheduled time until all specified telephones have received the firmware.
7. In the **Beginning Station** field, enter the first extension number in the range of telephones to which you want to download the firmware. Up to 1000 stations can be included in a scheduled download.
8. In the **Ending Station** field, enter the last extension number in the range of telephones to which you want to download firmware. Up to 1000 stations can be included in a scheduled download.
9. Press **Enter** to save your changes. The firmware download is set to run at the scheduled time. If you entered **n** in the **Schedule Download?** field, pressing **Enter** immediately initiates the download to the specified range of telephones.

Displaying firmware download status

You can use the **`status firmware download`** command to display status information for an active download schedule.

To display download status:

- Type **`status firmware download`**.

  The **Status Firmware Station Download** screen appears.

Figure 42: Status Firmware Download screen

```
 status firmware station-download

                              STATUS FIRMWARE STATION DOWNLOAD

Image file:
Schedule Download?               Continue daily until completed?
Start Date/Time:                 Stop Date/Time:

Terminal type for download:
Extension range:     to:         Number of stations in range:
Stations completed:              Stations unsuccessful:
```

Note:

If you add the qualifier **`last`** to the **`status firmware download`** command, status information on the last download schedule is displayed.

Disabling firmware downloads

You can use the **`disable firmware download`** command to disable any active download schedule.

To disable active downloads:

- Type **`disable firmware download`**.

  This command disables any active download schedule and the system displays **Command successfully completed** at the bottom of the screen.

Chapter 4: Managing Telephone Features

Adding Feature Buttons

Once you add a telephone to the system, you can use the **Station** screen to change the settings for the telephone, such as adding or changing feature button assignments. The system allows you to assign features or functionality to each programmable button. It is up to you to decide which features you want for each telephone and which feature you want to assign to each button.

If you have 6400-series telephones, your users can administer some of their own feature buttons. See Setting up Terminal Self-Administration on page 169 for more information.

Note:

An NI-BRI telephone with Communication Manager has only the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, none of which requires administration. On an NI-BRI telephone, you can assign additional feature buttons only as call appearances. As a result, NI-BRI telephone users must access all other features of Communication Manager using feature access codes.

Additionally, the number of call appearance buttons administered in Communication Manager (the default is three) must match the number of call appearances programmed on the telephone.

Finally, Communication Manager does not support bridged call appearances for NI-BRI telephones.

To assign feature buttons:

1. Type **`change station`** ***`nnnn`***, where ***`nnnn`*** is the extension for the telephone you want to modify. Press **Enter**.
2. Press **Next Page** until you locate the **Button Assignment** section of the **Station screen (page 4)**.

   Some telephones have several feature button groups. Make sure that you are changing the correct button. If you do not know which button on the telephone maps to which button-assignment field, see your telephone's manual, or see Telephone Reference on page 653.

Figure 43: Station screen - button assignments page

```
add station nnnn                                                    Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                         Headset? n
          Jack: ___                                             Speaker? n
         Cable: ___                                            Mounting: d
         Floor: _______                                    Cord Length: 0_
      Building: _______                                       Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                5:limit-call
 2: call-appr                6:team     Ext: 5381231          Rg:
 3: call-appr                7:cfwd-enh Ext:
 4: audix-rec  Ext: 4000     8:cfwd-enh Ext: 5502
```

3. Enter the button name that corresponds to the feature you want to assign a feature button. To determine feature button names, press **Help**, or refer to Telephone Feature Buttons Table on page 134.

   Note:

   For certain newer telephones with expanded text label display capabilities, you can customize feature button labels to accept up to 13 alphanumeric characters. For more information about this feature, see Increasing Text Fields for Feature Buttons on page 131.

4. Press **Enter** to save your changes.

Some telephones have default assignments for buttons. For example, the 8411D includes defaults for 12 softkey buttons. It already has assignments for features like Leave Word Calling and Call Forwarding.

If you do not use an alias, you can easily assign different features to these buttons if you have different needs.

If you use an alias you must leave the default softkey button assignments. The system allows you to change the button assignments on the screen and the features work on the alias telephone, however the labels on the display do not change.

Increasing Text Fields for Feature Buttons

If you are using certain newer phones with expanded text label display capabilities, the Increase Text Fields for Feature Buttons feature allows you to program and store up to 13 character labels for associated feature buttons and call appearances. This feature is available for the following telephones:

- 2410 (Release 2 or newer)
- 2420 (Release 4 or newer)
- 4610 (IP Telephone Release 2.2 or later)
- 4620 (IP Telephone Release 2.2 or later)
- 4621 (IP Telephone Release 2.2 or later)
- 4622 (IP Telephone Release 2.2 or later)
- 4625 (IP Telephone Release 3.1 or later)

Enabling extended text fields for feature buttons

To enable extended text fields for feature buttons:

1. Type **add station next** or **change station *nnnn***, where ***nnnn*** is the extension of the telephone you want to customize feature button labels for. The Station screen appears.

Figure 44: Station screen

```
add station next                                                   Page 1 of X
                              STATION

Extension:                    Lock Messages? n                         BCC: 0
     Type:                    Security Code:                            TN: 1
     Port:                  Coverage Path 1:                           COR: 1
     Name:                  Coverage Path 2:                           COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2            Personalized Ringing Pattern: 3
             Data Module? n                        Message Lamp Ext: 1014
            Speakerphone: 2-way                 Mute button enabled? y
        Display Language? English
                   Model:                             Expansion Module?

 Survivable GK Node Name:                            Media Complex Ext:
          Survivable COR:                                 IP Softphone? y
   Survivable Trunk Dest?                         Remote Office Phone? y
                                                   IP Video Softphone?
                                                             IP Video?

                                                  Customizable Labels? y
```

2. Ensure that **Customizable Labels** is set to **y**. This allows the user to enter 13-character labels for all feature buttons and call appearances associated with this station.
3. Press **Enter** to save your changes.
4. Assign specific feature buttons as described in Adding Feature Buttons on page 129.

   Note:

   You can also use the existing Abbreviated Dialing (AD) button type (**Abr Program**) to program AD labels. However, if you choose to utilize the **Abr Program** button to program AD labels, you are limited to 5 upper case characters. For more information on Abbreviated Dialing, see Adding Abbreviated Dialing Lists on page 157.

Restricting customization of feature button types

In order to manage the usage of your system's allocation of customized button labels to ensure that VIP users have the button label customization resource available to them, you can restrict button label customization of up to 50 specified button types for users who are not considered to be VIP users.

To restrict customization of specific feature button types:

1. Type **`change button-restriction`**. The **Button Type Customization Restrictions** screen appears.

Figure 45: Button Type Customization Restrictions screen

```
 change button-restriction                                    Page 1 of x

                        Button Type Customization Restrictions

Restrict Customization Of Button Types? y

Restrict Customization Of Labels For The Following Button Types:

   abr-spchar   ~p ____________  ____________  ____________  ____________
   abrv-dial    s  ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________
   ____________    ____________  ____________  ____________  ____________

Caution: Adding new button types to this list will cause all button labels
         associated with that button type to be lost
```

2. Ensure that the **Restrict Customization Of Button Types** field is set to **y**.
3. In the fields under **Restrict Customization Of Labels For The Following Button Types**, enter the button type you want to restrict users from customizing.

   Note:

   When you enter the special button types **abr-spchar** or **abrv-dial**, an additional field appears to the right of the button type as shown in Figure 45. Use this special field to specify the special character associated with the **abr-spchar** button type or the Abbreviated Dialing List associated with the **abrv-dial** button type.
4. Press **Enter** to save your changes.

Telephone Feature Buttons Table

The following table provides descriptions of the feature buttons that you can administer on multiappearance telephones. It also lists the administrable software names and recommended button label names. Display buttons support telephones equipped with alphanumeric displays. Note that some buttons might require 1-lamp or 2-lamp buttons. Some buttons are not allowed on some systems and on some telephones.

Note:

An NI-BRI telephone with Communication Manager has only the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, none of which requires administration. On an NI-BRI telephone, you might assign additional feature buttons only as call appearances. As a result, NI-BRI telephone users must access all other features of Communication Manager using feature access codes.

Additionally, the number of call appearance buttons administered in Communication Manager (the default is three) must match the number of call appearances programmed on the telephone.

Finally, Communication Manager does not support bridged call appearances for NI-BRI telephones.

Table 2: Telephone feature buttons

Button name	Button label	Description	Maximum
#	AD	You can administer the # button as an autodial feature button by entering the Audix number in the **BUTTON ASSIGNMENTS** field on the Station screen.	1 per station
abr-prog	Abr Program	Abbreviated Dialing Program: allows users to program abbreviated dialing and autodial buttons or to store or change numbers in a personal list or group list associated with the station.	1 per station
abr-spchar	AbrvDial (char)	Abbreviated Dialing Special Character: allows users to enter an associated special character [~, ~m (mark), ~p (pause), ~s (suppress), ~w (wait for dial tone), or ~W (wait forever)] when programming an abbreviated dialing list entry.	1 each per station
			1 of 23

Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
abrdg-appr (Ext: _____)	(extension)	Bridged Appearance of an analog telephone: allows the user to have an appearance of a single-line telephone extension. Assign to a 2-lamp appearance button.	Depends on station type
abrv-dial (List: __ DC: __)	AD	Abbreviated Dialing: dials the stored number on the specified abbreviated dialing list. List: specify the list number 1 to 3 where the destination number is stored DC: specify the dial code for the destination number	1 per AD list per dial code
abrv-ring	AbRng	Abbreviated and Delayed Ringing: allows the user to trigger an abbreviated or delayed transition for calls alerting at an extension.	
ac-alarm	AC Alarm	Administered Connection alarm notification: allows the user to monitor when the number of failures for an administered connection has met the specified threshold.	1 per station
aca-halt	Auto-Ckt Halt	Automatic Circuit Assurance (*display button*): allows users of display telephones to identify trunk malfunctions. The system automatically initiates a referral call to the telephone when a possible failure occurs. When the user presses ACA Halt, the system turns off ACA monitoring for the entire system. The user must press ACA Halt again to restart monitoring.	1 per system
account	Account	Account: allows users to enter Call Detail Recording (CDR) account codes. CDR account codes allow the system to associate and track calls according to a particular project or account number.	1 per station
admin	Admin	Administration: allows a user to program the feature buttons on their 6400-series telephone.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
after-call Grp:___	AfterCall	After Call Work Mode: allows an agent to temporarily be removed from call distribution in order for the agent to finish ACD-related activities such as completing paperwork. Grp: specify the ACD split group number.	1 per split group
alrt-agchg	Alert Agent	Alert Agent: indicates to the agent that their split/skill hunt group changed while active on a call. This button blinks to notify the agent of the change.	1 per station
alt-frl	Alternate FRL	Alternate Facility Restriction Level (FRL): activates or deactivates an alternate facility restriction level for the extension.	1 per system
ani-requst	ANI Request	Automatic Number Identification Request: allows the user to display the calling party's number from incoming trunks during the voice state of call. The trunk must support this functionality.	1 per station
assist (Group: __)	Assist	Supervisory Assistance: used by an ACD agent to place a call to a split supervisor. Group: specify the ACD split group number.	1 per split group
asvn-halt	ASVN Halt	Authorization Code Security Violation Notification: activates or deactivates call referral when an authorization code security violation is detected.	1 per system
atd-qcalls	AttQueueCall	Attendant Queue Calls (display button): tracks the number of calls in the attendant group's queue and displays the queue status. Assign this button to any user who you want to backup the attendant.	1 per station
			3 of 23

Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
atd-qtime	AttQueueTime	Attendant Queue Time (display button): tracks the calls in the attendant group's queue according to the oldest time a call has been queued, and obtains a display of the queue status.	1 per station
audix-rec	Audix Record	Audix One-Step Recording (display button): activates/deactivates recording of the current call. An Audix hunt group extension that is valid for the user must be entered in the **Ext:** field after the name.	1 per station
aut-msg-wt (Ext: ___)	Msg (name or ext #)	Automatic Message Waiting: associated status lamp automatically lights when an LWC message has been stored in the system for the associated extension (can be a VDN). This lamp will not light on the mapped-to physical station for messages left for virtual extensions.	1 per aut-mst-ext
auto-cback	Auto CallBack	Automatic Call Back: when activated, allows inside user who placed a call to a busy or unanswered telephone to be called back automatically when the called telephone becomes available to receive a call.	1 per station
auto-icom (Group: __)	Autoic (name or ext #)	Automatic Intercom: places a call to the station associated with the button. The called user receives a unique alerting signal, and a status lamp associated with a Intercom button flashes. Grp: Intercom — Auto-Icom group number. This extension and destination extension must be in the same group.	1 per group per dial code
auto-in (Group: __)	Auto In	Auto-In Mode: allows the user to become automatically available for new ACD calls upon completion of an ACD call. Grp: The split group number for ACD.	1 per split group
			4 of 23

Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
auto-wkup	Auto Wakeup	Automatic Wakeup (*display button*): allows attendants, front-desk users, and guests to request a wakeup call to be placed automatically to a certain extension (cannot be a VDN extension) at a later time.	1 per station
autodial	SD	Allows a user to dial a number that is not part of a stored list.	
aux-work (RC: __) (Group: __)	AuxWork	Auxiliary Work Mode: removes agent from ACD call distribution in order to complete non-ACD-related activities. RC: Optional assignment for the 1- or 2-digit Reason Code to be used to change to Aux Work using this button, when Reason Codes is active. Multiple Aux Work buttons, each with a different RC, can be assigned to the same station set. Grp: The split group number for ACD.	1 per split group
brdg-appr (Btn: __ Ext: ___)	(extension)	Bridged Call Appearance: provides an appearance of another user's extension on this telephone. For example, an assistant might have a bridged appearance of their supervisor's extension. The bridged appearance button functions exactly like the original call appearance, for instance it indicates when the appearance is active or ringing. You can assign **brdg-appr** buttons only to 2-lamp appearance buttons. You must indicate which extension and which call appearance button the user wants to monitor at this telephone.	Depends on station type
btn-ring	Button Ring	Station User Button Ring Control: allows users to toggle between audible and silent call alerting.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
btn-view	Button View	Button View: allows users to view, on the telephone's display, the contents of any feature button. Button View does more than the "View" or "stored-num" feature button; these only display what is contained in abbreviated dialing and autodial buttons. When the user presses the **btn-view** button and then a specific feature button, they see the feature name and any auxiliary data for that button. This allows users to review the programming of their feature buttons. You can assign this soft-key button to any 6400-, 7400-, or 8400-series display telephone.	
busy-ind (TAC/Ext: __)	Busy	Busy Indication: indicates the busy or idle status of an extension, trunk group, terminating extension group (TEG), hunt group, or loudspeaker paging zone. Users can press the **busy-ind** button to dial the specified extension. You can assign this button to any lamp button and must specify which Trunk or extension the user wants to monitor.	1 per TAC/ Ext
call-appr	extension	Call Appearance: originates or receives calls. Assign to a 2-lamp appearance button.	Depends on station type
call-disp	Return Call	Call Displayed Number (*display button*): initiates a call to the currently displayed number. The number can be from a leave word calling message or a number the user retrieved from the Directory.	1 per station
call-fwd (Ext: ___)	CFrwd (Ext #) Call Forward (no ext #)	Activates or deactivates Call Forwarding All Calls.	64 per extension
call-park	Call Park	Allows the user to place the current call in the call park state so it can be retrieved from another telephone.	1 per station
call-pkup	Call Pickup	Allows the user to answer a call that is ringing in the user's pickup group.	1 per station
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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
call-timer	Call Timer	Used only on the 6400 sets. Allows users to view the duration of the call associated with the active call appearance button.	1 per station
callr-info	Caller Info	(*display button*) Used with Call Prompting to allow users to display information collected from the originator.	1 per station
cas-backup	CAS Backup	Centralized Attendant Service Backup: used to redirect all CAS calls to a backup extension in the local branch if all RLTs are out-of-service or maintenance busy. The associated status lamp indicates if CAS is in the backup mode.	1 per station
cdr1-alrm	CDR 1 Fail	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the primary CDR output device has occurred.	1 per station
cdr2-alrm	CDR 2 Fail	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the secondary CDR output device has occurred.	1 per station
cfwd-bsyda	CFBDA	Call Forward Busy/Don't Answer: activates and deactivates call forwarding for calls when the extension is busy or the user does not answer.	64 per extension
cfwd-enh	ECFwd (ext #) Enhanced Cfwd (no ext #)	Call Forwarding - Enhanced allows the user to specify the destination extension for both internal and external calls.	
check-in	Check In	Check In (*display button*): changes the state of the associated guest room to occupied and turns off the outward calling restriction for the guest room's station.	1 per station
			7 of 23

Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
check-out	Check Out	Check Out (*display button*): Changes the state of the associated guest room to vacant and turns on the outward calling restriction for the guest room's station. Also clears (removes) any wake-up request for the station.	1 per station
clk-overid	ClkOverride	Clocked Manual Override (*display button*): Used only by authorized attendants and system administrators, in association with Time of Day Routing, to override the routing plan in effect for the system. The override is in effect for a specified period of time. This feature can only be assigned to display telephones.	1 per station
conf-dsp	Conf Display	Allows a user to display information about each party of a conference call. This button can be assigned to stations and attendant consoles.	1 per station
consult	Consult	The Consult button allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation. Activating Consult places the caller on hold and establishes a private connection between the principal and the covering user. The covering user can then add the caller to the conversation, transfer the call to the principal, or return to the caller.	1 per station
cov-cback	CovrCallBack	Allows a covering party to store a leave word calling message for the principal (called party).	1 per station
cov-msg-rt	Covr Msg Ret	Coverage Message Retrieval (*display button*): places a covering station into the message retrieval mode for the purposes of retrieving messages for the group.	1 per station
cpn-blk	CPN Block	Blocks the sending of the calling party number for a call.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
cpn-unblk	CPN Unblock	Deactivates calling party number (CPN) blocking and allows the CPN to be sent for a single call.	1 per station
crss-alert	Crisis Alert	Crisis Alert (*display button*): provide this button to the telephones or consoles that you want to notify when any user makes an emergency call. (You define which calls are emergency calls on the **AAR/ARS Analysis** screen by setting the **Call Type** to **alrt**.) After a user receives an alert, they can press the **crss-alert** button to disable the current alert. If tenant partitioning is active, the attendants within a partition can receive emergency notification only from callers in the same partition.	1 per station 10 per system
data-ext	Data (data ext #)	Data Extension: sets up a data call. Can be used to pre-indicate a data call or to disconnect a data call. Cannot be a VDN or ISDN-BRI extension.	1 per data-extension group
date-time	Time/Date	Date and Time (*display button*): displays the current date and time. Do not assign this button to 6400-series display telephones as they normally show the date and time.	1 per station
delete-msg	Delete Msg	Delete message (*display button*): deletes a stored LWC message or wakeup request.	1 per station
dial-icom (Grp: ___)	Dial Icom	Dial Intercom: accesses the intercom group assigned to the button. Grp: Intercom — Dial (Dial Icom) group number.	1 per group
did-remove	DID Remove	DID Remove (*display button*): allows DID assignments to be removed.	1 per station
did-view	DID View	DID View (*display button*): allows DID assignments to be displayed and changed. Allows choice between XDID and XDIDVIP numbers.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
directory	Directory	Directory (*display button*): allows users with display telephones to access the integrated directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory. The directory contains the names and extensions that you have assigned to the telephones administered in your system. If you assign a directory button, you should also assign a **Next** and **Call-Disp** button to the telephone. These buttons allow the user to navigate within the integrated directory and call an extension once they find the correct one. **Note:** Vector Directory Numbers do not appear in the integrated directory. Also, if you assign a name beginning with two tildes (~~} to a telephone, and **Display Character Set** on the **System Parameters Country-Options** screen is set to **Roman**, the name does not appear in the integrated directory. Note that this is the only way to hide a name in the integrated directory.	1 per station
dir-pkup	Dir Pickup	Directed call pickup: allows the user to answer a call ringing at another extension without having to be a member of a pickup group.	
disp-chrg	Disp Charges	Provides your display telephone with a visual display of accumulated charges on your current telephone call. Used exclusively outside the U.S. and Canada.	1 per station
disp-norm	Local/ Normal	Normal (*display button*): Toggles between LOCAL display mode (displays time and date) and NORMAL mode (displays call-related data). LED off = LOCAL mode and LED on = NORMAL.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
dn-dst	DoNotDisturb	Places the user in the do not disturb mode.	1 per station
drop	Drop	Allows users to drop calls. Users can drop calls from automatic hold or drop the last party they added to a conference call.	
ec500	EC500	Administers an Extension to Cellular feature button on the office telephone. When you enter this value, the **Timer** subfield displays, and defaults to **n**. Set the optional Timer subfield to **y** to include an Extension to Cellular timer state for the administered feature button. When the timer state is included, the Extension to Cellular user can activate a one-hour timer to temporarily disable Extension to Cellular through this administered feature button. Leaving the default setting of n excludes the timer state.	1 per station
exclusion	Exclusion	Exclusion: allows multiappearance telephone users to keep other users with appearances of the same extension from bridging onto an existing call. If the user presses the **Exclusion** button while other users are already bridged onto the call, the other users are dropped. There are two means of activating exclusion. ● Manual Exclusion — when the user presses the **Exclusion** button (either before dialing or during the call). ● Automatic Exclusion — as soon as the user picks up the handset. To turn off Automatic Exclusion during a call, the user presses the **Exclusion** button. To use Automatic Exclusion, set the **Automatic Exclusion by COS** field to **y** on the **Feature-Related System Parameters** screen.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
ext-dn-dst	ExtDoNotDisturb	**Extension — Do Not Disturb** (*display button*): used by the attendant console or hotel front desk display telephone to activate do not disturb and assign a corresponding deactivate time to an extension.	1 per station
extnd-call	Extend Call	Allows the user to extend the current call to an Off-PBX/EC500 telephone.	1 per station
fe-mute	Far End Mute	Allows a user to mute a selected party on a conference call. This button can be assigned to stations and attendant consoles.	1 per station
flash	Flash	1) Allows a station on a trunk call with Trunk Flash to send a Trunk Flash signal to the far end (e.g., Central Office); 2) allows a station on a CAS main call to send a Trunk Flash signal over the connected RLT trunk back to the branch to conference or transfer the call.	1 per station
goto-cover	Goto Cover	Go To Coverage: sends a call directly to coverage instead of waiting for the called inside-user to answer. Go to Cover forces intercom and priority calls to follow a coverage path. **Note:** Go to Cover cannot be activated for calls placed to a Vector Directory Number extension. Go to Cover can be used to force a call to cover to a VDN if the called principal has a VDN as a coverage point.	1 per station
grp-dn-dst	GrpDoNotDstrb	Group Do Not Disturb (*display button*): places a group of telephones into the do not disturb mode.	1 per station
grp-page (Number:___)	GrpPg	Allows users to make announcements to groups of stations by automatically turning on their speakerphones. Number: The extension of the page group.	
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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
headset	Headset	Signals onhook/offhook state changes to Communication Manager. The green LED is on for offhook state and off (dark) for onhook state.	1 per station
hunt-ns (Grp: ___)	HuntNS	Hunt-Group Night Service: places a hunt-group into night service. Grp: Hunt group number.	3 per hunt group
in-call-id (Type: __ Grp: ___)	INCallID (group #, type, name, or ext #)	The Coverage Incoming Call Identification (ICI) button allows a member of a coverage answer group or hunt group to identify an incoming call to that group even though the member does not have a display telephone. In the **Type** field, enter **c** for coverage answer groups and type of **h** for a hunt group. In the **Grp** field, enter the group number.	1 per group-type per group
inspect	Inspect	Inspect (*display button*): allows users on an active call to display the identification of an incoming call. Inspect also allows users to determine the identification of calls they placed on Hold.	1 per station
int-aut-an	IntAutoAnswer	Internal Automatic Answer: causes any hybrid or digital station to automatically answer incoming internal calls.	1 per station
last-numb	LastNumb Dialed	Last Number Dialed (redial): originates a call to the number last dialed by the station.	1 per station
lic-error	License Error	License-Error: indicates a major License File alarm. Pressing the button does not make the light go out. The button goes out only after the error is cleared and Communication Manager returns to License-Normal Mode. You can administer this button on telephones and attendant consoles.	1 per telephone 20 per system (Server CSI)

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
limit-call	LimitInCalls	Limit Number of Concurrent Calls feature: allows user to limit the number of concurrent calls at a station to one call, where normally multiple call appearances can terminate at the station.	1 per station
link-alarm (link# ___)	Link Fail (link #)	Link Alarm: associated status lamp indicates that a failure has occurred on one of the Processor Interface circuit pack data links. Link: Link number — 1 to 8 for multi-carrier cabinets or 1 to 4 for single-carrier cabinets.	8 per station
lsvn-halt	LSVN Halt	Login Security Violation Notification: activates or deactivates referral call when a login security violation is detected.	1 per system
lwc-cancel	Cancel LWC	Leave Word Calling Cancel: cancels the last leave word calling message originated by the user.	1 per station
lwc-lock	Lock LWC	Leave Word Calling Lock: locks the message retrieval capability of the display module on the station.	1 per station
lwc-store	Store LWC	Leave Word Calling Store: leaves a message for the user associated with the last number dialed to return the call to the originator.	1 per station
major-alrm	Major Alarm	Major Alarm: assign to a status lamp to notify the user when major alarms occur. Major alarms usually require immediate attention.	1 per station
man-msg-wt (Ext: ___)	Msg Wait (name or ext #)	Manual Message Waiting: allows a multiappearance telephone user to press a button on their telephone in order to light the Manual Message Waiting button at another telephone. You can administer this feature only to pairs of telephones, such as an assistant and an executive. For example, an assistant can press the **man-msg-wt** button to signal the executive that they have a call.	None

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
man-overid (TOD: _)	ManOverid	Immediate Manual Override (*display button*): allows the user (on a system with Time of Day Routing) to temporarily override the routing plan and use the specified TOD routing plan. TOD: specify the routing plan the user wants to follow in override situations.	1 per station
manual-in (Group: __)	Manual In	Manual-In Mode: prevents the user from becoming available for new ACD calls upon completion of an ACD call by automatically placing the agent in the after call work mode. Grp: The split group number for ACD.	1 per split group
mct-act	MCT Activate	Malicious Call Trace Activation: sends a message to the MCT control extensions that the user wants to trace a malicious call. MCT activation also starts recording the call, if your system has a MCT voice recorder.	1 per station
mct-contr	MCT Control	Malicious Call Trace Control: allows the user to take control of a malicious call trace request. Once the user becomes the MCT controller, the system stops notifying other MCT control extensions of the MCT request. NOTE: To add an extension to the MCT control group, you must also add the extension on the **Extensions Administered to have an MCT-Control Button** screen. When the user presses the MCT Control button, the system first displays the called party information. Pressing the button again displays the rest of the trace information. The MCT controller must dial the MCT Deactivate feature access code to release control.	1 per station
mf-da-intl	Directory Assistance	Multifrequency Operator International: allows users to call Directory Assistance.	1 per station
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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
mf-op-intl	CO attendant	Multifrequency Operator International: allows users to make international calls to the CO attendant.	1 per station
mj/mn-alrm	Mj/Mn Alarm	Minor Alarm: assign to a status lamp to notify the user when minor or major alarms occur. Minor alarms usually indicate that only a few trunks or a few stations are affected.	1 per station
mm-basic	MM Basic	Multimedia Basic: used to place a multimedia complex into the "Basic" mode or to return it to the "Enhanced" mode.	1 per station
mm-call	MM Call	Multimedia Call: used to indicate a call is to be a multimedia call.	1 per station
mm-cfwd	MM Call Fwd	Multimedia Call Forward: used to activate forwarding of multimedia calls as multimedia calls, not as voice calls.	1 per station
mm-datacnf	MM Data Cnf	Multimedia Data Conference: used to initiate a data collaboration session between multimedia endpoints; requires a button with a lamp.	1 per station
mm-multnbr	MM Mult Nbr	Indicate that the user wants to place calls to 2 different addresses using the 2 B-channels.	1 per station
mm-pcaudio	MM PC Audio	Switches the audio path from the telephone (handset or speakerphone) to the PC (headset or speakers/ microphone).	1 per station
msg-retr	Msg Retrieve	Message Retrieval (*display button*): places the station's display into the message retrieval mode.	1 per station
mwn-act	MsgWaitAct	Message Waiting Activation: lights a message waiting lamp on an associated station.	1 per station
mwn-deact	MsgWaitDeact	Message Waiting Deactivation: dims a message waiting lamp on an associated station.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
next	Next	Next (*display button*): steps to the next message when the telephone's display is in Message Retrieval or Coverage Message Retrieval mode. Shows the next name when the telephone's display is in the Directory mode.	1 per station
night-serv	Night Service	Night Service Activation: toggles the system in or out of Night Service mode.	1 per station
noans-alrt	NoAnsAlrt	Redirection on No Answer Alert: indicates a Redirection on No Answer timeout has occurred for the split.	1 per hunt group
no-hld-cnf	No Hold Conf	No Hold Conference: can automatically conference another party while continuing the existing call.	1 per station
normal	Normal Mode	Normal (*display button*): places the station's display into normal call identification mode.	1 per station
off-bd-alm	OffBoardAlarm	Off board Alarm: associated status lamp lights if an off-circuit pack major, minor, or warning alarm is active on a circuit pack. Off-board alarms (loss of signal, slips, misframes) relate to problems on the facility side of the DS1, ATM, or other interface.	1 per attendant
per-COline (Grp: ___)	COLine (line #)	Personal CO Line: allows the user to receive calls directly via a specific trunk. Grp: CO line group number.	1 per group
pms-alarm	PMS Failure	Property Management System alarm: associated status lamp indicates that a failure in the PMS link occurred. A major or minor alarm condition raises the alarm.	1 per station
post-msgs	Posted MSGs	Posted Messages: Allows the user to display a specific message to callers.	1 per station
pr-awu-alm	AutoWakeAlarm	Automatic Wakeup Printer Alarm: associated status lamp indicates that an automatic wakeup printer interface failure occurred.	1 per station
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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
pr-pms-alm	PMS Ptr Alarm	PMS Printer Alarm: associated status lamp indicates that a PMS printer interface failure occurred.	1 per station
pr-sys-alm	Sys Ptr Alarm	System Printer Alarm: associated status lamp indicates that a system printer failure occurred.	1 per station
print-msgs	Print Msgs	Print Messages: allows users to print messages for any extension by pressing the button and entering the extension and a security code.	1 per station
priority	Priority Call	Priority Calling: allows a user to place priority calls or change an existing call to a priority call.	1 per station
q-calls (Grp: ___)	QueueCall	Queue Calls: associated status lamp flashes if a call warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
q-time (Grp: ___)	QueueTime	Queue Time: associated status lamp flashes if a time warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
release	Release	Releases an agent from an ACD call.	1 per station
ring-stat	Ringer Status	Users can display the ringer status for a line or bridged appearance by pressing the **ring-stat** button followed by a **call-appr**, **brdg-appr** or **abrdg-appr** button. Depending on the ringer status, the display shows: • **Ringer On** • **Ringer Off** • **Ringer Delayed** • **Ringer Abbreviated**	1 per station
ringer-off	Ringer Off	Ringer-Cutoff: silences the alerting ringer on the station.	1 per station
rs-alert	ResetAlert	The associated status lamp lights if a problem escalates beyond a warm start.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
rsvn-halt	RSVN Halt	Remote Access Barrier Code Security Violation Notification Call: activates or deactivates call referral when a remote access barrier code security violation is detected.	1 per system
scroll	Scroll	Scroll (*display button*): allows the user to select one of the two lines (alternates with each press) of the 16-character LCD display. Only one line displays at a time.	1 per station
send-calls (Ext: ___)	SAC (ext #)	Send All Calls allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Assign to a lamp button.	64 per extension
send-term	Send TEG	Send All Calls For Terminating Extension Group: allows the user to forward all calls directed to a terminating extension group.	1 per TEG
serv-obsrv	Service Obsrv	Service Observing: activates Service Observing. Used to toggle between a listen-only and a listen-talk mode.	1 per station
share-talk	Share Talk	Share Talk: enables multiple DCP or H323 IP endpoints that are registered to the same extension to share talk capability. Normally, when more than one endpoint requests RTP (Real Time Transfer Protocol) media, only one of the endpoints (Base Set) is capable of talking and listening, while the other endpoints are connected in listen-only mode. This button allows all the endpoints that are associated with the extension to share the talk capability. Note that in Communication Manager 5.0, only AE Server DMCC (Device, Media, and Call Control) endpoints are capable of requesting RTP while they are sharing control of the extension. For more information on DMCC, see *Avaya MultiVantage® Application Enablement Services Administration and Maintenance Guide*, 02-300357.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
signal (Ext: ___)	Sgnl (name or ext #)	Signal: allows the user to use one button to manually signal the associated extension. The extension cannot be a VDN extension.	1 per signal extension
ssvn-halt	SSVN Halt	Toggle whether or not station security code violation referrals are made to the referral destination.	1 per station
sta-lock	Station Lock	When Station Lock is enabled, the only calls that can be made from the station are those allowed by the COR administered in the **Station Lock COR** field.	1 per station
start-bill	Start Bill	After an ACD agent answers a call, the agent can press this button to send an ISDN CONNECT message to the PSTN network to start the PSTN call–billing for a call at the PSTN switch.	1 per station
stored-num	Stored Number	Enables a display mode that displays the numbers stored in buttons.	1 per station
stroke-cnt (Code:_)	Stroke Count (#)	Automatic Call Distribution Stroke Count # (0, 1, 2, 3, 4, 5, 6, 7, 8, or 9) sends a message to CMS to increment a stroke count number.	up to 10 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
team	Team	The Team Button has two generic functions, a display function and an execution function. The display function allows any member of a team (monitoring station) to observe the station state of other team members (monitored station). As an execution function, the Team Button can be used as Speed Dial Button or Pick-Up Button where a call to the monitored station is established directly or a ringing call is picked from the monitored station. **Ext:** This field appears when you enter the button type **team**. Enter the extension of the principal station of the virtual "team." **Rg** This field appears when you enter the button type **team**. Enter the kind of audible ringing for the team button. Valid entries are **a(bbreviated)**, **d(elayed)**, **n(o-ring)**, and **r(ing)**.	15 per monitoring station
term-x-gr (Grp: ___)	TermGroup (name or ext #)	Terminating Extension Group: provides one or more extensions. Calls can be received but not originated with this button. Grp: TEG number.	1 per TEG
timer	Timer	Used only on the 6400 sets. Allows users to view the duration of the call associated with the active call appearance button.	1 per station
togle-swap	Toggle-Swap	Allows a user to toggle between two parties before completing a conference or a transfer.	1 per station
trk-ac-alm	FTC Alarm	Facility Test Call Alarm: associated status lamp lights when a successful Facility Test Call (FTC) occurs.	1 per station
trk-id	Trunk ID	Trunk Identification (*display button*): identifies the tac (trunk access code) and trunk member number associated with a call.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
trunk-name	Trunk Name	(*display button*) Displays the name of the trunk as administered on the CAS Main or on a server without CAS.	1 per station
trunk-ns (Grp: ___)	Trunk NS	Trunk-Group Night Service: places a trunk-group into night service. Grp: Trunk group number.	3 per trunk group
usr-addbsy	Add Busy Indicator	Adds the busy indicator.	1 per station
usr-rembsy	Remove Busy Indicator	Removes the busy indicator.	1 per station
uui-info	UUI-Info	Allows users to see up to 32 bytes of ASAI-related UUI-IE data.	1 per station
verify	Verify	Busy Verification: allows users to make test calls and verify a station or a trunk.	1 per station
vip-chkin	VIP Check In	VIP Check-in (*display button*): allows user to assign the XDIDVIP number to the room extension.	1 per station
vip-retry	VIP Retry	VIP Retry: starts to flash when the user places a VIP wakeup call and continues to flash until the call is answered. If the VIP wakeup call is not answered, the user can press the VIP Retry button to drop the call and reschedule the VIP wakeup call as a classic wakeup call. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station
vip-wakeup	VIP Wakeup	VIP Wakeup: flashes when a VIP wakeup reminder call is generated. The user presses the button to place a priority (VIP) wakeup call to a guest. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station
voa-repeat	VOA Repeat	VDN of Origin Announcement. VDN of Origin Announcement must be enabled.	1 per station

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Table 2: Telephone feature buttons (continued)

Button name	Button label	Description	Maximum
voice-mail	Message	This is not an administrable button, but maps to the fixed hard "message" button on newer telephones.	1 per station
vu-display (format: __ ID: __)	Vu Display #	VuStats Display: allows the agent to specify a display format for the statistics. If you assign a different VuStats display format to each button, the agent can use the buttons to access different statistics. You can assign this button only to display telephones. format: specify the number of the format you want the button to display ID (optional): specify a split number, trunk group number, agent extension, or VDN extension	limited to the number of feature buttons on the telephone
whisp-act	WhisperAct	Whisper Page Activation: allows a user to make and receive whisper pages. A whisper page is an announcement sent to another extension who is active on a call where only the person on the extension hears the announcement; any other parties on the call cannot hear the announcement. The user must have a class of restriction (COR) that allows intra-switch calling to use whisper paging.	1 per station
whisp-anbk	WhisperAnbk	Whisper Page Answerback: allows a user who received a whisper page to respond to the user who sent the page.	1 per station
whisp-off	WhisperOff	Deactivate Whisper Paging: blocks other users from sending whisper pages to this telephone.	1 per station
work-code	Work Code	Call Work Code: allows an ACD agent after pressing "work-code" to send up to 16 digits (using the dial pad) to CMS.	1 per station
			23 of 23

Adding Abbreviated Dialing Lists

Abbreviated dialing is sometimes called speed dialing. It allows you to dial a short code in place of an extension or telephone number.

When you dial abbreviated-dialing codes or press abbreviated-dialing buttons, you access stored numbers from special lists. These lists can be personal (a list of numbers for an individual telephone), group (a department-wide list), system (a system-wide list), or enhanced numbers (allows for a longer list of numbers). The version and type of your system determine which lists are available and how many entries you can have on each list.

Note:

You can designate all group-number lists, system-number lists, and enhanced-number lists as "privileged." Calls automatically dialed from a privileged list are completed without class of restriction (COR) or facility restriction level (FRL) checking. This allows access to selected numbers that some telephone users might otherwise be restricted from manually dialing. For example, a user might be restricted from making long-distance calls. However, you can program the number of a branch office that is long distance into an AD list as privileged. Then, the user can call this office location using AD, while still being restricted from making other long-distance calls.

SECURITY ALERT:

Privileged group-number, system-number, and enhanced-number lists provide access to numbers that typically would be restricted.

Instructions

As an example, we will program a new group list:

1. Type **`add abbreviated-dialing group next`**. Press **Enter**.

   The **Abbreviated Dialing List** screen appears. In our example, the next available group list is group 3.

Figure 46: Abbreviated Dialing List screen

```
change abbreviated-dialing group                                    Page 1 of X
                              ABBREVIATED DIALING LIST
                                 Group List: ____
      Size (multiple of 5):   5  Program Ext: ________   Privileged? n
   DIAL CODE
        01: ________________
        02: ________________
        03: ________________
        04: ________________
        05: ________________
```

2. Enter a number (in multiples of 5) in the **Size** field. This number defines the number of entries on your dialing list.

   For example, if you have 8 telephone numbers you want to store in the list, type **10** in the **Size** field.

3. If you want another user to be able to add numbers to this list, enter their extension in the **Program Ext** field.

   For example, if you want the user at 4567 to be able to change group list 3, enter **4567** in this field.

4. Enter the telephone numbers you want to store, one for each dial code.

   Each telephone number can be up to 24 digits long.

5. Press **Enter** to save your changes.

You can display your new abbreviated-dialing list to verify that the information is correct or print a copy of the list for your paper records.

Once you define a group list, you need to define which stations can use the list. For example, we will set up station 4567 so it has access to the new group list.

To give station 4567 access to the group list:

1. Type **`change station 4567`**. Press **Enter**.
2. Press **Next Page** until you see Station screen (page 4)**,** containing the Abbreviated Dialing List fields.

Figure 47: Station screen

```
add station nnnn                                                  Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                        Headset? n
          Jack: ___                                            Speaker? n
         Cable: ___                                           Mounting: d
         Floor: _______                                    Cord Length: 0_
      Building: _______                                      Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                  6:limit-call
 2: call-appr                  7:team     Ext: 5381231          Rg:
 3: call-appr                  8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000       9:cfwd-enh Ext: 5502
 5: release                   10:aux-work RC: 1 Group:

voice-mail Number:
```

3. Type **group** in any of the **List** fields. Press **Enter**.

   A blank list number field appears.

4. Type **3** in the list number field.

   When you assign a group or personal list, you must also specify the personal list number or group list number.

5. Press **Enter** to save your changes.

The user at extension 4567 can now use this list by dialing the feature access code for the list and the dial code for the number they want to dial. Alternatively, you can assign an abbreviated dialing button to this station that allows the user press one button to dial a specific stored number on one of their three assigned abbreviated lists.

Fixing problems

Problem	Possible causes	Solutions
A user cannot access a dial list	• The specific list might not be assigned to the user's telephone.	Resolution: 1. Type **`display station nnnn`**, where ***nnnn*** is the user's extension. Press **Enter**. 2. Review the current settings of the **List1**, **List2**, and **List3** fields to determine if the list the user wants to access is assigned to their telephone.
	• If the user attempted to use a feature access code to access the list, they might have dialed the incorrect feature access code.	Resolution: 1. Type **`display feature-access-codes`**. Press **Enter**. 2. Verify that the user is dialing the appropriate feature access code.
	• If the user attempted to press a feature button, they might have pressed the incorrect feature button.	Resolution: 1. Type **`display station nnnn`**, where ***nnnn*** is the user's extension. Press **Enter**. 2. Review the current feature button assignments to determine if the user was pressing the assigned button.
		1 of 2

Problem	Possible causes	Solutions
A user cannot access a dial list - *continued*	• If the user attempted to press the correct feature button, the button might not be set up correctly.	Resolution: 1. Type **`display station nnnn`**, where ***`nnnn`*** is the user's extension. Press **Enter**. 2. Review the current feature button assignments to see if the list number and dial code are correct.
A user complains that using an abbreviated dial list dials the wrong number.	• The user could be using the wrong dial code. • The dial code could be defined incorrectly.	Resolution: 1. Ask the user what number they dialed or button they pressed to determine which list and dial code they attempted to call. 2. Access the dialing list and verify that the number stored for the specific dial code corresponds to the number the user wanted to dial. (For example to access a group list, type **`display abbreviated-dialing group`** ***`x`***. Press **Enter**, where ***`x`*** is a group list number.) 3. If the user dialed the wrong code, give them the correct code. 4. If the dial code is wrong, press **Cancel** and use the appropriate change command to re-access the abbreviated dialing list. Correct the number. Press **Enter**.
		2 of 2

More information

There are limits to the total number of abbreviated dialing list entries, the number of personal dial lists, and the number of group dial lists that your system can store. Because of these limitations, you should avoid storing the same number in more than one list. Instead, assign commonly dialed numbers to the system list or to a group list. You can determine the abbreviated dialing storage capacity, by referring to the **System Capacity** screen for the abbreviated dialing values (type **`display capacity`**). For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Related topics

For more information, see "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Setting up Bridged Call Appearances

Think of a bridged call appearance as a telephone (the primary set) with an extension (the bridged-to appearance). Both telephones can be used to call in and out and both show when a line is in use. A call to the primary telephone is bridged to a specific appearance, or button, on the secondary telephone. The secondary telephone retains all its functions, and a specific button is dedicated as the bridged-to appearance from the primary telephone.

Bridged call appearances have to be assigned to telephones with double-lamp buttons, or lights. The telephone types do not need to match, but as much consistency as possible is recommended for all telephones in a bridged group. When a call comes in on bridged telephones, the buttons assigned to the bridged appearances flash. You can assign as many bridged appearances as there are line appearances on the primary telephone, and you can assign ringing (alerting) to one or more of the telephones.

Instructions

To create a bridged call appearance:

1. Note the extension of the primary telephone.

   A call to this telephone lights the button and, if activated, rings at the bridged-to appearance on the secondary telephone.

2. If you want to use a new telephone for the bridged-to extension, duplicate a station.

   For information, see Duplicating telephones on page 88.

3. Type **`change station`** and the bridged-to extension. Press **Enter**.
4. Press **Next Page** until the **Feature Options** page of the **Station screen** appears.

5. **For the Per Button Ring Control** field (digital sets only):
   - If you want to assign ringing separately to each bridged appearance, type **y**.
   - If you want all bridged appearances to either ring or not ring, leave the default **n**.

Figure 48: Station screen

```
change station nnnn                                               Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
             CDR Privacy? n                            Data Restriction? n
   Redirect Notification? y                       Call Waiting Indication:
 Per Button Ring Control? n                 Attd. Call Waiting Indication:
   Bridged Call Alerting? n                    Idle Appearance Preference? n
        Switchhook Flash? n                  Bridged Idle Line Preference? y
    Ignore Rotary Digits? n                      Restrict Last Appearance? y
  Active Station Ringing: single        Conf/Trans On Primary Appearance? n
                                                       EMU Login Allowed?
        H.320 Conversion? n         Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed      Busy Auto Callback without Flash? y
         Multimedia Mode: basic
    MWI Served User Type: ______               Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                     Select Last Used Appearance? n
                               _                Coverage After Forwarding? _
     Recall Rotary Digit? n                       Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n        IP Audio Hairpinning? n
Precedence Call Waiting? y
```

6. Move to **Bridge Call Alerting**.

   If you want the bridged appearance to ring when a call arrives at the primary telephone, type **y**. Otherwise, leave the default **n**.

7. Complete the appropriate field for your telephone type.

If. . .	Then. . .
your primary telephone is analog	move to the **Line Appearance** field and enter **abrdg-appr**
your primary telephone is digital	move to the **BUTTON ASSIGNMENTS** field and enter **brdg-appr**

8. Press **Enter**.

   Btn and **Ext** fields appear. If **Per Button Ring Control** is set to **y** on the **Station** screen for the digital set, **Btn**, **Ext**, and **Ring** fields appear.

Figure 49: Station screen (analog set)

```
 add station nnnn                                                Page 3 of X
                                   STATION

     SITE DATA
          Room: _______                                    Headset? n
          Jack: ___                                        Speaker? n
         Cable: ___                                       Mounting: d
         Floor: _______                               Cord Length: 0_
      Building: _______                                 Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                 5:
 2: call-appr                 6:
 3: call-appr                 7:
 4: audix-rec  Ext: 4000      8:
```

Figure 50: Station screen (digital set)

```
add station nnnn                                                 Page 3 of X
                                   STATION

     SITE DATA
          Room: _______                                    Headset? n
          Jack: ___                                        Speaker? n
         Cable: ___                                       Mounting: d
         Floor: _______                               Cord Length: 0_
      Building: _______                                 Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                 5:
 2: call-appr                 6:
 3: call-appr                 7:
 4: audix-rec  Ext: 4000      8:
```

9. Enter the primary telephone's button number that you want to assign as the bridged call appearance.

   This button flashes when a call arrives at the primary telephone.

10. Enter the primary telephone extension.
11. If the **Ring** field appears:
    - If you want the bridged appearance to ring when a call arrives at the primary telephone, type **y**.
    - If you do not want the bridged appearance to ring, leave the default **n**.
12. Press **Enter** to save your changes.

To see if an extension has any bridged call appearances assigned, type **`list bridge`** and the extension. Press **Enter**.

More information

Following are a list of example situations where you might want to use bridged appearances.

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- Visitor telephones

  An executive might have another telephone in their office that is to be used by visitors. It might be desirable that the visitor be able to bridge onto a call that is active on the executive's primary extension number. A bridged call appearance makes this possible.

- Service environments

  It might be necessary that several people be able to handle calls to a particular extension number. For example, several users might be required to answer calls to a hot line number in addition to their normal functions. Each user might also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.

- A user frequently using telephones in different locations

  A user might not spend all of their time in the same place. For this type of user, it is convenient to have their extension number bridged at several different telephones.

Setting up Extension to Cellular

Use the Extension to Cellular feature to extend your office calls and Communication Manager features to a cellular telephone. For a detailed description of the Extension to Cellular feature and how to administer it, see "Extension to Cellular" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, or *Avaya Extension to Cellular User's Guide*, 210-100-700.

The following table provides a quick reference to the screens and fields used in administering the Extension to Cellular feature.

Table 3: Screens for administering Extension to Cellular

Screen name	Purpose	Fields
Stations with Off-PBX Telephone Integration	Map station extensions to application types and external telephone numbers.	All
Off-PBX Telephone Mobile-Feature-Extension	Administer CTI feature.	**Mobile Call (CTI) Extension**
Feature Access Code (FAC)	Set up access codes for Communication Manager features.	**Feature Access Code**
Extension to Call Which Activate Features by Name	Map a dialed extension to activate a feature (FNE) within Communication Manager from a cell phone. Some FNEs require FAC administration.	**Extension**
Telecommuting Access	Create an Extension to Cellular remote access number.	All
Security-Related System Parameters	Define a system-wide station security code length.	**Minimum Station Security Code Length**
Station	Assign feature buttons and timers.	**BUTTON ASSIGNMENTS**
Language Translations	To review the office telephone feature button assignments.	All
Numbering-Public/ Unknown Format	Assign 10-digit caller identification.	All
		1 of 2

Table 3: Screens for administering Extension to Cellular (continued)

Screen name	Purpose	Fields
Coverage Path	Set up number of unanswered rings prior to coverage.	**Number of Rings**
Trunk Group	Enable Call Detail Recording for outgoing trunk.	**CDR Reports**
DS1 Circuit Pack	Administer a DS1 Circuit pack for R2MFC for EC500 use.	**Signaling Mode: CAS** **Interconnect: CO**
Trunk Group	Administer a trunk group for EC500 use. **Note**: For more information, see "Extension to Cellular" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.	**Group Type** **Trunk Type** **Trunk Type** **Outgoing Dial Type** **Incoming Dial Type** **Receive Answer Supervision?**
Multifrequency-signaling-related-parameters	Administer MFC parameters needed for EC500. **Note**: For more information, see "Guidelines for administering Multifrequency Signaling" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.	**Incoming Call Type: group-ii-mfc** (for MFC signaling) **Outgoing Call Type: group-ii-mfc** (for MFC signaling) **Request Incoming ANI (non-AR/ARS)? y**
System Capacity	Verify used, available, and system station limits.	**Off-PBX Telephone - EC500** **Off-PBX Telephone - OPS** **Off-PBX Telephone - PBFMC** **Off-PBX Telephone - PVFMC**
		2 of 2

Setting up an Extension to Cellular Feature Access Button

Extension to Cellular provides the capability to administer an Extension to Cellular feature access button on the user's office telephone to enable and disable the feature. You can also configure an optional timer. You administer this feature button on page 3 of the **Station** screen for the "host" office extension to which Extension to Cellular is linked.

The process described below explains how to administer an Extension to Cellular feature button and include the optional Extension to Cellular timer. The Extension to Cellular feature button is available on telephones which support administrable feature buttons.

To set up an Extension to Cellular feature button with optional timer:

1. Type **`change station n`** (where ***n*** is the extension of an Extension to Cellular-enabled station - in this example, `1034`). Press **Enter**.
2. Press the **Next Page** button twice to display the **Station screen (page 4)**.

Figure 51: Station screen - page 3

```
add station nnnn                                                      Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                           Headset? n
          Jack: ___                                               Speaker? n
         Cable: ___                                              Mounting: d
         Floor: _______                                       Cord Length: 0_
      Building: _______                                         Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                6:limit-call
 2: call-appr                7:team     Ext: 5381231          Rg:
 3: call-appr                8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000     9:cfwd-enh Ext: 5502
 5: release                 10:aux-work RC: 1 Group:

voice-mail Number:
```

3. Select an available feature button under the **BUTTON ASSIGNMENTS** header (button **4** was used in this example) and type **ec500** to administer an Extension to Cellular feature button on the office telephone.
4. Press **Enter**.

   Note:

   The **Timer** subfield displays, and defaults to **n**. Leaving the default setting of **n** excludes the timer state.

5. Set the optional **Timer** subfield to **y** to include an Extension to Cellular timer state for the administered feature button.

   When the timer state is included, the Extension to Cellular user can activate a one-hour timer to temporarily disable Extension to Cellular through this administered feature button.

6. Press **Enter**.

   The corresponding feature button on the office telephone is now administered for Extension to Cellular.

   Note:

   The feature status button on the office telephone indicates the current state of Extension to Cellular regardless of whether the feature was enabled remotely or directly from the office telephone.

For additional information, see the *Avaya Extension to Cellular User's Guide*, 210-100-700*.*

Setting up Terminal Self-Administration

Terminal self-administration (TSA) allows users to administer some of their own feature buttons from their telephones. TSA is available for 6400-series, and 4612 and 4624 telephones. Users are prompted, via the telephone's display, to choose features to assign to buttons on their telephones.

Before you start

To prevent users from changing another user's telephone administration, you can enable the system-wide option that requires users to enter a station security code before they can administer their telephone. To enable this option:

1. Set the **Station Security Code for Terminal Self-Administration Required** on the **Security-Related System Parameters** screen to **y**.

   If you enable this option, the user is prompted for the station security code when they press the **Admin** button. The user must enter the security code, followed by the pound (#) button or the **Done** softkey.

Instructions

You need to assign a security code to the user's **Station** screen for each user you want to enable access to TSA. You also need to assign the user an Admin feature button.

For example, to assign a security code of 12345678 to extension 4234, complete the following steps:

1. Type **`change station 4234`**. Press **Enter**.

   The **Station screen** for extension 4234 appears.

2. In the **Security Code** field, type **12345678**.

   You should assign unique security codes for each user. Once you enter the code and move off the field, the system changes the field to '*' for extra security.

3. In one of feature button fields, type **admin**.

   You can assign this button to a feature button or a softkey.

4. Press **Enter** to save your changes.

More information

Before a user can enter the TSA Admin mode, their telephone must be completely idle. After a user presses the **Admin** button and enters a security code (if necessary), they are prompted, via the telephone's display, to choose features to administer to buttons on their telephone.

The user can add, replace, or delete any of the following feature-button types from their telephone.

- CDR Account Code
- Automatic Dial
- Blank
- Call Forwarding
- Call Park
- Call Pickup
- Directed Call Pickup
- Group Page
- Send All Calls
- Toggle Swap
- Activate Whisper Page
- Answerback for Whisper Page
- Whisper Page Off

End-user button changes are recorded to the Communication Manager server's history log so that remote services can know what translations are changed.

Fixing problems

- When a telephone is in the Admin mode, the telephone cannot accept any calls — the telephone is treated as if it were busy. Also, a user cannot make calls while in the Admin mode.
- Any button state a telephone is in when the telephone enters the Admin mode stays active while the telephone is in the Admin mode.
- ACD agents who wish access to the Admin mode of TSA must be logged off before pressing the Admin button. If they are not logged off when they attempt to enter the Admin mode, they receive a denial (single-beep) tone.
- Call Forwarding can be active and works correctly in the Admin mode. An active Call Forwarding button cannot be removed when the telephone is in the Admin mode.
- Since the telephone must be on-hook to go into the Admin mode, the Headset On/Off button must be in the OFF position.
- A telephone that is in the Admin mode of TSA cannot be remotely unmerged by the PSA feature.

  If a user has Abbreviated and Delayed Ringing active, a call can be silently ringing at a telephone and the user might not realize it. This ringing prevents the user from entering the Admin mode of TSA.

Setting Up Enterprise Mobility User

Enterprise Mobility User (EMU) is a software-only feature that provides the ability to associate the buttons and features of a primary telephone to a telephone of the same type anywhere within your company's enterprise.

A home station can be visited by another EMU user while the user is registered as an EMU visitor elsewhere. A home station can be used as a visited station while the principal user's EC500 or other Off-PBX applications are active. And the principal user can activate an Off-PBX application even if their home station is being visited by another EMU user.

Note that in this document, any telephone that is not the primary telephone is referred to as the "visited" telephone and any server that is not the home server of the primary telephone is referred to as the "visited server."

System Requirements for EMU

The following is a list of requirements that you need for the EMU feature:

- QSIG must be the private networking protocol in the network of Communication Manager systems. This requirement also includes QSIG MWI.

  Note:

  All systems in a QSIG network must be upgraded to Communication Manager 4.0 or later in order for the Enterprise Mobility User feature to function properly. If only some systems are upgraded, and their extensions expanded, the EMU feature might not work with the systems that have not been upgraded. See your Avaya technical representative for more information

- Communication Manager Release 3.1 or later software must be running on the home server and all visited servers.
- All servers must be on a Linux platform. EMU is not supported on DEFINITY servers.
- The visited telephone must be the same model type as the primary telephone to enable a optimal transfer of the image of the primary telephone. If the visited telephone is not the same model type, only the call appearance (call-appr) buttons and the message waiting light are transferred.
- All endpoints must be terminals capable of paperless button label display.
- Uniform Dial Plan (UDP)

To activate the EMU feature, a user enters the EMU activation feature access code (FAC), the extension number of their primary telephone, and the security code of the primary telephone on the dial pad of a visited telephone. The visited server sends the extension number, the security code, and the set type of the visited telephone to the home server. When the home server receives the information, it:

- Checks the class of service (COS) for the primary telephone to see if it has PSA permission
- Compares the security code with the security code on the **Station** screen for the primary telephone
- Compares the station type of the visited telephone to the station type of the primary telephone. If both the visited telephone and the primary telephone are of the same type, the home server sends the applicable button appearances to the visited server. If a previous registration exists on the primary telephone, the new registration is accepted and the old registration is deactivated

If the registration is successful, the visited telephone assumes the primary telephone's extension number and some specific administered button types. The display on the primary telephone shows **Visited Registration Active: <Extension>**. The extension number that displays is the extension number of the visited telephone.

Note:

The speed dialing list that is stored on the primary telephone and the station logs are not downloaded to the visited telephone.

Configuring your system for Enterprise Mobility User

To configure your system for the Enterprise Mobility User feature:

1. Type **`display cos`** to view your Class of Service settings.

   The system displays the **Class of Service** screen.

Figure 52: Class of Service screen

```
Page   1 of   2
                                CLASS OF SERVICE

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Fwd-All Calls             n  y  y  y  y  n  n  y  y  n  n  y  y  y  y  y
 Data Privacy                   n  y  y  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling               n  y  y  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions            y  y  n  y  n  n  n  n  y  n  n  n  y  n  y  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    y  n  y  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  n  n  y  y  y  y  y  n  y  y  y  y  y  n  n
 Call Forwarding Busy/DA        n  y  y  n  n  n  n  n  y  n  n  n  n  y  y  y
 Personal Station Access (PSA)  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Extended Forwarding All        n  y  y  n  n  n  n  n  y  n  n  n  n  y  y  y
 Extended Forwarding B/DA       n  y  y  n  n  n  n  n  y  n  n  n  n  y  y  y
 Trk-to-Trk Transfer Override   n  y  y  n  n  n  n  n  n  n  n  n  n  n  y  y
 QSIG Call Offer Originations   n  y  y  n  n  n  n  n  y  n  n  n  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

2. Verify that the **Personal Station Access (PSA)** field is set to **y**. This field applies to the primary telephone and must be set to **y** for EMU.
3. Type **`display feature-access-codes`**.
4. The system displays the **Feature Access Code (FAC)** screen. Page down until you see the fields for **Enterprise Mobility User Activation** and **Deactivation**.

Figure 53: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page   2 of   x
                              FEATURE ACCESS CODE (FAC)
                    Contact Closure Pulse Code:
                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code:
     Emergency Access to Attendant Access Code: *11
         EC500 Self Administration Access Code:
                     Enhanced EC500 Activation: *81     Deactivation:#81
           Enterprise Mobility User Activation: *55     Deactivation:#56
  Extended Call Fwd Activate Busy D/A: *23 All: *24     Deactivation:#23
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code: *88
             Group Control Restrict Activation: *15     Deactivation: #15
                    Hunt Group Busy Activation: *82     Deactivation: #83
                              ISDN Access Code:
              Last Number Dialed Access Code: *54
     Leave Word Calling Message Retrieval Lock: *48
   Leave Word Calling Message Retrieval Unlock: #45
```

5. The feature access codes (FACs) for both EMU activation and EMU deactivation must be set on all servers using EMU. You must enter the FAC of the server in the location from which you are dialing.

   Note:

   To avoid confusion, Avaya recommends that all the servers in the network have the same EMU feature access codes.

6. On page 3 of the **Feature Related System Parameters** screen, use the **EMU Inactivity Interval for Deactivation (hours)** field to administer a system-wide administrable interval for EMU deregistration at a visited switch.

7. Click **Enter** to save your changes.

Setting EMU options for stations

To set EMU options for stations:

1. Type **`add station next`**.

   The system displays the **Station** screen.

Figure 54: Station screen

```
add station next                                              Page   1 of   4
                                     STATION

Extension: 20096                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code: *           TN: 1
     Port:                              Coverage Path 1:            COR: 1
     Name:                              Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                      Message Lamp Ext: 20096
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

                                                Media Complex Ext:
                                                     IP SoftPhone? n
                                               Remote Office Phone? n
```

2. Enter the security code of your primary telephone when you activate or deactivate EMU. The security code is administered on page one of the Station screen. The security code can be up to eight numbers. No letters or special characters are allowed. Once the security code is entered, the system displays a * in the **Security Code** field.
3. On the **Station** screen, page down till you find the **EMU Login Allowed** field.

   The **EMU Login Allowed** field applies to the visited station and must be set to **y** for EMU. The valid entries to this field are **y** or **n**, with **n** as the default. You must set this field to **y** to allow this telephone to be used as a visited station by an EMU user.
4. Click **Enter** to save your changes.

Defining options for calling party identification

To define the options for calling party identification:

1. Type **`display trunk-group x`**, where ***x*** is the number of the trunk group.

   The system displays the **Trunk Group** screen.

Figure 55: Trunk Group screen

```
add trunk-group next                                                  Page 3 of x
                              TRUNK FEATURES
           ACA Assignment? _          Measured: ____     Wideband Support? _
 Long Holding Time(hours): _                            Maintenance Tests? _
  Short Holding Time(sec): _  Data Restriction? _     NCA-TSC Trunk Member: _
  Short Holding Threshold: __        Send Name: _      Send Calling Number: _
             Used for DCS? _                          Send EMU Visitor CPN? y

Suppress # Outpulsing? _         Format: _______
Outgoing Channel ID Encoding: _________    UUI IE Treatment: ______
                                         Maximum Size of UUI IE Contents: ___
                                               Replace Restricted Numbers? _
                                              Replace Unavailable Numbers? _
                                                    Send Connected Number: _
                                                Hold/Unhold Notifications? _

        Send UUI IE? _
          Send UCID? _                 BRS Reply-best DISC Cause Value: __
                                                    Ds1 Echo Cancellation? _

                                        US NI Delayed Calling Name Update? _

                          Network (Japan) Needs Connect Before Disconnect? _

Time (sec) to Drop Call on No Answer:_
Outgoing ANI:_
R2 MFC Signaling:_

DSN Term? n       Precedence Incoming _______   Precedence Outgoing _______
```

2. Page down till you see the **Send EMU Visitor CPN** field.

   This field controls calling party identification, that is, the extension of the primary telephone or the extension of the visited telephone that is used when a call is made from a visited telephone.

3. If you want the system to display calling party information of the primary telephone, the **Send EMU Visitor CPN** field must be set to **y**. There are areas where public network trunks disallow a call if the calling party information is invalid. In this case, there can be instances where the extension of the primary telephone is considered invalid and the extension of the visited telephone must be used. To use the extension of the visited telephone, set the **Send EMU Visitor CPN** field to **n**.

   Note:

   If you set the **Send EMU Visitor CPN** field to **y**, you must set the **Format** field on the same page to either **public** or **unk-pvt**.

4. Click **Enter** to save your changes.

Activating EMU

Use the following steps to activate a visited telephone:

1. At the visited telephone, enter the EMU activation facility-access-code (FAC). You must enter the EMU activation FAC of the server in the location where you are dialing from.
2. Enter the extension of your primary telephone set.
3. Enter the security access code of your primary telephone set. This is the security code administered on the primary telephone's station form on the home server.

   If the registration is successful, you hear confirmation tone.

   If the registration is not successful, you hear audible intercept. Audible intercept is provided when:

   - The registration was rejected by the home server.
   - The telephone where the registration attempt is made is not administered for EMU use.
   - The T1 timer expires at the visited server.

If the home server receives a request from a visited server for a telephone that already has an EMU visitor registration active, the old registration is terminated and the new registration is approved.

If the primary telephone is in-use when a registration attempt is made, the registration attempt fails.

Deactivating EMU

Use the following steps to deactivate the visited telephone:

1. At the visited telephone, enter the EMU deactivation FAC.

   You must enter the EMU deactivation FAC of the server in the location where you are dialing from.

2. Enter the extension number of the primary telephone.

3. Enter the security code of the visited telephone.

   If the visited telephone does not deactivate, the telephone remains in the visited state. To deactivate the visited telephone you can:

   - Perform a busy-out, release busy-out at the visited server.
   - Enter the EMU feature deactivation code and the security code of the visited telephone at the home server location.
   - Press the <mute> RESET function on the IP telephone.

   Note:

   Anytime the visited telephone performs a reset, the EMU registration is deactivated.

   - Unplug the visited DCP set for a period of one minute.

Unplugging or disconnecting a 4600 series set will not deactivate the set.

Chapter 5: Managing Attendant Consoles

Attendant Consoles

The attendant console is the main answering position for your organization. The console operator is responsible for answering incoming calls and for efficiently directing or "extending" calls to the appropriate telephone.

The attendant console also can allow your attendants to monitor:

- system problems
- toll fraud abuse
- traffic patterns

The number of consoles you can have in your organization varies depending on your Avaya solution.

302 attendant consoles

Avaya Communication Manager supports the following 302 attendant consoles: the 302A/B, 302C, and 302D consoles. You might have a basic or enhanced version of these consoles. Figure 56 shows the 302A/B console and Figure 57 shows the 302C console. The next two figures show the button layouts on the Feature area and on the optional Selector console.

Figure 56: 302A and 302B1 attendant console

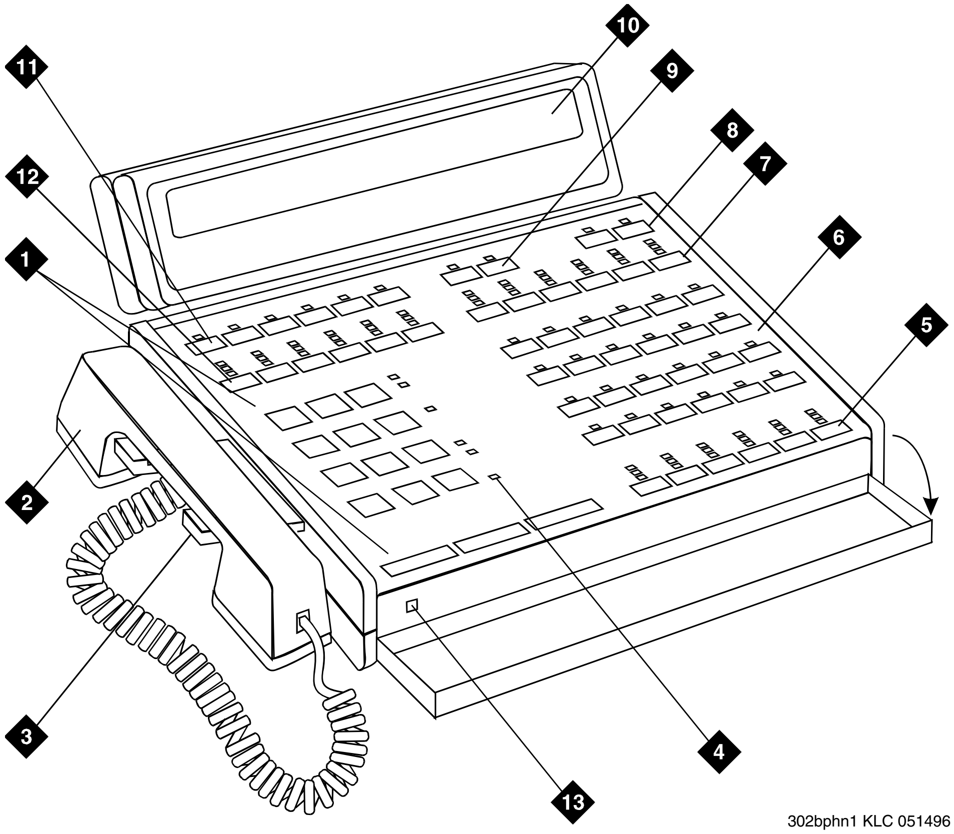


Figure notes:

1. Call processing area
2. Handset
3. Handset cradle
4. Warning lamps and call waiting lamps
5. Call appearance buttons
6. Feature area
7. Trunk group select buttons
8. Volume control buttons
9. Select buttons
10. Console display panel
11. Display buttons
12. Trunk group select buttons
13. Lamp Test Switch

Figure 57: 302C attendant console

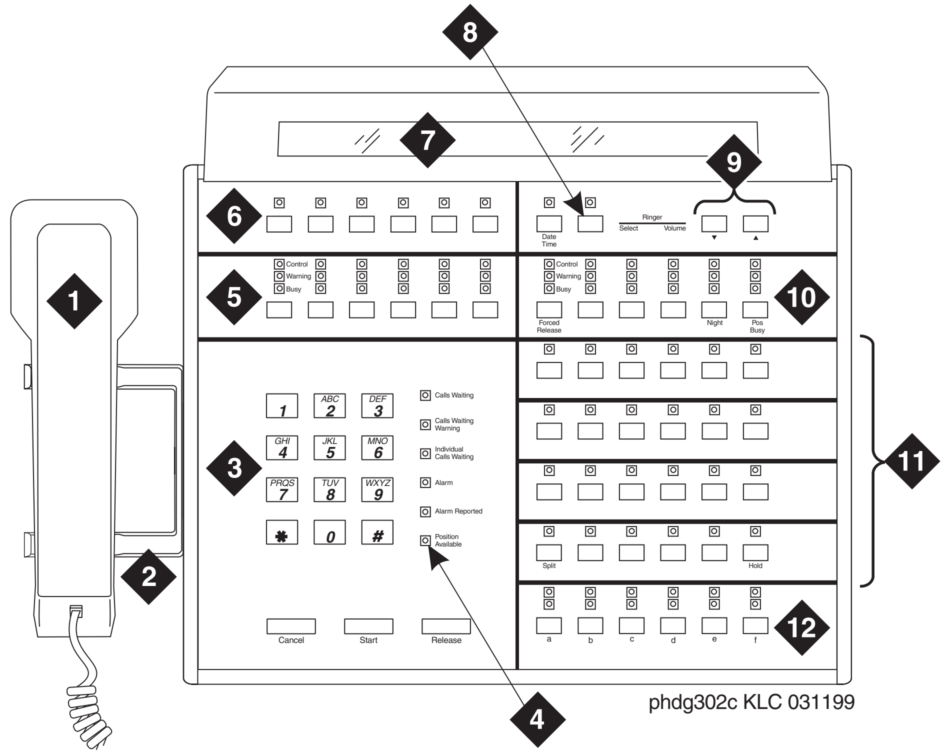


Figure notes:

1. Handset
2. Handset cradle
3. Call processing area
4. Warning lamps and call waiting lamps
5. Outside-line buttons
6. Display buttons
7. Display
8. Select buttons
9. Volume control buttons
10. Outside-line buttons
11. Feature buttons
12. Call appearance buttons

Figure 58: Console feature button layout

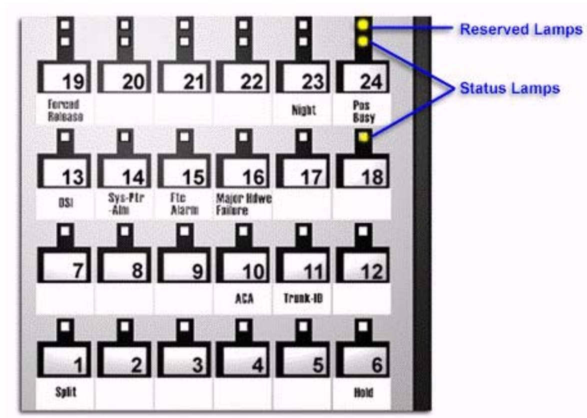
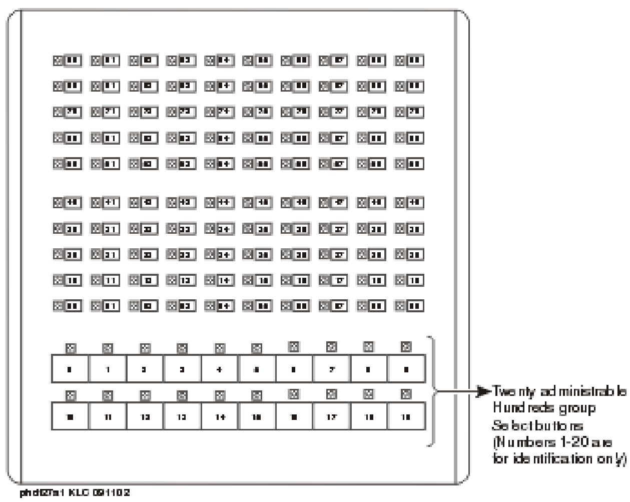


Figure 59: Enhanced Selector Console



302D Console

The 302D console provides the following enhancements to the 302C console:

- Modular handset/headset connection

  The console accepts a standard RJ11, 4-pin modular handset or headset. This connection replaces the quarter-inch, dual-prong handset/headset connection.

- Activate/deactivate push-button

  You can use the push-button on the left side of the console to activate or deactivate the console. A message appears on the console identifying that the button must be pressed to activate the console.

- Two-wire DCP compatibility

  The console is compatible with two-wire DCP circuit packs only, not four-wire DCP circuit packs.

- Headset volume control

  The console can now control the volume of an attached headset.

- Noise expander option

  The console has circuitry to help reduce background noise during pauses in speech from the console end of a conversation. This option is normally enabled.

- Support for Eurofont or Katakana character set

  The console can show the Eurofont or Katakana character set. Administration of these character sets must be coordinated with the characters sent from Avaya Communication Manager.

Avaya PC consoles

The Avaya PC Console is a Microsoft Windows-based call handling application for Avaya Communication Manager attendants. It provides an ideal way to increase your productivity and to better serve your customers.

PC Console offers all the call handling capabilities of the hardware-based Avaya 302 attendant console with a DXS module, plus several enhanced features and capabilities. The enhanced features provide you with the ability to see up to six calls at once, and to handle all calls more efficiently.

PC Console also provides a powerful directory feature. You are able to perform searches, display user information, including a photo. You are able to place a call immediately from the directory.

And, because PC Console resides on a Windows-based PC, you are able to use other software applications at the same time. If a call comes in while you are in another application, you are able to handle it immediately.

For more information about the Avaya PC Console, contact your Avaya account team or representative.

SoftConsole IP Attendant

The SoftConsole is a Windows-based application that can replace the 302B hard console. The SoftConsole is similar to PC Console, but it performs call answering and routing through a PC interface via IP. For more information, contact your Avaya account team or representative.

Adding an Attendant Console

Usually Avaya connects and administers your primary attendant console during cutover. However, you might find a need for a second attendant console, such as a backup console that is used only at night. Note that these instructions do not apply to adding a PC Console or SoftConsole. For more information, see the appropriate console documentation.

To add a night-only attendant console, complete the following steps:

1. Type **`add attendant 2`**. Press **Enter**.

   The **Attendant Console** screen appears.

Figure 60: Attendant Console Button Layout

```
add attendant n                                          Page   1 of  x
                             ATTENDANT CONSOLE 1

         Type: console          Name: 27 character attd cons name
    Extension: 1000            Group: 1               Auto Answer: none
 Console Type: principal          TN: 1               Data Module? y
         Port: 01C1106           COR: 1         Disp Client Redir? n
Security Code:                   COS: 1          Display Language: english
                                                H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote             Local Remote
 1: 9                   5:                       9:
 2: 82                  6:                      10:
 3:                     7:                      11:
 4:                     8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

2. In the **Type** field, enter **302**.

   This is the type of attendant console.

3. If you want this attendant to have its own extension, enter one in the **Extension** field.

   Tip:

   If you assign an extension to the console, the class of restriction (COR) and class of service (COS) that you assign on this **Attendant Console** screen override the COR and COS you assigned on the **Console Parameters** screen. To avoid unexpected behavior, you should assign the same COR and same COS on both screens.

   If you give your attendants an individual extension, users can call the attendant directly by dialing the extension.

   Individual attendant extensions also allow attendants to use features that an attendant group cannot use — for example, you can assign them to hunt groups.

4. In the **Console Type** field, enter **night-only**.

   This indicates how this console is used in your organization—as a principal, day only, night only, or day/night console. You can have only one night-time console (night only or day/night) in the system.

5. In the **Port** field, enter the port address for this console.
6. Type a name to associate with this console in the **Name** field.
7. In the **DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS** fields, enter trunk access codes for the trunks you want the attendant to be able to select with just one button.
8. If you are using the **Enhanced Selector** console, set the **HUNDREDS SELECT BUTTON ASSIGNMENTS** that you want this console to have.

   If you want this console to be able to access extensions in the range 3500 to 3999, you need to assign them 5 **Hundreds Select Buttons**: **35** for extensions 3500 to 3599, **36**, **37**, **38**, and **39**.

9. Assign the Feature Buttons that you want the 302 console to have.

   To determine which buttons you can assign to a console, see Attendant Console Feature Buttons

   Tip:

   Feature buttons are not numbered top-to-bottom on the attendant console, as you might expect. Button numbers map to physical positions on the console as shown in Figure 58.

10. Press **Enter** to save your changes.

Attendant Console Feature Buttons

The following table lists the feature buttons that you can assign to an attendant console.

Table 4: Attendant console feature buttons

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Abbreviated Dialing	AD	abrv-dial (List:___ DC:___)	1 per List/ DC	1
Administered Connection [status lamp]	AC Alarm	ac-alarm	1	
Automatic Call Distribution (ACD)	After Call Work	after-call (Grp. No.__)	N	2
	Assist	assist (Grp. No:__)	1 per split group	2
	Auto In	auto-in (Grp. No.__)	1 per split group	2
	Auxiliary Work	aux-work (Grp. No.__)	1 per split group	2
	Manual-In	manual-in (Grp. No.__)	1 per split group	2
	Release	release	1	
	Work Code	work-code	1	
	Stroke (0-9)	stroke-cnt (Code:_)	1	3
Attendant Console (Calls Waiting)	CW Aud Off	cw-ringoff	1	
Attendant Control of Trunk Group Access (Activate)	Cont Act	act-tr-grp	1	
Attendant Control of Trunk Group Access (Deactivate)	Cont Deact	deact-tr-g	1	
				1 of 7

Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Attendant Direct Trunk Group Select	Local TG Remote TG	local-tgs (TAC:__) remote-tgs (LT:__) (RT:__)	12	4
Attendant Crisis Alert	Crisis Alert	crss-alert	1	
Attendant Display [display buttons]	Date/Time	date-time	1	
	Inspect Mode	inspect	1	
	Normal Mode	normal	1	
	Stored Number	stored-num	1	
Attendant Hundreds Group Select	Group Select _	hundrd-sel (Grp:__)	20 per console	5
Attendant Room Status	Occupied Rooms Status	occ-rooms	1	6
	Maid Status	maid-stat	1	6
Attendant Override	Override	override	1	
Automatic Circuit Assurance	ACA	aca-halt	1 per system	
Automatic Wakeup (Hospitality)	Auto Wakeup	auto-wkup	1	
Busy Verification	Busy Verify	verify	1	
Call Coverage	Cover Cback	cov-cback	1	
	Consult	consult	1	
	Go To Cover	goto-cover	1	
Call Coverage [display button]	Cover Msg Rt	cov-msg-rt	1	
Call Offer (Intrusion)	Intrusion	intrusion	1	
				2 of 7

Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Call Prompting [display button]	Caller Info	callr-info	1	
Call Type	Call Type	type-disp	1	
Centralized Attendant Service	CAS-Backup	cas-backup	1	
Check In/Out (Hospitality) [display buttons]	Check In	check-in	1	
	Check Out	check-out	1	
Class of Restriction [display button]	COR	class-rstr	1	
Conference Display [display button]	Conference Display	conf-dsp	1	
Demand Print	Print Msgs	print-msgs	1	
DID View	DID View	did-view	1	
Do Not Disturb (Hospitality)	Do Not Disturb	dn-dst	1	
Do Not Disturb (Hospitality) [display buttons]	Do Not Disturb Ext	ext-dn-dst	1	
	Do Not Disturb Grp	grp-dn-dst	1	
Don’t Split	Don’t Split	dont-split	1	
Emergency Access To the Attendant	Emerg. Access To Attd	em-acc-att	1	
Facility Busy Indication [status lamp]	Busy (trunk or extension#)	busy-ind (TAC/Ext: _)	1 per TAC/ Ext.	7
Facility Test Calls [status lamp]	FTC Alarm	trk-ac-alm	1	
Far End Mute [display button]	Far End Mute for Conf	fe-mute	1	
				3 of 7

Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Group Display	Group Display	group-disp	1	
Group Select	Group Select	group-sel	1	
Hardware Failure [status lamps]	Major Hdwe Failure	major-alrm	10 per system	
	Auto Wakeup	pr-awu-alm	1	
	DS1 (facility)	ds1-alarm	10 per system	
	PMS Failure	pms-alarm	1	
	PMS Ptr Alm	pr-pms-alm	1	
	CDR 1 Failure	cdr1-alrm	1	
	CDR 2 Failure	cdr2-alrm	1	
	Sys Ptr Alm	pr-sys-alm	1	
Hold	Hold	hold	1	
Integrated Directory [display button]	Integrtd Directory	directory	1	
Incoming Call Identification	Coverage (Group number, type, name, or ext.#)	in-call-id	N	
Intrusion (Call Offer)	Intrusion	intrusion	1	
Leave Word Calling	Cancel LWC	lwc-cancel	1	
	LWC	lwc-store	1	
Leave Word Calling [display buttons]	Delete Msg	delete-msg	1	
	Next	next	1	
	Call Display	call-disp	1	
				4 of 7

Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Leave Word Calling (Remote Message Waiting) [status lamp]	Msg (name or extension #)	aut-msg-wt (Ext:___)	N	
Link Failure	Link Failure (Link No.__)	link-alarm (Link No.__)	1 per Link #	8
Login Security Violation	lsvn-halt	lsvn-halt	1 per system	
Message Waiting	Message Waiting Act.	mwn-act	1 per system	
	Message Waiting Deact.	mwn-deact	1 per system	
Night Service	Trunk Grp. NS	trunk-ns (Grp. No.__)	1 per trunk group	9
No Answer Alert	noans-altr	noans-altr	1 per group	
Off Board Alarm	off-bd-alm	off-bd-alm	1 per group	
Page 1 Link Alarm Indication	PAGE1 Alarm	pg1-alarm	1 per station	
Page 2 Link Alarm Indication	PAGE2 Alarm	pg2-alarm	1 per station	
PMS Interface [display buttons]	PMS display			
Priority Attendant Group	prio-grp	prio-grp	1	
Priority Calling	Prior Call	priority	N	
Position Busy	Position Busy	pos-busy	1	
Queue Status Indications (ACD) [display buttons]	AQC	atd-qcalls	1	
	AQT	atd-qtime		
				5 of 7

Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Queue Status Indications (ACD) [status lamps]	NQC	q-calls (Grp:_)	1	10
	OQT	q-time Grp:_)	1 per hunt group	10
Remote Access Security Violation	rsvn-halt	rsvn-halt	1 per system	
Ringing	In Aud Off	in-ringoff	1	
Security Violation Notification Halt	ssvn-halt	ssvn-halt	1 per system	
Serial Call	Serial Call	serial-cal	1	
Split/Swap	Split-swap	split-swap	1	11
System Reset Alert	System Reset Alert [status lamp]	rs-alert	1	
Station Security Code Notification Halt	ssvn-halt	ssvn-halt	1 per system	
Night Service (ACD)	Hunt Group	hunt-ns (Grp. No.__)	3 per hunt group	12
Time of Day Routing [display buttons]	Immediate Override	man-ovrid	1	
	Clocked Override	clk-overid	1	
Timed Reminder	RC Aud Off	re-ringoff	1	
Timer	Timer	timer	1	
Trunk Identification [display button]	Trunk-ID	trk-id	1	
Trunk Group Name [display button]	Trunk-Name	trunk-name	1	

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Table 4: Attendant console feature buttons (continued)

Feature or Function	Recommended Button Label	Value Entered on Attendant Console Screen	Maximum Allowed	Notes
Visually Impaired Service (VIAS)	VIS	vis	1	
	Console Status	con-stat	1	
	Display	display	1	
	DTGS Status	dtgs-stat	1	
	Last Message	last-mess	1	
	Last Operation	last-op	1	
VDN of Origin Announcement Repeat	VOA Repeat	voa-repeat	1	12
VuStats	VuStats	vu-display	1	
N = any number of buttons on the telephone can be assigned to this feature. For telephone feature button descriptions, see Telephone Feature Buttons Table on page 134.				
				7 of 7

1. List: List number 1 to 3 where the destination number is stored.
   DC: Dial codes of destination number.
2. Grp: The split group number for ACD.
3. Code: Enter a stroke code (0 through 9).
4. TAC: local-tgs — TAC of local TG
   remote-tgs — (L-TAC) TAC of TG to remote PBX
   remote-tgs — (R-TAC) TAC of TG on remote PBX
   The combination of local-tgs/remote-tgs per console must not exceed 12 (maximum).
   Label associated button appropriately so as to easily identify the trunk group.
5. Grp: Enter a hundreds group number (1 through 20).
6. **Enhanced Hospitality** must be enabled on the **System-Parameters Customer-Options (Optional Features)** screen.
7. Ext: Can be a VDN extension.
8. Link: A link number — 1 to 8 for multi-carrier cabinets, 1 to 4 for single-carrier cabinets.
9. Grp: A trunk group number.
10. Grp: Group number of the hunt group.
11. Allows the attendant to alternate between active and split calls.
12. VDN of Origin must be enabled.

Setting Console Parameters

You can define system-wide console settings on the **Console Parameters** screen.

For example, if you want to warn your attendants when there are more than 3 calls in queue or if a call waits for more than 20 seconds, complete the following steps:

1. Type **`change console-parameters`**. Press **Enter**.

   The Console Parameters screen appears.

Figure 61: Console Parameters screen

```
change console-parameters                                        Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                      COR: 0
       Calls in Queue Warning: 5                        Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
                                              Night Service Act. Ext.:
                 IAS (Branch)? n              IAS Tie Trunk Group No.:
         IAS Att. Access Code:                  Alternate FRL Station:
              Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

2. In the **Calls in Queue Warning** field, enter **3**.

   The system lights the console's second call waiting lamp if the number of calls waiting in the attendant queue exceeds 3 calls. Click **Next** to display page 2.

3. In the **Time in Queue Warning** field, enter **20**.

   The system issues a reminder tone if a call waits in the attendant queue for more than 20 seconds.

4. Press **Enter** to save changes.

Note that some of the settings on the individual **Attendant Console** screens can override your system-wide settings.

Removing an Attendant Console

Before you physically remove an attendant from your system, check the attendant's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove attendant 3, which also is assigned extension 4345:

1. Type `status attendant 3`. Press **Enter**.

   The **Attendant Status** screen appears.

2. Make sure that the attendant:
   - is plugged into the jack
   - is idle (not making or receiving calls)

3. Type `list usage extension 4345`. Press **Enter**.

   The **Usage** screen shows where the extension is used in the system.

4. Press **Cancel**.

5. If the attendant extension appears on the **Usage** screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type `change hunt group 2` and delete the extension from the list.

6. Type `remove attendant 3`. Press **Enter**.

   The system displays the Attendant Console screen so you can verify that you are removing the correct attendant.

7. If this is the correct attendant, press **Enter**.

   If the system responds with an error message, the attendant is busy or still belongs to a group. Press **Cancel** to stop the request, correct the problem, and type `remove attendant 3` again.

8. Remove the extension from voice mail service if the extension has a voice mailbox.

9. Type `save translations`. Press **Enter** to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the console from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Providing Backup for an Attendant

Avaya Communication Manager allows you to configure your system so that you have backup positions for your attendant. Attendant Backup Alerting notifies backup telephones that the attendant need assistance in handling calls. The backup telephones are alerted when the attendant queue reaches the queue warning level or when the console is in night service.

Once a backup telephone receives an alert, the user can dial the Trunk Answer Any Station (TAAS) feature access code (FAC) to answer the alerting attendant calls.

Tip:

You can find more information about attendant backup in the *GuestWorks Technician Handbook.*

Before you start

- You can assign the attendant backup alerting only to multiappearance telephones that have a client room class of service (COS) set to No. For more information, see Class of Service on page 852.
- If you have not yet defined a Trunk Answer Any Station (TAAS) feature access code, you need to define one and provide the feature access code to each of the attendant backup users. For more information, see Feature Access Code (FAC) on page 980.

Instructions

To enable your system to alert backup stations, you need to administer the **Console Parameters** screen for backup alerting. You also need to give the backup telephones an attendant queue calls feature button and train your backup users how to answer the attendant calls.

To configure the system to provide backup alerts and to setup extension 4345 to receive these alerts, complete the following steps:

1. Type `change console-parameters`. Press **Enter**.

   The **Console Parameters** screen appears.
2. In the **Backup Alerting** field, enter **y**.
3. Press **Enter** to save changes.

   The system will now notify anyone with an attendant queue calls button when the attendant queue reaches the warning level or when the console is in night service.

4. Type **`change station 4345`**. Press **Enter**.

   The **Station screen** appears.

5. In one of the Button Assignment fields, enter **atd-qcalls**.

   The atd-qcalls button provides the visual alerting for this telephone. When this button is dark (idle state), there are no calls in the attendant queue. When the button shows a steady light (busy state), there are calls in the attendant queue. When button shows a flashing light (warning state), the number of calls in the attendant queue exceeds the queue warning. The backup-telephone user also hears an alerting signal every 10 seconds.

6. Press **Enter** to save changes.

Now you need to train the user how to interpret the backup alerting and give them the TAAS feature access code so that they can answer the attendant calls.

Chapter 6: Managing Displays

Displaying Caller Information

This chapter provides information on the messages that appear on the screens of display telephones.

Your system uses automatic incoming call display to provide information about incoming calls to a display telephone that is in use, or active on a call. The information is displayed for 30 seconds on all telephones except for CALLMASTER telephones, where the display goes blank after 30 seconds. However, the information for each new call overrides the existing message.

Call information appears on the display only if the call terminates at the telephone. For example, if the call is forwarded to another extension, no call information appears.

For more information on the buttons and languages you can set up for the messages that appear on the display, see the "Telephone Displays" feature description in the *Feature Description and Implementation for Avaya Communication Manager,* 555-245-505.

Displaying ANI Calling Party Information

Calling party information might consist of either a billing number that sometimes is referred to as Automatic Number Identification (ANI), or a calling party number. Your telephone might display the calling party number and name, or the incoming trunk group name.

Instructions

We will set up a tie trunk group to receive calling party information and display the calling party number on the telephone of the person called.

1. Type **`change trunk group`** ***`nnnn`***, where ***`nnnn`*** is the trunk group you want to change.

   Press **Next Page** until you see the **Trunk Parameters** fields on the **Trunk Group screen (page 2)**.

Figure 62: Trunk Group screen

```
add trunk-group next                                            Page   2 of  x
      Group Type: co                  Trunk Type:

TRUNK PARAMETERS

    Outgoing Dial Type: tone                           Cut-Through? n
     Trunk Termination: rc                      Incoming Dial Type: tone
                                           Disconnect Timing(msec): 500


            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none
     Analog Loss Group: 6                           Digital Loss Group: 11
                                  Trunk Gain: high

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 10            Receive Answer Supervision? n
```

2. Type **tone** in the **Incoming Dial Type** field.
3. Press **Next Page** and type ***ANI*DNIS** in the **Incoming Tone (DTMF) ANI** field.
4. Press **Enter** to save your changes.

Displaying ICLID Information

Communication Manager collects the calling party name and number (Incoming Call Line Identification, or ICLID) received from the central office (CO) on analog trunks.

Before you start

Be sure the **Analog Trunk Incoming Call ID** field is set to **y** on the **System-Parameters Customer-Options (Optional Features)** screen.

See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207 for information on the required circuit pack.

Instructions

We will set up the analog diod trunk group 1 to receive calling party information and display the calling party number on the telephone of the person called.

1. Type **`change trunk group 1`**.

   The **Trunk Group** screen for trunk group 1 appears. The **Group Type** field is already set to **diod**.

2. Press **Next Page** to display the Trunk Features fields on the Trunk Group screen (page 3).

Figure 63: Trunk Features screen

```
change trunk-group n                                                Page 3 of x
                                 TRUNK GROUP
TRUNK FEATURES
          ACA Assignment? n          Measured: none
                                                          Maintenance Tests? y
                             Data Restriction? n
   Suppress # Outpulsing? n
                                   Receive Analog Incoming Call ID: Bellcore
Incoming Tone (DTMF) ANI: no
```

3. Type **Bellcore** in the **Receive Analog Incoming Call ID** field.
4. Press **Next Page** to display the **Administrable Timers** screen.

Figure 64: Administrable Timers screen

```
add trunk-group next                                                    Page 3 of x
                           ADMINISTRABLE TIMERS
                Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                              Outgoing Dial Guard(msec): _____
  Incoming Glare Guard(msec): _____         Outgoing Glare Guard(msec): _____

                                 Outgoing Rotary Dial Interdigit (msec): ____
       Ringing Monitor(msec): _____                 Incoming Seizure(msec): ____
   Outgoing End of Dial(sec): ___       Outgoing Seizure Response(sec): ___
 Programmed Dial Pause(msec): _____       Disconnect Signal Error(sec): ___
          Flash Length(msec): ____
                                                      Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ____    Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

5. Type **120** in the **Incoming Seizure (msec)** field.
6. Press **Enter** to save your changes.

Changing the Display Language

This section explains how to set or change the display language.

Setting the display language

To set or change the display language:

1. Type **`change station nnnn`**, where ***nnnn*** is the extension of the station that you want to change. Press **Enter**.

   The system displays the **Station** screen.

Figure 65: Station screen

```
add station next                                            Page 1 of X
                              STATION

Extension:                    Lock Messages? n                      BCC: 0
     Type:                    Security Code:                         TN: 1
     Port:                  Coverage Path 1:                        COR: 1
     Name:                  Coverage Path 2:                        COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2            Personalized Ringing Pattern: 3
             Data Module? n                       Message Lamp Ext: 1014
            Speakerphone: 2-way                Mute button enabled? y
        Display Language? English
                   Model:                          Expansion Module?

 Survivable GK Node Name:                         Media Complex Ext:
          Survivable COR:                              IP Softphone? y
   Survivable Trunk Dest?                      Remote Office Phone? y
                                                IP Video Softphone?
                                                          IP Video?

                                               Customizable Labels?
```

2. In the **Display Language** field, enter the display language you want to use.

Tip:

Time of day is displayed in 24-hour format (00:00 - 23:59) for all languages except **english**, which is displayed in 12-hour format (12:00 a.m. to 11:59 p.m.).To display time in 24-hour format and display messages in English, set the **Display Language** field to **unicode**. When you enter **unicode**, the station displays time in 24-hour format, and if no Unicode file is installed, displays messages in English by default. For more information on Unicode, see Administering Unicode display on page 203.

3. Press **Enter** to save your changes.

Entering translations for a user-defined language

To enter translations for an existing message using a user-defined language:

1. Type **`change attendant n`**, where ***n*** is the number of the attendant console you want to change. Press **Enter**.

   The system displays the **Attendant Console** screen.

2. In the **Display Language** field, enter **user-defined**.
3. Press **Enter** to save your changes.
4. Type **`change display-messages n`**, where ***n*** is the message for which you want to translate the display language. Click **help** to view the messages that you can choose to translate. Press **Enter**.

   The system displays the **Language Translations** screen for the type of message that you want to translate.

Figure 66: Language Translations screen

```
change display-messages transfer-conference                     Page 1 of x
                         LANGUAGE TRANSLATIONS

 1.      English: Transfer completed.
     Translation: ********************************

 2.      English: Call next party.
     Translation: ****************************************

 3.      English: Press conference to add party.
     Translation: ****************************************

 4.      English: ^-party conference in progress.
     Translation: ****************************************

 5.      English: Conference canceled.
     Translation: ****************************************

 6.      English: Select line ^ to cancel or another line.
     Translation: ****************************************
```

5. In the **Translation** field, type the translation of the message in the user-defined language.
6. Press **Enter** to save your changes.

Note:

As of July 1, 2005, new messages are no longer added to the **Language Translations** screens, so these screens might not show all available Communication Manager messages. The preferred method for entering translations for user-defined phone messages is to use the *Avaya Message Editing Tool (AMET)*. This tool is available for download from http://support.avaya.com/amet. For more information, see *Avaya Message Editing Tool (AMET) Job Aid*.

Administering Unicode display

To use Unicode display languages, you must have the appropriate Avaya Unicode Message files loaded on Communication Manager. These files are named *avaya_unicode.txt* (standard phone messages), *custom_unicode.txt* (posted messages and system labels), *avaya_user-defined.txt* (standard phone messages using Eurofont), and *custom_user-defined.txt* (posted messages and system labels using Eurofont).

To use the Phone Message files *avaya_unicode.txt* and *custom_unicode.txt*, you must have Unicode-capable stations, such as the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, and Avaya Softphone R5.0. Unicode is also an option for the

2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

Only Unicode-capable stations have the script (font) support that is required to match the scripts that the Unicode Phone Message file uses. To use the user-defined messages files *avaya_user-defined.txt* and *custom_user-defined.txt* you must use an Avaya digital phone that supports Eurofont or Kanafont.

Note:

To view the dial pad letter/number/symbol mapping tables used for the integrated directory, see "Telephone Display" in Feature Description and Implementation for Avaya Communication Manager, 555-245-205.

For Communication Manager 2.2 and later, the following languages are available using Unicode display:

- Chinese
- Czech
- Danish
- Dutch
- German
- Hebrew
- Hungarian
- Icelandic
- Italian
- Japanese
- Korean
- Macedonian
- Polish
- Romanian
- Russian
- Servian
- Slovak
- Swedish
- Ukrainian

Obtaining and installing Phone Message files

A Unicode Message file for each supported language is available in a downloadable ZIP file on the Avaya support Web site (http://www.avaya.com/unicode). You can also create a new translation or edit an existing translation with the Avaya Message Editing Tool (AMET) (http://support.avaya.com/amet). Additional languages are periodically becoming available, so check this site often for the most up-to-date message files.

Note:

Refer to the *Communication Manager Messages Job Aid* for details on the following procedures.

To administer Unicode display:

1. Download the appropriate Unicode message file to your PC. For an existing translation, download the desired language from http://www.avaya.com/unicode.
2. If necessary, create a new translation, or modify an existing translation, using the Avaya Message Editing Tool (AMET), available at http://support.avaya.com/amet.

   Note:

   Only the Avaya Message Editing Tool (AMET) can be used for translation edits, using any other editor will not update the Phone Message File correctly and such files will fail to install. See the *Avaya Message Editing Tool (AMET) Job Aid* in the Generic Phone Message Package file for more details on using AMET.
3. Transfer the Phone Message file to an Avaya S8XXX Server that is running Communication Manager 2.2 or later, using the Avaya Web pages, the Avaya Installation Wizard, or ftp.
4. Install Phone Message files with the Avaya Install Web Page. The Avaya Installation Wizard only supports install of Unicode Phone Message files. Note that the Installation Wizard is the same wizard that you use to transfer Phone Message files to an Avaya S8XXX Server that is running Communication Manager 2.2 or later.
5. After you install *avaya_unicode.txt* or *avaya_user-defined.txt*, you must do a reset 4 to load the file in to Communication Manager memory. Note that a reset 4 is *not* required to load the files *custom_unicode.txt* and *custom-defined.txt*.
6. Set the **Display Language** field on the **Station** screen to **unicode**. Note that the keyword **unicode** only appears if a Unicode-capable telephone is entered in the **Station** screen **Type** field. To use a user-defined file, set the **Display Language** field on the **Station** screen to **user-defined**.

   Note:

   There is no uninstall option for Phone Message files. You can reload a new Phone Message file. This will overwrite existing Phone Message files.

Checking the status of Phone Message file loads

To verify that a Unicode Phone Message file is loaded correctly, run **`status station`** ***`xxxx`*** on any administered station. If the Unicode Phone Message file is loaded correctly, the **Display Messages Scripts** field on the second page contains the scripts that are in this file.The **General Status** screen for stations contains three Unicode script-related fields. To access the **General Status** screen, type **`status station`** ***`xxxx`***, where ***`xxxx`*** is the extension of the station. The **General Status** screen appears. Click **Next** to display page 2 of the screen:

Figure 67: General Status screen

```
 status station nnnn                                                 page 2 of x

                                GENERAL STATUS

CONNECTED STATION INFORMATION
               Part ID Number: unavailable
                Serial Number: unavailable
          Station Lock Active? No

UNICODE DISPLAY INFORMATION
          Native Name Scripts: 0x00000007:Latn;Lat1;LatA
     Display Messages Scripts: 0x04000007:Latn;Lat1;LatA;Jpan
    Station Supported Scripts: 0x7c000007:Latn;Lat1;LatA;Jpan;Chis;Chit;Korn
```

"Scripts" are a collection of symbols used to represent text in one or more writing systems. The three script fields shown in the **UNICODE DISPLAY INFORMATION** section are as follows:

Native Name Scripts: Scripts supported in the Unicode station name.

Display Messages Scripts: The scripts used in the Unicode Display Language.

Station Supported Scripts: The scripts supported in the IP station that is registered to an extension.

Unicode "Native Name" support

Communication Manager supports Unicode for the "Name" associated with Vector Directory Numbers (VDNs), trunk groups, hunt groups, agent login id, vector names, station names, Invalid Number Dialed Display (**Feature-Related System Parameters** screen) and Restricted Number Dialed Display (**Feature-Related System Parameters** screen). The Unicode Name (also referred to as *Native Name* and *Name 2)* fields are hidden fields that are associated with the name fields you administer on the respective screens for each. These fields can only be administered using Avaya Site Administration (ASA) or MultiSite Administrator (MSA).

- The Unicode VDN Name is associated with the name administered in the **Name** field on the **Vector Directory** screen. You must use MSA.

- The Unicode Trunk Group Name is associated with the name administered in the **Group Name** field on the **Trunk Group** screen. You must use MSA.
- The Unicode Hunt Group Name is associated with the name administered in the **Group Name** field on the **Hunt Group** screen. You must use MSA.
- The Unicode Station Name is associated with the name administered in the **Name** field on the **Station** screen. You must use ASA or MSA.

Script Tags and Abbreviations

The following table defines the script tags and spells out the script abbreviations.

Script Number	Script Tag Bit (hex)	Start Code.. End Code	Script or Block Name	SAT Screen Name
1	00000001	0000..007F	Basic Latin	Latn
2	00000002	0080..00FF	Latin-1 Supplement	Lat1
3	00000004	0100..017F	Latin Extended-A	LatA
4	00000008	0180..024F	Latin Extended-B	LatB
5	00000010	0370..03FF	Greek and Coptic	Grek
6	00000020	0400..04FF	Cyrillic	Cyrl
6	00000020	0500..052F	Cyrillic Supplementary	Cyrl
7	00000040	0530..058F	Armenian	Armn
8	00000080	0590..05FF	Hebrew	Hebr
9	00000100	0600..06FF	Arabic	Arab
10	00000200	0900..097F	Devanagari	Deva
11	00000400	0980..09FF	Bengali	Beng
12	00000800	0A00..0A7F	Gurmukhi	Guru
13	00001000	0A80..0AFF	Gujarati	Gujr
14	00002000	0B00..0B7F	Oriya	Orya
15	00004000	0B80..0BFF	Tamil	Taml
16	00008000	0C00..0C7F	Telugu	Telu
17	00010000	0C80..0CFF	Kannada	Knda
18	00020000	0D00..0D7F	Malayalam	Mlym

Script Number	Script Tag Bit (hex)	Start Code.. End Code	Script or Block Name	SAT Screen Name
19	00040000	0D80..0DFF	Sinhala	Sinh
20	00080000	0E00..0E7F	Thai	Thai
21	00100000	0E80..0EFF	Lao	Laoo
22	00200000	1000..109F	Myanmar	Mymr
23	00400000	10A0..10FF	Georgian	Geor
32	80000000	1100..11FF	Hangul Jamo	Hang
24	00800000	1700..171F	Tagalog	Tglg
25	01000000	1780..17FF	Khmer	Khmr
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	2E80..2EFF	CJKV Radicals Supplement	Jpan ChiS ChiT Korn Viet
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	2F00..2FDF	Kangxi Radicals	Jpan ChiS ChiT Korn Viet
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	3000..303F	CJKV Symbols and Punctuation	Jpan ChiS ChiT Korn Viet
27	04000000	3040..309F	Hiragana	Jpan
27	04000000	30A0..30FF	Katakana	Jpan
29	10000000	3100..312F	Bopomofo	ChiT
32	80000000	3130..318F	Hangul Compatibility Jamo	Hang
29	10000000	31A0..31BF	Bopomofo Extended	ChiT
27	04000000	31F0..31FF	Katakana Phonetic Extensions	Jpan

Script Number	Script Tag Bit (hex)	Start Code.. End Code	Script or Block Name	SAT Screen Name
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	3200..32FF	Enclosed CJK Letters and Months	Jpan ChiS ChiT Korn Viet
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	3300..33FF	CJKV Compatibility	Jpan ChiS ChiT Korn Viet
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	3400..4DBF	CJKV Unified Ideographs Extension A	Jpan ChiS ChiT Korn Viet
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	4E00..9FFF	CJKV Unified Ideographs	Jpan ChiS ChiT Korn Viet
32	80000000	AC00..D7AF	Hangul Syllables	Hang
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	F900..FAFF	CJK Compatibility Ideographs	Jpan ChiS ChiT Korn Viet
	00000100	FB50..FDFF	Arabic Presentation Forms-A	Arab
27 28 29 30 31	04000000 08000000 10000000 20000000 40000000	FE30..FE4F	CJK Compatibility Forms	Jpan ChiS ChiT Korn Viet
	00000100	FE70..FEFF	Arabic Presentation Forms-B	Arab
26	02000000	FF00..FFEF	Halfwidth and Fullwidth Forms	Kana

Administering displays for QSIG trunks

For proper transmission of QSIG name data for display, administer the following settings:

- **Trunk Group** screen
  - **Group Type**: **ISDN**
  - **Character Set for QSIG Names**: **iso8859-1**
  - **Outgoing Display**: **y**
  - **Send Calling Number**: **y**
  - **Send Name**: **y**
- **Signaling Group** screen
  - **Supplementary Service Protocol**: **b**
- **System-Parameters Country-Options** screen
  - **Display Character Set**: **Roman**

Fixing problems

Symptom	Cause and Solution
Characters that display are not what you thought you entered.	This feature is case sensitive. Check the table to make sure that you entered the right case.
You entered "~c", and "*" appears on the display instead.	Lower-case "c" has a specific meaning in Avaya Communication Manager, and therefore cannot be mapped to any other character. An asterisk "*" appears in its place.
You entered "~->" or "~<-" and nothing appears on the display.	These characters do not exist as single keys on the standard US-English keyboard. Therefore the system is not programmed to handle them.
Enhanced display characters appear in fields that you did not update.	If an existing display field contains a tilde (~) followed by Roman characters, and you update and submit that screen after this feature is activated, that field will display the enhanced character set.
	1 of 2

Symptom	Cause and Solution
Nothing displays on the terminal at all.	Some unsupported terminals do not display anything if a special character is presented. Check the model of display terminal that you are using.
You entered a character with a descender and part of it appears cut off in the display.	Some of the unused characters in Group2a have descenders that do not appear entirely within the display area. These characters are not included in the character map. For these characters (g,j,p,q,y), use Group1 equivalents.
	2 of 2

Related topics

See the "Telephone Displays" and the "Administrable Display Languages" feature descriptions in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205 for more information.

To view the dial pad letter/number/symbol mapping tables used for the integrated directory, see "Telephone Display" in Feature Description and Implementation for Avaya Communication Manager, 555-245-205.

Setting up Directory Buttons

Your Communication Manager integrated directory contains the names and extensions that are assigned on each **Station** screen. Display-telephone users can use a telephone button to access the directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory.

Note:

When you assign a name beginning with two tildes (~~) to a telephone, and **Display Character Set** on the **System Parameters Country-Options** screen is set to **Roman**, the name does not appear in the integrated directory. Note that this is the only way to hide a name in the integrated directory.

Instructions

We will assign directory telephone buttons for extension 2000. Our button assignment plan is set up so that telephone buttons 6, 7, and 8 are used for the directory. Remember, the name you type in the **Name** field on the first page of the **Station** screen is the name that appears when the integrated directory is accessed on a telephone display, except when the name is "hidden," as described in the Note above.

1. Type `change station 2000`. Press **Enter**.
2. Press **Next Page** to move to the **BUTTON ASSIGNMENTS** section on **Station screen (page 4)**.

Figure 68: Station screen

```
add station nnnn                                                    Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                          Headset? n
          Jack: ___                                              Speaker? n
         Cable: ___                                             Mounting: d
         Floor: _______                                     Cord Length: 0_
      Building: _______                                       Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                 6:limit-call
 2: call-appr                 7:team     Ext: 5381231         Rg:
 3: call-appr                 8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000      9:cfwd-enh Ext: 5502
 5: release                  10:aux-work RC: 1 Group:

voice-mail Number:
```

3. In **Button Assignment** field 6, type **directory**.
4. In **Button Assignment** field 7, type **next**.
5. In **Button Assignment** field 8, type **call-display**.
6. Press **Enter** to save your changes.

Chapter 7: Handling Incoming Calls

Setting up Basic Call Coverage

This chapter shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is not available.

Basic incoming call coverage:

- provides for automatic redirection of calls to alternate destinations when the called party is not available or not accepting calls
- provides the order in which Avaya Communication Manager redirects calls to alternate telephones or terminals
- establishes up to 6 alternate termination points for an incoming call
- establishes redirection criteria that govern when a call redirects
- redirects calls to a local telephone number (extension) or an off-switch telephone number (public network)

Call coverage allows an incoming call to redirect from its original destination to an extension, hunt group, attendant group, uniform call distribution (UCD) group, direct department calling (DDC) group, automatic call distribution (ACD) split, coverage answer group, Audio Information Exchange (AUDIX), or vector for a station not accepting calls.

Administering system-wide call coverage characteristics

This section shows you how to set up system-wide call coverage characteristics that govern how coverage is handled.

Instructions

The **System Parameters Call Coverage/Call Forwarding** screen sets up the global parameters which direct Communication Manager how to act in certain situations.

1. Leave all default settings as they are set for your system.
2. If you desire to customize your system, carefully read and understand each field description before you make any changes.

   See the **System Parameters Call Coverage/Call Forwarding** screen for more information.

Related topics

For more information on redirecting calls, see Covering calls redirected to an off-site location on page 220.

For information on setting the Caller Response Interval before a call goes to coverage, see "Caller Response Interval" in the Call Coverage section of *Feature Description and Implementation for Avaya Communication Manage*r, 555-245-205.

Creating coverage paths

This section explains how to administer various types of call coverage. In general, call coverage refers to what happens to incoming calls. You can administer paths to cover all incoming calls, or define paths for certain types of calls, such as calls to busy telephones. You can define where incoming calls go if they are not answered and in what order they reroute to other locations. For example, you can define coverage to ring the called telephone, then move to a receptionist if the call is not answered, and finally access a voice mailbox if the receptionist is not available.

With call coverage, the system redirects a call to alternate answering extensions when no one answers at the first extension. An extension can have up to 6 alternate answering points. The system checks each extension in sequence until the call connects. This sequence of alternate extensions is called a coverage path.

The system redirects calls based on certain criteria. For example, you can have a call redirect to coverage without ever ringing on the principal set, or after a certain number of rings, or when one or all call appearances (lines) are busy. You can set coverage differently for internal (inside) and external (outside) calls, and you can define coverage individually for different criteria. For example, you can decide that external calls to busy telephones can use the same coverage as internal calls to telephones with Do Not Disturb active.

Note:

If a call with a coverage path is redirected to a coverage point that is not available, the call proceeds to the next coverage point regardless of the type of coverage administered in the point that was unavailable. For example, if the unavailable coverage point has a hunt group coverage path administered, the hunt group coverage path would not be used by a call coming into the hunt group through the higher-level coverage path. The hunt group coverage path would be used only for calls coming directly into the hunt group extension.

Instructions

To create a coverage path:

1. Type **`add coverage path next`**. Press **Enter**.

   The system displays the **Coverage Path** screen.

Figure 69: Coverage path screen

```
change coverage path n                                          Page  1 of  x
                                COVERAGE PATH

            Coverage Path Number: n
                                        Hunt After Coverage: n
                Next Path Number: ___   Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?            n             n
              Busy?            y             y
      Don't Answer?            y             y Number of Rings:2
               All?            n             n
DND/SAC/Goto Cover?            y             y
  Holiday Coverage?            n             y  Holiday Table: 1

COVERAGE POINTS
Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: _________ Rng: Point2: _________Rng:
     Point3: _________ Rng: Point4: _________Rng:
     Point5: _________ Rng: Point6: _________Rng:
```

The system displays the next undefined coverage path in the sequence of coverage paths. Our example shows coverage path number 2.

2. Type a coverage path number in the **Next Path Number** field.

   The next path is optional. It is the coverage path to which calls are redirected if the current path's coverage criteria does not match the call status. If the next path's criteria matches the call status, it is used to redirect the call; no other path is searched.

3. Fill in the **Coverage Criteria** fields.

   You can see that the default sets identical criteria for inside and outside calls. The system sets coverage to take place from a busy telephone, if there is no answer after a certain number of rings, or if the **DND** (do not disturb), **SAC** (send all calls), or **Go to Cover** button has been pressed or corresponding feature-access codes dialed.

4. Fill in the **Point** fields with the extensions, hunt group number, or coverage answer group number you want for coverage points.

   Each coverage point can be an extension, hunt group, coverage answer group, remote number, or attendant.

5. Press **Enter** to save your changes.

Tip:

If you want to see which extensions or groups use a specific coverage path, type **`display coverage sender group n`**, where ***n*** is the coverage path number. For example, you should determine which extensions use a coverage path before you make any changes to it.

Assigning a coverage path to users

Now assign the new coverage path to a user. For example, we will assign this new coverage path to extension 2054.

Note:

A coverage path can be used for more than one extension.

Instructions

To assign a coverage path:

1. Type **`change station 2054`**. Press **Enter**.

   The system displays the **Station screen** for extension 2054.

Figure 70: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
                                              Time of Day Lock Table:
              Loss Group: 2             Personalized Ringing Pattern: 3
             Data Module? n                        Message Lamp Ext: 1014
            Speakerphone: 2-way                 Mute button enabled? y
        Display Language? English
                   Model:                            Expansion Module?

 Survivable GK Node Name:                           Media Complex Ext:
          Survivable COR:                                IP Softphone? y
   Survivable Trunk Dest?                           Remote Office Phone? y
                                                    IP Video Softphone?
                                                              IP Video?

                                                   Customizable Labels?
```

2. Type **2** in the **Coverage Path 1** field.

   To give extension 2054 another coverage path, you can type a coverage path number in the **Coverage Path 2** field.

3. Press **Enter** to save your changes.

Related information

Assigning Coverage Options on page 439

Setting up Advanced Call Coverage

Advanced incoming call coverage:

- redirects calls based on time-of-day
- allows coverage of calls that are redirected to sites not on the local server running Communication Manager
- allows users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables).

Covering calls redirected to an off-site location

You can provide coverage for calls that have been redirected to an off-site location (for example, your home). This capability, called Coverage of Calls Redirected Off-Net (CCRON) allows you to redirect calls onto the public network and bring back unanswered calls for further coverage processing.

Before you start:

- On the System Parameters Customer-Options (Optional Features) screen, verify the **Coverage of Calls Redirected Off-Net Enabled** field is **y**. If not, contact your Avaya representative.
- You need call classifier ports for all situations except ISDN end-to-end signaling, in which case the ISDN protocol does the call classification. For all other cases, use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack. See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207 for more information on the circuit pack.
  - Call Classifier - Detector circuit pack.

To provide coverage of calls redirected to an off-site location:

1. Type **`change system-parameters coverage-forwarding`**. Press **Enter**.
2. Press **Next Page** until you see the **Coverage of Calls Redirected Off-Net (CCRON)** page of the System-Parameters Coverage-Forwarding screen.

Figure 71: System Parameters - Call Coverage/Call Forwarding screen

```
 change system-parameters coverage-forwarding                     page 2 of x

           SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage Of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                       Ignore Network Answer Supervision? y
                      Disable call classifier for CCRON over ISDN trunks? n
                      Disable call classifier for CCRON over SIP trunks?  n
```

3. In the **Coverage of Calls Redirected Off-Net Enabled** field, type **y**.

   This instructs Avaya Communication Manager to monitor the progress of an off-net coverage or off-net forwarded call and provide further coverage treatment for unanswered calls.

4. In the **Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point** field, leave the default as **y**.
5. In the **Ignore Network Answer Supervision** field, leave the default as **n**.
6. Press **Enter** to save your changes.

Defining coverage for calls redirected to external numbers

You can administer the system to allow calls in coverage to redirect to off-net (external) or public-network numbers.

Standard remote coverage to an external number allows you to send a call to an external telephone, but does not monitor the call once it leaves your system. Therefore, if the call is busy or not answered at the external number, the call cannot be pulled back to the system. With standard remote call coverage, make the external number the last coverage point in a path.

With newer systems, you might have the option to use the Coverage of Calls Redirected Off-Net feature. If this feature is active and you use an external number in a coverage path, the system can monitor the call to determine whether the external number is busy or does not answer. If necessary, the system can redirect a call to coverage points that follow the external number. With this feature, you can have a call follow a coverage path that starts at the user’s extension, redirects to the user’s home telephone, and if not answered at home, returns to redirect to their voice mail box.

The call will not return to the system if the external number is the last point in the coverage path.

To use a remote telephone number as a coverage point, you need to define the number in the Remote Call Coverage Table and then use the remote code in the coverage path.

Instructions

For example, to add an external number to coverage path 2:

1. Type **change coverage remote**. Press **Enter**.

   The system displays the Remote Call Coverage Table screen.

Figure 72: Remote Call Coverage Table screen

```
change coverage remote

                                        REMOTE CALL COVERAGE TABLE
                                        ENTRIES FROM 1   TO 1000

01: ________________      16: ________________      31: ________________
02: ________________      17: ________________      32: ________________
03: ________________      18: ________________      33: ________________
04: ________________      19: ________________      34: ________________
05: ________________      20: ________________      35: ________________
06: ________________      21: ________________      36: ________________
07: ________________      22: ________________      37: ________________
08: ________________      23: ________________      38: ________________
09: ________________      24: ________________      39: ________________
10: ________________      25: ________________      40: ________________
11: ________________      26: ________________      41: ________________
12: ________________      27: ________________      42: ________________
13: ________________      28: ________________      43: ________________
14: ________________      29: ________________      44: ________________
15: ________________      30: ________________      45: ________________
```

2. Type **93035381000** in one of the remote code fields.

   If you use a digit to get outside of your network, you need to add the digit before the external number. In this example, the system requires a ‘9’ to place outside calls.

3. Be sure to record the remote code number you use for the external number.

   In this example, the remote code is r01.

4. Press **Enter** to save your changes.

5. Type **change coverage path 2**. Press **Enter**.

   The system displays the Coverage Path screen.

   Tip:

   Before making changes, you can use **display coverage sender group 2** to determine which extensions or groups use path 2.

Figure 73: Coverage Path screen

```
change coverage path n                                         Page  1 of  x
                                COVERAGE PATH

            Coverage Path Number: n
                                        Hunt After Coverage: n
                Next Path Number: ___   Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?           n             n
              Busy?           y             y
      Don't Answer?           y             y Number of Rings:2
               All?           n             n
DND/SAC/Goto Cover?           y             y
  Holiday Coverage?           n             y  Holiday Table: 1

COVERAGE POINTS

   Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: _________ Point2: _________ Point3: _________
     Point4: _________ Point5: _________ Point6: _________
```

6. Type **r1** in a coverage **Point** field.

   In this example, the coverage rings at extension 4101, then redirects to the external number. If you administer Coverage of Calls Redirected Off-Net and the external number is not answered or is busy, the call redirects to the next coverage point. In this example, the next point is Point 3 (h77 or hunt group 77).

   If you do not have the Coverage of Calls Redirected Off-Net feature, the system cannot monitor the call once it leaves the network. The call ends at the remote coverage point.

7. Press **Enter** to save your changes.

Related topics

For more information on coverage, see "Call Coverage" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Defining time-of-day coverage

The Time of Day Coverage Table on your system lets you redirect calls to coverage paths according to the time of day and day of the week when the call arrives. You need to define the coverage paths you want to use before you define the time of day coverage plan.

For example, let us say you want to administer the system so that incoming calls to extension 2054 redirect to a coworker in the office from 8:00 a.m. to 5:30 p.m., and to a home office from 5:30 p.m. to 8:00 p.m. on weekdays. You want to redirect the calls to voice mail after 8:00 p.m. weekdays and on weekends.

Instructions

To set up a time-of-day coverage plan that redirects calls for our example above:

1. Type **`add coverage time-of-day next`**. Press **Enter**.

   The system displays the Time of Day Coverage Table screen, and selects the next undefined table number in the sequence of time-of-day table numbers. If this is the first time-of-day coverage plan in your system, the table number is 1.

Figure 74: Time of Day Coverage Table screen

```
change coverage time-of-day 3
                   TIME OF DAY COVERAGE TABLE 1___

           Act    CVG      Act    CVG      Act    CVG      Act    CVG      Act    CVG
           Time   PATH     Time   PATH     Time   PATH     Time   PATH     Time   PATH

  Sun      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Mon      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Tue      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Wed      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Thu      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Fri      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
  Sat      00:00  ____     __:__  ____     __:__  ____     __:__  ____     __:__  ____
```

   Record the table number so that you can assign it to extensions later.

2. To define your coverage plan, enter the time of day and path number for each day of the week and period of time.

   Enter time in a 24-hour format from the earliest to the latest. For this example, assume that coverage path 1 goes to the coworker, path 2 to the home, and path 3 to voice mail.

   Define your path for the full 24 hours (from 00:01 to 23:59) in a day. If you do not list a coverage path for a period of time, the system does not provide coverage for that time.

3. Press **Enter** to save your changes.

Now assign the time-of-day coverage to a user. For example, we use extension 2054:

1. Type **`change station`** ***`nnnn`***, where ***`nnnn`*** is the extension number. Press **Enter**.

   The system displays the **Station screen**.

Figure 75: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
                                                  Time of Day Lock Table:
              Loss Group: 2             Personalized Ringing Pattern: 3
             Data Module? n                         Message Lamp Ext: 1014
            Speakerphone: 2-way                  Mute button enabled? y
        Display Language? English
                   Model:                           Expansion Module?

 Survivable GK Node Name:                          Media Complex Ext:
          Survivable COR:                               IP Softphone? y
   Survivable Trunk Dest?                        Remote Office Phone? y
                                                  IP Video Softphone?
                                                            IP Video?

                                                  Customizable Labels?
```

2. Move your cursors to Coverage Path 1 and type **t** plus the number of the Time of Day Coverage Table.

3. Press **Enter** to save your changes.

Now calls to extension 2054 redirect to coverage depending on the day and time that each call arrives.

Creating coverage answer groups

You can create a coverage answer group so that up to 8 telephones simultaneously ring when calls cover to the group. Anyone in the answer group can answer the incoming call.

Instructions

To add a coverage answer group:

1. Type **`add coverage answer-group next`**. Press **Enter**.

   The system displays the **Coverage Answer Group** screen.

Figure 76: Coverage Answer Group screen

```
change coverage answer-group n                                  Page 1 of 1
                        COVERAGE ANSWER GROUP

                      Group Number: 3___
                      Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ __________________________
2: ________ ___________________________ 6: ________ __________________________
3: ________ ___________________________ 7: ________ __________________________
4: ________ ___________________________ 8: ________ __________________________
```

2. In the **Group Name** field, enter a name to identify the coverage group.
3. In the **Ext** field, type the extension of each group member.
4. Press **Enter** to save your new group list.

   The system automatically completes the **Name** field when you press **Enter**.

Related topics

See Assigning a coverage path to users on page 218 for instructions on assigning a coverage path.

Setting up Call Forwarding

This section explains how to administer various types of automatic call forwarding. To provide call forwarding to your users, assign each extension a class of service (COS) that allows call forwarding. Then assign call-forwarding buttons to the user telephones (or give them the feature access code (FAC) for call forwarding) so that they can easily forward calls. Use the **Station** screen to assign the COS and any call-forwarding buttons.

Within each class of service, you can determine whether the users in that COS have the following call forwarding features:

- Call Forwarding All Calls — allows users to redirect all incoming calls to an extension, attendant, or external telephone number.
- Call Forwarding Busy/Don't Answer — allows users to redirect calls only if their extensions are busy or they do not answer.
- Restrict Call Fwd-Off Net — prevents users from forwarding calls to numbers that are outside your system network.

As the administrator, you can administer system-wide call-forwarding parameters to control when calls are forwarded. Use the **System Parameters Call Coverage/Call Forwarding** screen to set the number of times an extension rings before the system redirects the call because the user did not answer (CFWD No Answer Interval). For example, if you want calls to ring 4 times at an extension and, if the call is not answered, redirect to the forwarding number, set this parameter to 4.

You also can use the **System Parameters Call Coverage/Call Forwarding** screen to determine whether the forwarded-to telephone can override call forwarding to allow calls to the forwarded-from telephone (Call Forward Override). For example, if an executive forwards incoming calls to an attendant and the attendant needs to call the executive, the call can be made only if the **Call Forwarding Override** field is set to **y**.

Instructions

To determine which extensions have call forwarding activated:

1. Type `list call-forwarding`. Press **Enter**.

   This command lists all the extensions that are forwarded along with each forwarding number.

   Note:

   If you have a V1, V2, or V3 system, you can see if a specific extension is forwarded only by typing `status station` ***`nnnn`***, where ***`nnnn`*** is the specific extension.

Related topics

See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Setting up call forwarding for users

This section shows you how to give your users access to call forwarding.

Instructions

We will change a call forwarding access code from a local telephone with a Class of Service of 1:

1. Type **change feature-access-codes**. Press **Enter**.

   The system displays the Feature Access Code (FAC) screen.

Figure 77: Feature Access Code (FAC) screen

```
change feature-access-codes                                     Page 1 of x
                                   FEATURE ACCESS CODE (FAC)
             Abbreviated Dialing List1 Access Code: ____
             Abbreviated Dialing List2 Access Code: ____
             Abbreviated Dialing List3 Access Code: ____
    Abbreviated Dial - Prgm Group List Access Code: ____
                          Announcement Access Code: ____
                           Answer Back Access Code: ____
                             Attendant Access Code: ____
          Auto Alternate Routing (AAR) Access Code: ____
            Auto Route Selection ARS) Access Code1: ____   Access Code 2: ____
                     Automatic Callback Activation: ____    Deactivation: ____
    Call Forwarding Activation Busy/DA: ____    All: ____    Deactivation: ____
       Call Forwarding Enhanced Status: ____    Act: ____    Deactivation: ____
____                         Call Park Access Code: ____
                           Call Pickup Access Code: ____
    CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                       CDR Account Code Access Code: ____
                            Change COR Access Code: ____
                       Change Coverage Access Code: ____
                         Contact Closure Open Code: ____      Close Code: ____
                        Contact Closure Pulse Code: ____
```

2. In the **Call Forwarding Activation Busy/DA** field, type ***70**.

   The ***70** feature access code activates the call forwarding option so incoming calls forward when your telephone is busy or does not answer.

3. In the **Call Forwarding Activation All** field, type ***71**.

   The ***71** feature access code forwards all calls.

4. In the **Call Forwarding Deactivation** field, type **#72**.

   The **#72** feature access code deactivates the call forwarding option.

5. Press **Enter** to save your changes.

6. Type **`change cos`**. Press **Enter**.

   The system displays the **Class of Service** screen.

Figure 78: Class of Service screen

```
change cos                                                      Page 1 of x
                            CLASS OF SERVICE

                                    0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                   n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls              n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                    n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling                n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions             n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                  n n n n n n n n n n  n  n  n  n  n  n
    Client Room                     n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net       n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA            n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access (PSA)   n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All         n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA        n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations    n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion             n n n n n n n n n n  n  n  n  n  n  n
```

7. On the **Call Fwd-All Calls** line, in the **1** column, type **y**.

   This allows the user with this Class of Service to forward their calls. The **1** column is for telephones with a Class of Service of **1**.

8. On the **Console Permissions** line, in the **1** column, type **y**. This allows the user to define call forwarding on any station, not just the dialing station.

9. On the **Restrict Call Fwd-Off Net** line, in the **1** column, type **y**.

   This restricts your users from forwarding calls off-site. If you want your users to be able to call off-site, leave this field as **n**.

10. On the **Call Forward Busy/DA** line, in the **1** column, type **y**.

    This forwards a user's calls when the telephone is busy or doesn't answer after a programmed number of rings.

11. Press **1** to save your changes.

Allowing users to specify a forwarding destination

Now that you have set up system-wide call forwarding, have your users use this procedure if they want to change their call forwarding destination from their work (local) station.

1. They dial either their Call Forwarding Activation Busy/DA or Call Forwarding Activation All feature access code. If your users have buttons assigned, they press those buttons, listen for dial tone, and dial the digits.

   Note:

   Both Call Forwarding Activation Busy/DA or the Call Forwarding Activation All cannot be active for the same telephone at the same time.

   In this example, enter ***71** for **Call Forwarding Activation All**.

2. They dial their "forwarding-to" off-site or on-site number.

   In this example, enter **2081**. This is a local number; for off-site forwarding, include the AAR/ARS feature access code.

3. When they hear the 3-beep confirmation tone, they hang up.

Changing the forwarding destination remotely

Now that you have set up all of the required system administration for call forwarding, have your users use this procedure if they want to change their call forwarding destination from a telecommuting (off-site) telephone.

1. They dial their telecommuting extension.

   See Telecommuting Access on page 1637 for more information. In this example, enter **555-9126**.

2. When they get dial tone, they dial either their Extended Call Forward Activate Busy/DA or the Extended Call Forward Activate All feature access code.

   In this example, enter ***61** for the **Extended Call Forward Activate All** number.

3. When they get dial tone, they dial their extension number. Press the '**#**' key.

   In this example, enter **1014**, then **#**.

4. Even though there is no dial tone, they dial their security code. Press the '**#**' key.

   In this example, enter **4196**, then **#**.

5. When they get dial tone, they dial their "forwarding-to" off-site or on-site number.

   In this example, enter **9-555-2081**.

6. When they hear the 3-beep confirmation tone, they hang up.

Allowing users to change coverage remotely

This section shows you how to allow users to change their call coverage path from a local or telecommuting (off-site) telephone.

Instructions

To change call coverage from off-site:

1. Type **`change feature-access-codes`**. Press **Enter**.

   The system displays the **Feature Access Code (FAC)** screen.

2. In the **Change Coverage Access Code** field, type ***85**.

   Use the ***85** feature access code to change a coverage path from a telephone or remote station.

3. Press **Enter** to save your changes.

4. Type **`change cor`**. Press **Enter**.

   The system displays the **Class of Restriction** screen.

5. In the **Can Change Coverage** field, type **y**.

   This permits users to select one of two previously administered coverage paths.

6. Press **Enter** to save your changes.

7. Type **`change station 1014`**. Press **Enter**.

   The system displays the **Station** screen for extension 1014.

8. In the **Security Code** field, type **4196**.

   In this example, this is your security code. See Security-Related System Parameters on page 1455 for more information on setting the length of your security code.

9. In the **Coverage Path 1** and **Coverage Path 2** fields, verify that both are defined enabling your user to move from one coverage path to another.

   The **t1** and **t2** are the numbers of the Time of Day Coverage Tables.

10. Press **Enter** to save your changes.

Using Enhanced Call Forwarding

There are three types of Enhanced Call Forwarding:

- Use Enhanced Call Forwarding Unconditional to forward all calls
- Use Enhanced Call Forwarding Busy to forward calls when the user’s line is busy
- Use Enhanced Call Forwarding No Reply to forward calls when the user does not answer the call

The user can activate or deactivate any of these three types from their phone, and can specify different destinations for calls that are from internal and external sources. Users receive visual display and audio feedback on whether or not Enhanced Call Forwarding is active.

Display messages on the phone guide the user through the process of activating and de-activating Enhanced Call Forwarding, and for viewing the status of their forwarding.

Users can choose whether they want, at any one time, Call Forwarding or Enhanced Call Forwarding activated. The regular Call Forwarding feature (called "Classic Call Forwarding" to distinguish it from Enhanced Call Forwarding) continues to be available to users and has not changed.

Each of the three types of Enhanced Call Forwarding can have different destinations based on whether a call is internal or external. Therefore, six different destinations are possible to set up:

- Enhanced Call Forwarding Unconditional - internal
- Enhanced Call Forwarding Unconditional - external
- Enhanced Call Forwarding Busy - internal
- Enhanced Call Forwarding Busy - external
- Enhanced Call Forwarding No Reply - internal
- Enhanced Call Forwarding No Reply - external.

Each of these types of call forwarding can be activated either by feature access codes or by feature button.

When Enhanced Call Forwarding is deactivated, the destination number is kept. When the user activates Enhanced Call Forwarding again, the same destination number can be used without having to type it again.

When Enhanced Call Forwarding is not activated for a call, the call will go to a coverage path, if one has been set up.

Activating Enhanced Call Forwarding

To activate Enhanced Call Forwarding using a feature button:

1. Press the feature button labeled **cfwd-enh**

   The phone goes off hook.
2. Press 1 to activate Enhanced Call Forwarding.
3. Press
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 for Enhanced Call Forwarding No Reply
4. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
5. Dial the destination number to which calls will be forwarded.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the activation was successful.

To activate Enhanced Call Forwarding using a feature access code:

1. Press the feature access code for activating Enhanced Call Forwarding.

   The phone goes off hook.
2. Press:
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 for Enhanced Call Forwarding No Reply
3. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
4. Dial the destination number to which calls will be forwarded.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the activation was successful.

Deactivating Enhanced Call Forwarding

To deactivate Enhanced Call Forwarding using a feature button:

1. Press the feature button labeled **cfwd-enh**

   The phone goes off hook.
2. Press 2 to deactivate Enhanced Call Forwarding.
3. Press
   - 0 for all Enhanced Call Forwarding
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 to show the status for Enhanced Call Forwarding No Reply
4. Press
   - 1 for internal calls
   - 2 for external calls
   - 3 for all calls

   You hear a confirmation tone if the deactivation was successful.

To deactivate Enhanced Call Forwarding using a feature access code:

1. Press the feature access code for deactivating Enhanced Call Forwarding.

   The phone goes off hook.
2. Press
   - 0 to deactivate all Enhanced Call Forwarding
   - 1 to deactivate Enhanced Call Forwarding Unconditional
   - 2 to deactivate Enhanced Call Forwarding Busy
   - 3 to deactivate Enhanced Call Forwarding No Reply
3. Press
   - 1 for internal calls
   - 2 for external calls
   - 3 for all calls

   You hear a confirmation tone if the deactivation was successful.

Reactivating Enhanced Call Forwarding

To activate or reactivate Enhanced Call Forwarding using a feature button:

1. Press the feature button labeled **cfwd-enh**

   The phone goes off hook.
2. Press 1 to reactivate Enhanced Call Forwarding
3. Press
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 to show the status for Enhanced Call Forwarding No Reply
4. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
5. Optionally, dial the destination number to which calls will be forwarded.

   If you do not enter a destination number, the previous destination number will be used.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the action was successful.

To reactivate Enhanced Call Forwarding using a feature access code:

1. Press the feature access code for activating Enhanced Call Forwarding.

   The phone goes off hook.
2. Press
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
3. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
4. Optionally, dial the destination number to which calls will be forwarded.

   If you do not enter a destination number, the previous destination number will be used.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the action was successful.

Displaying Enhanced Call Forwarding Status

To display Enhanced Call Forwarding status using a feature button:

1. Press the feature button labeled **cfwd-enh**

   The phone goes off hook.

2. Press 3 to display status.

   Your phone will display the status of the different types of Enhanced Call Forwarding.

To display Enhanced Call Forwarding status using a feature access code:

1. Press the feature access code for displaying Enhanced Call Forwarding status.

   The phone goes off hook.

2. Press 3 to display status.

   You phone will display the status of the different types of Enhanced Call Forwarding.

Activating and deactivating Enhanced Call Forwarding from an off-network phone

To activate Enhanced Call Forwarding from an off-network phone:

1. Dial the remote access number, including barrier code or authentication code.
2. Press the feature access code for activating Enhanced Call Forwarding.
3. Press:
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 for Enhanced Call Forwarding No Reply
4. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
5. Dial the forwarding station extension.
6. Dial the destination number to which calls will be forwarded.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the activation was successful.

To deactivate Enhanced Call Forwarding from an off-network phone:

1. Dial the remote access number, including barrier code or authentication code.
2. Press the feature access code for deactivating Enhanced Call Forwarding.

3. Press:
   - 0 for all Enhanced Call Forwarding
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 for Enhanced Call Forwarding No Reply
4. Press
   - 1 for internal calls
   - 2 for external calls
   - 3 for all calls
5. Dial the forwarding station extension.

   You hear a confirmation tone if the activation was successful.

Activating and deactivating Enhanced Call Forwarding from a phone with console permission

To activate Enhanced Call Forwarding from a phone with console permission:

1. Press the feature access code for activating Enhanced Call Forwarding.

   The phone goes off hook.
2. Press:
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy
   - 3 for Enhanced Call Forwarding No Reply
3. Press
   - 1 to forward internal calls
   - 2 to forward external calls
   - 3 to forward all calls
4. Dial the forwarding station extension.
5. Dial the destination number to which calls will be forwarded.

   Dial # at the end of an external destination number, or wait for the timeout to expire.

   You hear a confirmation tone if the activation was successful.

To deactivate Enhanced Call Forwarding from a phone with console permission:

6. Press the feature access code for activating Enhanced Call Forwarding.

   The phone goes off hook.

7. Press:
   - 0 for all Enhanced Call Forwarding
   - 1 for Enhanced Call Forwarding Unconditional
   - 2 for Enhanced Call Forwarding Busy

Setting up Night Service

You can use night service to direct calls to an alternate location when the primary answering group is not available. For example, you can administer night service so that anyone in your marketing department can answer incoming calls when the attendant is at lunch or has left for the day.

Once you administer night service to route calls, your end-users merely press a button on the console or a feature button on their telephones to toggle between normal coverage and night service.

There are five types of night service:

- Night Console Night Service — directs all attendant calls to a night or day/night console
- Night Station Night Service — directs all incoming trunk or attendant calls to a night service destination
- Trunk Answer from Any Station (TAAS) — directs incoming attendant calls and signals a bell or buzzer to alert other employees that they can answer the calls
- Trunk Group Night Service — directs incoming calls to individual trunk groups to a night service destination
- Hunt Group Night Service — directs hunt group calls to a night service destination

Setting up night station service to voice mail

The night station service (also known as Listed Directory Number (LDN) Night Service) sends calls directed to an LDN to voice mail when the system is in night service.

Instructions

What is described below is a common setup; however, you can use a regular extension in this field, but it will not follow coverage.

Note:
You can use a dummy hunt group (one with no members) or an exported station with a coverage path. The instructions below use a hunt group.

To set up a night station service to voice mail:

1. Type **`add hunt-group next`**. Press **Enter**.

   The system displays the [Hunt Group](#) screen.

Figure 79: Hunt Group screen

```
add hunt-group next                                              Page 1 of X
                                      HUNT GROUP

           Group Number: 4__                                        ACD? _
             Group Name: ____________________________             Queue? y
                                                             Queue Limit:____
        Group Extension: ________                                Vector? _
             Group Type: _______                          Coverage Path: ____
                     TN: ___                   Night Service Destination: ____
                    COR: _                              MM Early Answer? _
          Security Code: ____                     Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___    Port: x____  Extension: ____
 Time Warning Threshold: ___    Port: x____  Extension: ____
```

   The **Group Number** field fills automatically with the next hunt group number.

2. In the **Group Name** field, type the name of the group.

   In our example, type **ldn nights**. There should be no members in this hunt group.

3. Press **Enter** to save your changes.

   Note:

   If you are using tenant partitioning, the command for the next step will be **`change tenant x`**. If you are using tenant partitioning, the **Night Destination** field does not appear on the **Listed Directory Numbers** screen. Instead, it is on the **Tenant** screen.

4. Type **`change listed-directory-numbers`**. Press **Enter**.

   The system displays the [Listed Directory Numbers](#) screen.

Figure 80: Listed Directory Numbers screen

```
change listed-directory-number                              Page   1 of   2
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext      Name                         TN
      1:
      2:                                       1
      3:                                       1
      4:                                       1
      5:                                       1
      6:                                       1
      7:                                       1
      8:                                       1
```

5. In the **Night Destination** field, add the night destination on the listed directory telephone.

   In our example, type **51002**.
6. Press **Enter** to save your changes.
7. Type **`change console-parameters`**. Press **Enter**.

   The system displays the [Console Parameters](#) screen.

Figure 81: Console Parameters screen

```
change console-parameters                                     Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                COR: 0
       Calls in Queue Warning: 5                 Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                          Night Service Act. Ext.:
                 IAS (Branch)? n          IAS Tie Trunk Group No.:
         IAS Att. Access Code:              Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

8. In the **DID-LDN Only** to **LDN Night Ext** field, type **n**.
9. Press **Enter** to save your changes.

10. From a telephone with console permissions, dial the call forwarding feature access code, then the hunt group's extension, followed by the main number of AUDIX.

    In our example, dial 51002.

    Note:

    You should receive the confirmation tone (3 beeps). This step is very important as calls to the LDN night service extension do not follow coverage.

11. In voice mail, build your auto attendant with the extension of the Listed Directory Number, not the hunt group.

    The originally dialed number was the LDN. That is what Communication Manager passes to the voice mail. In the case of the INTUITY and newer embedded AUDIX Voice Mail systems, you can use the Auto Attendant routing table to send the calls to a common Auto Attendant mailbox.

Setting up night console service

This section shows you how to set up night console service.

Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console.

Note:

Activating night console service also puts trunk groups into night service, except those for which a night service button has been administered. See Setting up trunk answer from any stationSetting up trunk answer from any station on page 244 for more information.

To activate and deactivate Night Console Service, the attendant typically presses the **Night** button on the principal attendant console or designated console.

Only the principal console can activate night service. In the absence of any console, a telephone can activate night service.

Instructions

We will put the attendant console (attendant 2) in a night service mode.

To set up Night Console Service:

1. Type **`change attendant 2`**. Press **Enter**.

   The system displays the **Attendant Console** screen.

Figure 82: Attendant Console screen

```
change attendant n                                           Page   1 of  x
                             ATTENDANT CONSOLE 1

         Type: console           Name: 27 character attd cons name
    Extension: 1000             Group: 1               Auto Answer: none
 Console Type: principal           TN: 1               Data Module? y
         Port: 01C1106            COR: 1         Disp Client Redir? n
Security Code:                    COS: 1          Display Language: english
                                                 H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote             Local Remote             Local Remote
 1: 9                    5:                        9:
 2: 82                   6:                       10:
 3:                      7:                       11:
 4:                      8:                       12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

2. In the **Console Type** field, type **principal**.

   There can be only one night-only or one day/night console in the system unless you administer Tenant Partitioning. Night Service is activated from the principal console or from the one station set per-system that has a **nite-serv** button.

3. Press **Enter** to save your changes.

Setting up night station service

You can use night station service if you want to direct incoming trunks calls, DID-LDN (direct inward dialing-listed directory number) calls, or internal calls to the attendant (dialed 'O' calls) to a night service destination.

Let us say your attendant, who answers extension (LDN) 8100, usually goes home at 6:00 p.m. When customers call extension 8100 after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

To set up night station service, you need to record the announcement (in our example, it is recorded at announcement extension 1234). See Managing Announcements on page 511 for information on setting up the announcement.

Tip:

All trunk groups that are routed through the attendant direct to this night service destination provided they already do not have a night service destination and, on the **Console Parameters** screen, the **DID-LDN Only to DID-LDN Night Ext** field is **n**. See Setting up trunk answer from any station on page 244.

Instructions

To set up night station service:

1. Type **`change listed-directory-numbers`**. Press **Enter**.

   The system displays the **Listed Directory Numbers** screen.

Figure 83: Listed Directory Number screen

```
change listed-directory-number                                Page   1 of   2
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext      Name                         TN
      1:
      2:                                       1
      3:                                       1
      4:                                       1
      5:                                       1
      6:                                       1
      7:                                       1
      8:                                       1
```

2. Enter **1234** in the **Night Destination** field.

   The destination can be an extension, a recorded announcement extension, a vector directory number, or a hunt group extension.

3. Press **Enter** to save your changes.

4. Type **`change console-parameters`**. Press **Enter**.

   The system displays the **Console Parameters** screen.

Figure 84: Console Parameters screen

```
change console-parameters                                         Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                    COR: 0
       Calls in Queue Warning: 5                      Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                             Night Service Act. Ext.:
                IAS (Branch)? n              IAS Tie Trunk Group No.:
         IAS Att. Access Code:                 Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

5. In the **DID-LDN Only to LDN Night Extension** field, type **n**.
6. Press **Enter** to save your changes.

   After you set up night station service, have the attendant use the night console button to activate and deactivate night service.

Setting up trunk answer from any station

There might be situations where you want everyone to be able to answer calls when the attendant is away. Use trunk answer any station (TAAS) to configure the system so that it notifies everyone when calls are ringing. Then, you can give users the trunk answer any station feature access code so they can answer these calls.

When the system is in night service mode, attendant calls redirect to an alerting device such as a bell or a buzzer. This lets other people in the office know when they should answer the telephone.

Note:

If no one answers the call, the call will not redirect to night service.

We will define a feature access code (we'll use 71) and configure the alerting device for trunk answer any station.

You need a ringing device and 1 port on an analog line circuit pack. See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207, for more information on the circuit pack.

To set the feature access code for TAAS:

1. Type **change feature-access-codes**. Press **Enter**.

   The system displays the **Feature Access Code (FAC)** screen.

2. Click **Next** until you see the **Trunk Answer Any Station Access Code** field.

Figure 85: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page 4 of x
                             FEATURE ACCESS CODE (FAC)
                       Station Lock Activation:       Deactivation:
      Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                 Remove:
  Station User Button Ring Control Access Code:
                          Telephone Activation: #*
             Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:       Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
           Whisper Page Activation Access Code:
```

3. In the **Trunk Answer Any Station Access Code** field, type **71**.

4. Press **Enter** to save your changes.

Once you set the feature access code, determine where the external alerting device is connected to the Communication Manager server (we'll use port 01A0702).

To set up external alerting:

1. Type **`change console-parameters`**. Press **Enter**.

   The system displays the **Console Parameters** screen.

Figure 86: Console Parameters screen

```
change console-parameters                                       Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
                                                Night Service Act. Ext.:
                IAS (Branch)? n                 IAS Tie Trunk Group No.:
         IAS Att. Access Code:                    Alternate FRL Station:
             Backup Alerting? n           DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

2. In the **EXT Alert Port (TAAS)** field, type **01A0702**.

   Use the port address assigned to the external alerting device.

3. Press **Enter** to save your changes.

Setting up external alerting night service

Calls redirected to the attendant via Call Forwarding or Call Coverage will not go to the LDN Night Station. If there is no night station specified, and the TAAS bell is being used, these calls ring the TAAS bell. A call following the coverage path rings the TAAS bell for the number of times indicated in the Coverage Don't Answer Interval for Subsequent Redirection (Rings) field. If not answered, the call proceeds to the next point in the station's coverage path. If the call was sent to the Attendant by Call Forwarding, it continues to ring the TAAS bell.

When night service is enabled, and there is a night service destination on the **Listed Directory Numbers** screen, calls covering to the attendant attempt to ring the night destination instead of the attendant position even if the handset is plugged in.

Instructions

To send LDN calls to the attendant during the day and to a guard's desk at night:

1. Type **`change listed-directory-numbers`**. Press **Enter**.

   The system displays the **Listed Directory Numbers** screen.

Figure 87: Listed Directory Numbers screen

```
change listed-directory-number                              Page   1 of   2
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext     Name                            TN
      1:
      2:                                         1
      3:                                         1
      4:                                         1
      5:                                         1
      6:                                         1
      7:                                         1
      8:                                         1
```

2. In the **Night Destination** field, verify this field is blank.
3. Press **Enter** to save your changes.
4. Type **`change console-parameters`**. Press **Enter**.

   The system displays the **Console Parameters** screen.

Figure 88: Console Parameters screen

```
change console-parameters                                     Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                     Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
                                              Night Service Act. Ext.:
                  IAS (Branch)? n             IAS Tie Trunk Group No.:
         IAS Att. Access Code:                  Alternate FRL Station:
              Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

5. In the **EXT Alert Port (TAAS)** field, type **01A0702**.

   This is the port address assigned to the external alerting device.

6. Press **Enter** to save your changes.

   The system is in Night Service.

   Any calls to extension 2000 now go to extension 3000 (the guard's desk).

   Any "0" seeking calls go to extension 3000 (the guard's desk).

To send LDN calls to the attendant during the day and to the TAAS bell at night:

1. Type **`change console-parameters`**. Press **Enter**.

   The system displays the **Console Parameters** screen.

Figure 89: Console Parameters screen

```
change console-parameters                                         Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                   COR: 0
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                                Night Service Act. Ext.:
                 IAS (Branch)? n                IAS Tie Trunk Group No.:
         IAS Att. Access Code:                    Alternate FRL Station:
              Backup Alerting? n          DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

2. In the **DID-LDN Only to Night Ext**. field, type **y**.

   This allows only listed directory number calls (LDN) to go to the listed directory night service number extension.

3. In the **Ext Alert Port (TAAS)** field, type **01A070**.

   This is the port address assigned to the external alerting device.

4. Press **Enter** to save your changes.

   The system is in night service.

   Any DNIS extension 2000 calls now go to the TAAS bell.

   Any "0" seeking calls now go to the TAAS bell.

Setting up trunk group night service

You can use trunk group night service if you want to direct individual trunk groups to night service. The system redirects calls from the trunk group to the group's night service destination.

Trunk group night service overrides night station service. For example, we will say you activate trunk group night service, and then your attendant activates night station service. In this case, calls to the trunk group use the trunk night service destination, rather than the station night service destination.

Instructions

We will direct night calls for trunk group 2 to extension 1245.

To set up trunk group night service:

1. Type **`change trunk-group 2`**. Press **Enter**.

   The system displays the Trunk Group screen.

Figure 90: Trunk Group screen

```
add trunk-group next                                            Page   1 of  x
                                TRUNK GROUP

Group Number: 1                  Group Type: co          CDR Reports: y
  Group Name: OUTSIDE CALL              COR: 1        TN: 1        TAC:
   Direction: two-way      Outgoing Display? n
 Dial Access? n              Busy Threshold: 255      Night Service:
Queue Length: 0          Country: 1            Incoming Destination:
   Comm Type: voice               Auth Code? n   Digit Absorption List:
    Prefix-1? y                 Trunk Flash? n         Toll Restricted? y

         Trunk Type:
```

2. Type **1245** in the **Night Service** field.

   The destination can be a station extension, a recorded announcement extension, a vector directory number, a hunt group extension, a terminating extension group, or attd if you want to direct the call to the attendant.

3. Press **Enter** to save your changes.

Setting up night service for hunt groups

You can administer hunt group night service if you want to direct hunt group calls to a night service destination.

Let us say your helpline on hunt group 3 does not answer calls after 6:00 p.m. When customers call after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

Instructions

To set up night service for your helpline, you need to record the announcement (in our example, the announcement is on extension 1234) and then modify the hunt group to send calls to this extension.

To administer the hunt group for night service:

1. Type **`change hunt-group 3`**. Press **Enter**.

   The system displays the **Hunt Group** screen for hunt group 3.

Figure 91: Hunt Group screen

```
change hunt-group n                                             Page 1 of X
                                    HUNT GROUP

           Group Number: n__                                        ACD? _
             Group Name: ____________________________             Queue? y
                                                            Queue Limit:____
        Group Extension: ________                                Vector? _
             Group Type: _______                          Coverage Path: ____
                     TN: ___                   Night Service Destination: ____
                    COR: _                              MM Early Answer? _
          Security Code: ____                     Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___   Port: x____  Extension: ____
 Time Warning Threshold: ___   Port: x____  Extension: ____
```

2. In the **Night Service Destination** field, type **1234**.

   The destination can be an extension, a recorded announcement extension, a vector directory number, a hunt group extension, or attd if you want to direct calls to the attendant.

   Calls to hunt group 3 will follow the coverage path assigned to extension 1234.

3. Press **Enter** to save your changes.
4. Now you need to program a night service button.

   See Adding Feature Buttons on page 129 for more information.

Related topics

See Managing Hunt Groups on page 274.

How do night service types interact?

Here is an example of how several types of night service might be used in one company.

Assume that you already administered the following night service settings:

- Night station night service redirects to extension 3000 and DID-LDN only to LDN Night Ext is set to **n**
- EXT Alert Port (TAAS) field is not defined
- Trunk group 4 redirects to extension 2000

Let us look at how calls for this company are directed after hours:

call type	directs to
An LDN call on a DID trunk	extension 3000
A call on trunk group 4	extension 2000
An internal call to ‘0’	extension 3000
A call that redirects to the attendant through a coverage path	the attendant queue

Adding Call Pickup

Users might need to answer a call that is ringing at a nearby desk. With Communication Manager, a user can answer a call that is ringing at another telephone in three ways:

- Use Call Pickup. With Call Pickup, you create one or more pickup groups. A pickup group is a collection, or list, of individual telephone extensions. A pickup group is the way to connect individual extensions together. For example, if you want everyone in the payroll department to be able to answer calls to any other payroll extension, you can create a pickup group that contains all of the payroll extensions.

  A user extension can belong to only one pickup group. Also, the maximum number of pickup groups might be limited by your system configuration.

  Using their own telephones, all members in a pickup group can answer a call that is ringing at another group member telephone. If more than one telephone is ringing, the system selects the extension that has been ringing the longest.

- Use Extended Call Pickup. With Extended Call Pickup, you can define one or more extended pickup groups. An extended pickup group is the way to connect individual pickup groups together.

  There are two types of extended pickup groups: simple and flexible. You administer the type of extended pickup groups on a system-wide basis. You cannot have both simple and flexible extended pickup groups on your system at the same time.

  Based on the type of extended pickup group that you administer, members in one pickup group can answer calls to another pickup group.

  For more information, see Setting up simple extended pickup groups on page 262, Setting up flexible extended pickup groups on page 265, and Changing extended pickup groups on page 269.

- Use Directed Call Pickup. With Directed Call Pickup, users specify what ringing telephone they want to answer. A pickup group is not required with Directed Call Pickup. You must first administer Directed Call Pickup before anyone can use this feature.

  For more information, see Setting up Directed Call Pickup on page 270.

Throughout this procedure on pickup groups and extended pickup groups, we show examples to make Call Pickup easier to understand.

Call Pickup Alerting

Members of a call pickup group know that another group member is receiving a call in two ways:

- Group members can hear the other telephone ring.
- The Call Pickup button status lamp on the telephones of all the group members flash.

Note:

You must activate Call Pickup Alerting in your system, and assign a Call Pickup button to the telephones of each pickup group member, before the Call Pickup button status lamps work properly.

For information how to set up Call Pickup Alerting, see Enabling Call Pickup Alerting on page 257.

If the **Call Pickup Alerting** field on the **Feature-Related System Parameters** screen is set to **n** (see Figure 93), members of the call pickup group must rely only on ringing to know when another group member receives a call. Pickup group members must be located close enough that they can hear the ringing of the other telephones.

To answer a call, a pickup group member can either press the Call Pickup button on the telephone, or dial the Call Pickup feature access code (FAC). For more information, see Assigning a Call Pickup button to a user telephone on page 258, and Assigning a Call Pickup feature access code on page 258.

The **Call Pickup Alerting** field on the **Feature-Related System Parameters** screen determines how the Call Pickup button status lamps operate.

- If the **Call Pickup Alerting** field is set to **n**, the Call Pickup Button status lamps on all pickup group member telephones do not flash when a call comes in. When a pickup group member hears the telephone of another group member ring and presses the Call Pickup button to answer the call, the:
    - Call Pickup button status lamp of the answering group member becomes steadily lit for the duration of the call.
    - Telephone of the called group member stops ringing.
- If the **Call Pickup Alerting** field is set to **y**, the Call Pickup Button status lamps on all pickup group member telephones flash when a call comes in. When a pickup group member sees the Call Pickup button status lamp flash and presses the Call Pickup button to answer the call, the:
    - Call Pickup button status lamp of the answering group member goes out for the duration of the call.
    - Call Pickup button status lamp of the called group member goes out.
    - Call Pickup button status lamps of the other pickup group members go out.
    - Telephone of the called group member stops ringing.

If another call comes into the pickup group while a group member is talking on a pickup call, the member on the pickup call cannot answer the incoming call. If the group member who is already on a pickup call attempts to answer the incoming call to another telephone using Call Pickup, the Call Pickup button status lamp "flutters" for a few seconds. The fluttering status lamp indicates denial of service. The call that the member is currently on is not interrupted.

In all scenarios, the call appearance button on the telephone of the called group member:

- Stays steadily lit if the **Temporary Bridged Appearance on Call Pickup?** field on the **Feature-Related System Parameters** screen is set to **y**. (For an example of this field, see Figure 93: Feature-Related System Parameters screen on page 257.) The called group member can join the call in progress by pressing the lit call appearance button. The person who picked up the call can either stay on the call or hang up.
- Goes out if the **Temporary Bridged Appearance on Call Pickup?** field on the **Feature-Related System Parameters** screen is set to **n**. The called group member cannot join the call in progress.

The system uses an algorithm to select what call is answered when multiple calls ring or alert in a call pickup group at the same time. The system searches the extensions of the call pickup group until the system finds an extension with a call that is eligible to be answered with Call Pickup. The system selects this call to be answered. The next time that a group member answers a call with Call Pickup, the system bypasses the extension that was answered most recently, and starts the search at the next extension.

For example, if a group member attempts to use Call Pickup when two calls are ringing at extension A and one call is ringing at extension B, the system selects the calls in the following order:

1. One of the calls to extension A
2. The call to extension B
3. The remaining call to extension A

The system also determines which call that a group member answers when multiple calls ring or alert at the same telephone. The system selects the call with the lowest call appearance, which is usually the call appearance that is nearest to the top of the telephone.

For example, when calls ring or alert at the second and the third call appearances, the system selects the call on the second call appearance for the user to answer.

The following steps are part of the administration process for the Call Pickup feature:

- Administering Call Pickup
  - Setting up Call Pickup
  - Enabling Call Pickup Alerting
  - Assigning a Call Pickup button to a user telephone
  - Assigning a Call Pickup feature access code
- Maintaining Call Pickup
  - Removing a user from a call pickup group
  - Deleting pickup groups
  - Changing a Call Pickup button on a user telephone
  - Removing a Call Pickup button from a user telephone

- Administering Extended Call Pickup
  - Setting up simple extended pickup groups
  - Setting up flexible extended pickup groups
- Maintaining Extended Call Pickup
  - Removing a pickup group from an extended pickup group
  - Changing extended pickup groups
- Administering Directed Call Pickup
  - Setting up Directed Call Pickup
- Maintaining Directed Call Pickup
  - Removing Directed Call Pickup from a user

Setting up Call Pickup

The first step in setting up any call pickup system is to create pickup groups and assign users to the groups. You can create one or many pickup groups, depending on your needs. A user extension can belong to only one pickup group.

In this exercise, you will:

- Add a pickup group and assign users to the pickup group.
- Enable Call Pickup alerting.
- Assign a Call Pickup button to each extension in the pickup group.
- Assign a feature access code (FAC).

Adding pickup groups

To add users to a new call pickup group:

1. Type **`add pickup-group next`**. Press **Enter**.

   The system displays the **Pickup Group** screen. The system also assigns the next available **Group Number** for the new pickup group (see Figure 92: Pickup Group screen on page 256).

Figure 92: Pickup Group screen

```
add pickup-group next                                                Page 1 of 2
                                    PICKUP GROUP

           Group Number: 11
             Group Name: Accounting
GROUP MEMBER ASSIGNMENTS

    Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
 1: 5430892  Jane Doe                   14:
 2: 5439711  Frank Smith                15:
 3: 5432783  Bonnie Franklin            16:
 4: 5438590  Cathy Miller               17:
 5: 5436602  Jeffrey Dingle             18:
 6: 5438843  Roger Greenspan            19:
 7:                                     20:
 8:                                     21:
 9:                                     22:
10:                                     23:
11:                                     24:
12:                                     25:
13:
```

Note:

The **Extended Group Number** field is not shown in this example because the system is set for **none** or **simple** extended pickup groups. For more information, see Setting up simple extended pickup groups on page 262. If the **Extended Group Number** field is visible on this screen (see Figure 96: Pickup Group screen on page 266), then your system is set up for **flexible** extended pickup groups. For more information, see Setting up flexible extended pickup groups on page 265.

2. Type a name for this pickup group in the **Group Name** field.
3. Type the extension of each group member.

   Up to 50 extensions can belong to one pickup group.
4. Press **Enter** to save your changes.

   The system automatically completes the **Name** field when you press **Enter**.

Example - This procedure shows how to set up a new pickup group 11 for Accounting. For the rest of these procedures, let us say that you also set up these pickup groups:

- 12 for Billing
- 13 for Credit Services
- 14 for Delinquency Payments
- 15 for Executives
- 16 for Finance

Enabling Call Pickup Alerting

Call Pickup Alerting allows members of pickup groups to know visually when the telephone of another member is ringing. Use Call Pickup Alerting if the telephones of other pickup group members are too far away to be heard. You must enable Call Pickup Alerting in your system.

To enable Call Pickup alerting:

1. Type **change system-parameters features**. Press **Enter**.

   The system displays the **Feature-Related System Parameters** screen.

2. Click **Next** until you see the **Call Pickup Alerting** field (Figure 93: Feature-Related System Parameters screen on page 257).

Figure 93: Feature-Related System Parameters screen

```
change system-parameters features                                    Page 4 of 16
                              FEATURE-RELATED SYSTEM PARAMETERS

                            Reserved Slots for Attendant Priority Queue: 5
                                             Time before Off-hook Alert: 10
                                 Emergency Access Redirection Extension:
                   Number of Emergency Calls Allowed in Attendant Queue: 5
                                                   Call Pickup Alerting? y
                            Temporary Bridged Appearance on Call Pickup? y
                                          Call Pickup on Intercom Calls? y
                                                   Directed Call Pickup? y
                                             Extended Group Call Pickup: simple
                      Deluxe Paging and Call Park Timeout to Originator? n
   Controlled Outward Restriction Intercept Treatment: tone
  Controlled Termination Restriction (Do Not Disturb): tone
            Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS        Authorization Codes Enabled? y
                                       Authorization Code Length: 7
                          Authorization Code Cancellation Symbol: #
                                            Attendant Time Out Flag? n
                                         Display Authorization Code? y
                 Controlled Toll Restriction Replaces: none
```

3. Set the **Call Pickup Alerting** field to **y**.

4. Press **Enter** to save your changes.

Assigning a Call Pickup button to a user telephone

After you define one or more pickup groups, assign a Call Pickup button for each extension in each pickup group. Users in a pickup group can press the assigned Call Pickup button to answer calls to any other extension in their pickup group.

To assign a Call Pickup button for each extension:

1. Type **`change station n`**, where ***n*** is an extension in the pickup group. Press **Enter**.

   The system displays the **Station** screen.
2. Click **Next** until you see the **BUTTON ASSIGNMENTS** area.
3. Move to the button number that you want to use for Call Pickup. You can use any of the available buttons.
4. Type **call-pkup** after the button number.
5. Press **Enter** to save your changes.
6. Repeat this procedure for each member of each pickup group.
7. Notify each user what button to use for Call Pickup.

Assigning a Call Pickup feature access code

After you define one or more pickup groups, assign and give each member the Call Pickup feature access code (FAC). Instead of using the Call Pickup button, users in a pickup group can dial the assigned FAC to answer calls to any other extension in their pickup group.

To create a pickup group feature access code (FAC):

1. Type **`change feature-access-codes`**. Press **Enter**.

   The system displays the Feature Access Code (FAC) screen.
2. In the **Call Pickup Access Code** field, type the desired FAC.

   Make sure that the FAC complies with your dial plan.
3. Press **Enter** to save your changes.
4. Communicate the FAC with each member of each pickup group.

Removing a user from a call pickup group

To remove a user from a pickup group:

1. Type **`change pickup-group n`**, where ***n*** is the number of the pickup group. Press **Enter**.

   The system displays the **Pickup Group** screen (Figure 92: Pickup Group screen on page 256).

2. Move to the extension that you want to remove.
3. Press **Clear** or **Delete**, depending on your system.
4. Press **Enter** to save your changes.

Deleting pickup groups

Before deleting a pickup group, you must verify if the pickup group is a member of any simple or flexible extended pickup group. If so, you must first delete the pickup group from all extended pickup groups. For more information on extended pickup groups, see Setting up simple extended pickup groups on page 262, and Setting up flexible extended pickup groups on page 265.

Follow these three steps to delete a pickup group:

- Get a list of all extended pickup groups.
- Verify and delete the pickup group from all extended pickup groups.
- Delete the pickup group.

Getting a list of extended pickup groups

To get a list of all extended pickup groups:

1. Type **`list extended-pickup-group`**. Press **Enter**.

   The system displays the **Extended Pickup Groups** screen (see Figure 94: Extended Pickup Groups screen on page 260).

Figure 94: Extended Pickup Groups screen

```
list extended-pickup-group

                           EXTENDED PICKUP GROUPS

                            Group           Number of
                            Number          Members

                            15              3
```

2. Print this screen or write down the existing Group Numbers so that you can check each extended pickup group.
3. Press **Cancel**.

Removing a pickup group from an extended pickup group

You must remove the pickup group from all extended pickup groups.

- If your system is set up for **simple** extended pickup groups, the pickup group can be a member of only one extended pickup group.
- If your system is set up for **flexible** extended pickup groups, the pickup group can be a member of many extended pickup groups.
- If your system is set up for no extended pickup groups (**none**) or has no extended pickup groups assigned, you can skip this section and see Deleting a pickup group.

To remove a pickup group from an extended pickup group:

1. Type **`change extended-pickup-group n`**, where ***n*** is the extended pickup group that you want to check. Press **Enter**.

   The system displays the **Extended Pickup Group** screen.
2. Perform one of the following actions:
   - If the pickup group that you want to delete is *not* a member of this extended pickup group, press **Cancel**.
   - If the pickup group that you want to delete is a member of this extended pickup group:
     - Select the pickup group.
     - Press **Clear** or **Delete**, depending on your system.
     - Press **Enter** to save your changes.
3. Repeat this procedure for each extended pickup group.

Deleting a pickup group

To delete a pickup group:

1. Type `remove pickup-group` ***n***, where ***n*** is the number of the pickup group that you want to delete. Press **Enter**.

   The system displays the **Pickup Group** screen.
2. Press **Enter**. The system deletes the pickup group.

Changing a Call Pickup button on a user telephone

To change a call pickup button on a user telephone:

1. Type `change station` ***n***, where ***n*** is the extension that you want to change. Press **Enter**.

   The system displays the **Station** screen.
2. Click **Next** until you see the **BUTTON ASSIGNMENTS** area.
3. Move to the existing **call-pkup** button.
4. Press **Clear** or **Delete**, depending on your system.
5. Move to the button number that you want to use for call pickup.
6. Type **call-pkup** after the button number.
7. Press **Enter** to save your changes.

Removing a Call Pickup button from a user telephone

To remove a call pickup button from a user telephone:

1. Type `change station` ***n***, where ***n*** is the extension that you want to change. Press **Enter**.

   The system displays the **Station** screen.
2. Click **Next** until you see the **BUTTON ASSIGNMENTS** area.
3. Move to the existing **call-pkup** button.
4. Press **Clear** or **Delete**, depending on your system.
5. Press **Enter** to save your changes.

Setting up simple extended pickup groups

What if you want to have members in one pickup group be able to answer calls for another pickup group? In our example, what if you want members in the Credit Services pickup group 13 to answer calls in the Delinquency Payments pickup group 14? You can do that by setting up extended pickup groups.

If you want members of pickup group 13 to answer calls for pickup group 14, *and* if you want members of pickup group 14 to answer calls for pickup group 13, set your system for simple extended pickup groups.

Simple extended pickup groups allow members of two or more individual pickup groups to answer each others calls. In a simple extended pickup group, an individual pickup group can be assigned to only one extended pickup group.

All members of one pickup group can answer the calls to the other pickup groups within the simple extended pickup group.

CAUTION:
Before you administer what type of extended pickup group to use (none, simple, or flexible), be sure that your pickup group objectives are well thought out and defined.

In this exercise, you will:

- Set up the system for simple extended pickup groups.
- Assign a FAC so that users can answer calls.
- Add pickup groups, if needed
- Assign two pickup groups to an extended pickup group.

Creating simple extended pickup groups

To create system-wide simple extended pickup groups:

1. Type **`change system-parameters features`**. Press **Enter**.

   The system displays the **Feature-Related System Parameters** screen.
2. Click **Next** until you see the **Extended Group Call Pickup** field.
3. In the **Extended Group Call Pickup** field, type **simple**.
4. Press **Enter** to save your changes.

   Your system is now set up for simple extended pickup groups.

Creating an extended pickup group feature access code

Users in an extended pickup group must dial an assigned FAC, followed by a 1-digit or 2-digit Pickup Numbers, to answer calls to an extension in another pickup group. Pickup groups must be in the same extended pickup group. Users cannot use a call pickup button with Extended Call Pickup.

To create an extended pickup group feature access code (FAC):

1. Type **`change feature-access-codes`**. Press **Enter**.

   The system displays the **Feature Access Code (FAC)** screen.
2. Click **Next** until you see the **Extended Group Call Pickup Access Code** field.
3. Perform one of the following actions:
   - If the **Extended Group Call Pickup Access Code** field contains a FAC**,** press **Cancel**.
   - If the **Extended Group Call Pickup Access Code** field does not contain a FAC:
     - Type the desired FAC.

       Make sure that the FAC complies with your dial plan.
     - Press **Enter** to save your changes.
4. Communicate the FAC, the list of pickup numbers, and the pickup group to which each pickup number is associated, to each pickup group member who is part of the extended pickup group. For information on pickup numbers, see Pickup Numbers on page 264.

To create individual pickup groups:

1. If you need to create any pickup groups, see Setting up Call Pickup on page 255.

Assigning pickup groups to a simple extended pickup group

To assign pickup groups to a simple extended pickup group:

1. Type **`change extended-pickup-group`** ***`n`***, where ***`n`*** is a number of the extended pickup group. In this example, type **`change extended-pickup-group 4`**. Press **Enter**.

   The system displays the **Extended Pickup Group** screen for extended pickup group 4 (see Figure 95: Extended Pickup Group screen on page 264).

Figure 95: Extended Pickup Group screen

```
change extended-pickup-group 4                                   Page 1 of 1
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 4

Pickup   Pickup Group                            Pickup   Pickup Group
Number     Number                                Number     Number

 0:      13                                      13:      _____
 1:      14                                      14:      _____
 2:      _____                                   15:      _____
 3:      _____                                   16:      _____
 4:      _____                                   17:      _____
 5:      _____                                   18:      _____
 6:      _____                                   19:      _____
 7:      _____                                   20:      _____
 8:      _____                                   21:      _____
 9:      _____                                   22:      _____
10:      _____                                   23:      _____
11:      _____                                   24:      _____
12:      _____
```

2. In the **Pickup Group Number** column, type the numbers of the pickup groups that you want to link together. In this example, add pickup group 13 (Credit Services) and pickup group 14 (Delinquency Payments).
3. Press **Enter** to save your changes.

Example - Pickup groups 13 and 14 are now linked together in extended pickup group 4. In addition to answering calls to their own pickup group:

- All members of pickup group 13 can answer calls to pickup group 14.
- All members of pickup group 14 can answer calls to pickup group 13.

Pickup Numbers

The **Pickup Number** column that is associated with the **Pickup Group Number** is the unique number that users must dial after dialing the **Extended Group Call Pickup Access Code** FAC to answer a call in that pickup group.

For example, let us say that the **Extended Group Call Pickup Access Code** FAC is ***39**. In the above example:

- A user in pickup group 13 must dial ***391** to answer a call to pickup group 14, because pickup group 14 is assigned to **Pickup Number 1**.
- A user in pickup group 14 must dial ***390** to answer a call to pickup group 13, because pickup group 13 is assigned to **Pickup Number 0**.

Note:

To minimize the number of digits that a user has to dial, first assign pickup groups to **Pickup Numbers** 0 to 9.

- By assigning **Pickup Numbers** 0 to 9, all users only needs to dial a single digit (**0** to **9**) after the FAC to answer the call.
- If you assign a number greater than 9 (10 to 24) to *any* pickup group, *all* users must dial two digits (**00** to **24**) after the FAC to answer the call.

Setting up flexible extended pickup groups

If you want members of a pickup group to answer calls for another pickup group, but you *do not* want the other pickup group to answer your calls, set your system for flexible extended pickup groups.

Flexible extended pickup groups still allow members of one or more individual pickup groups to answer calls of another pickup group. However, the reverse scenario is not always true. With flexible extended pickup groups, you can prevent members of one or more pickup groups from answering the calls to another pickup group.

Flexible extended pickup groups allows more control over what pickup groups can answer calls for other pickup groups. Unlike simple extended pickup groups, an individual pickup group can be in multiple flexible extended pickup groups.

The system displays the **Extended Group Number** field on the **Pickup Group** screen only when you set the **Extended Group Call Pickup** field on the **Feature-Related System Parameters** screen to **flexible**. When you populate the **Extended Group Number** field on the **Pickup Group** screen, you are associating, or "pointing," that pickup group to an extended pickup group. By pointing to an extended pickup group, members of the pickup group can answer calls made to any member of that extended pickup group.

A specific pickup group does *not* have to be a member of the extended pickup group that the pickup group points to. To help clarify flexible extended pickup groups, see the Example in this section.

CAUTION:

Before you administer what type of extended pickup group to use (none, simple, or flexible), be sure that your pickup group objectives are well thought out and defined.

In this exercise, you will:

- Set up the system for flexible extended pickup groups.
- Assign a FAC so that users can answer calls.
- Add or change pickup groups, and "point" a pickup group to an extended pickup group.

Creating flexible extended pickup groups

To create system-wide flexible extended pickup groups:

1. Type **`change system-parameters features`**. Press **Enter**.

   The system displays the **Feature-Related System Parameters** screen.

2. Click **Next** until you see the **Extended Group Call Pickup** field (see Figure 348: Feature-Related System Parameters screen on page 1017).

3. In the **Extended Group Call Pickup** field, type **flexible**.

4. Press **Enter** to save your changes.

   Your system is now set up for flexible extended pickup groups.

To create an extended pickup group FAC, see Creating an extended pickup group feature access code on page 263.

To associate individual pickup groups with an extended pickup group:

1. Type **`change pickup-group`** ***`n`***, where ***`n`*** is a pickup group number. In this example, let us change pickup group 15 (Executives). Type **`change pickup-group 15`**. Press **Enter**.

   The system displays the **Pickup Group** screen (see Figure 96: Pickup Group screen on page 266). Notice that the system displays the **Extended Group Number** field on the **Pickup Group** screen. This field appears because you set the **Extended Group Call Pickup** field on the **Feature-Related System Parameters** screen to **flexible**.

Figure 96: Pickup Group screen

```
change pickup-group 15                                        Page 1 of 2
                                  PICKUP GROUP

           Group Number: 15         Extended Group Number: 4
             Group Name: Executives
GROUP MEMBER ASSIGNMENTS

    Ext       Name (first 26 characters)      Ext       Name (first 26 characters)
 1: 5431010   James Martin                 14:
 2: 5439711   Susan Crawford-Smith         15:
 3:                                        16:
 4:                                        17:
 5:                                        18:
 6:                                        19:
 7:                                        20:
 8:                                        21:
 9:                                        22:
10:                                        23:
11:                                        24:
12:                                        25:
13:
```

Important:

If you change your system from simple to flexible extended pickup groups (see Changing extended pickup groups on page 269), the system automatically populates the **Extended Group Number** field on the **Pickup Group** screen for each pickup group member. For example, pickup groups 13 and 14 are members of extended pickup group 4. If you change the system from simple to flexible extended pickup groups, the system automatically populates the **Extended Group Number** field to **4** on the **Pickup Group** screen for these two pickup groups.

You are not required to keep the number that the system automatically populates in the **Extended Group Number** field. You can change the number in the **Extended Group Number** field to another pickup group number. You can also make the field blank.

2. If you want to associate, or "point" the pickup group to an extended pickup group, type the number of the extended pickup group for which this pickup group can answer calls in the **Extended Group Number** field. In this example, manually associate pickup group 15 (Executives) to extended pickup group 4. For this example, let us say that you followed the same procedure for pickup group 16 (Finance).

   Note:

   You do not have to populate the **Extended Group Number** field. You can leave the **Extended Group Number** field blank. You can just as easily point the pickup group to a different extended pickup group. For example, you can point pickup group 13 (Credit Services) to extended pickup group 2, even though pickup group 13 is not a member of extended pickup group 2.

   As you will see in the Example on page 268, this means that members in pickup group 13 can answer calls to any member that is in extended pickup group 2. The reverse is not true, however. Members that are in extended pickup group 2 cannot answer calls to pickup group 13.

3. Press **Enter** to save your changes.

Assigning pickup groups to a flexible extended pickup group

To assign pickup groups to a flexible extended pickup group:

1. Type **`change extended-pickup-group n`**, where ***n*** is the number of the extended pickup group. In this example, type **`change extended-pickup-group 4`**. Press **Enter**.

   The system displays the **Extended Pickup Group** screen for extended pickup group 4 (see Figure 97: Extended Pickup Group screen on page 268).

Figure 97: Extended Pickup Group screen

```
change extended-pickup-group 4                              Page   1 of  1
                                EXTENDED PICKUP GROUP

                              Extended Group Number: 4

Pickup    Pickup Group                              Pickup    Pickup Group
Number      Number                                  Number      Number

 0:       13                                        13:       _____
 1:       14                                        14:       _____
 2:       16                                        15:       _____
 3:       _____                                     16:       _____
 4:       _____                                     17:       _____
 5:       _____                                     18:       _____
 6:       _____                                     19:       _____
 7:       _____                                     20:       _____
 8:       _____                                     21:       _____
 9:       _____                                     22:       _____
10:       _____                                     23:       _____
11:       _____                                     24:       _____
12:       _____
```

2. Add pickup group 16 (Finance) to this extended pickup group. For information how to complete this screen, see Figure 95: Extended Pickup Group screen on page 264.
3. Press **Enter** to save your changes.

Example - Here is how flexible extended pickup groups work.

Notice that pickup groups 13, 14, and 16 are now members of extended pickup group 4. On the **Pickup Group** screen for pickup groups 13, 14, and 16, you also pointed each pickup group to extended pickup group 4.

Pickup group 15 (Executives) is *not* a member of extended pickup group 4. However, on the **Pickup Group** screen for group 15 (Figure 96: Pickup Group screen on page 266), you pointed pickup group 15 to extended pickup group 4.

In addition to answering calls to their own pickup group:

Notice that pickup groups 13, 14, and 16 are now members of extended pickup group 4. On the **Pickup Group** screen for pickup groups 13, 14, and 16, you also pointed each pickup group to extended pickup group 4.

Pickup group 15 (Executives) is *not* a member of extended pickup group 4. However, on the **Pickup Group** screen for group 15 (Figure 96), you pointed pickup group 15 to extended pickup group 4.

In addition to answering calls to their own pickup group:

- Any member of pickup group 13 can answer calls to pickup groups 14 and 16.
- Any member of pickup group 14 can answer calls to pickup groups 13 and 16.
- Any member of pickup group 16 can answer calls to pickup groups 13 and 14.
- Any member of pickup group 15 can answer calls to pickup groups 13, 14, and 16 because pickup group 15 points to extended pickup group 4.
- Any member of pickup groups 13, 14 and 16 *cannot* answer calls to pickup group 15 because pickup group 15 is not a member of extended pickup group 4.

Changing extended pickup groups

You define extended pickup groups on a system-wide basis. The system cannot support both simple and flexible extended pickup groups at the same time. You can, however, change your extended pickup groups from one type to another.

Changing from simple to flexible

If you want to change all extended pickup groups from simple to flexible, you can easily make the change. See Creating flexible extended pickup groups on page 266. The system automatically populates the **Extended Group Number** field on the **Pickup Group** screen for all pickup groups that are part of an extended pickup group.

Changing from flexible to simple

The process is more complex to change all extended pickup groups from flexible to simple. Before you can change the extended pickup group from flexible to simple, you must first delete all of the individual pickup groups from all of the extended pickup groups. Then you can change the extended pickup group from flexible to simple (see Creating simple extended pickup groups on page 262). After that step, you must re-administer all of the extended pickup groups again.

Setting up Directed Call Pickup

If you do not want to set up pickup groups and extended pickup groups, but still want selected people to answer other telephones, use Directed Call Pickup. Before a person can use this feature, you must enable Directed Call Pickup on your system.

- Telephones that can be answered by another extension using Directed Call Pickup must have a Class of Restriction (COR) that allows this feature.
- Telephones that can answer another extension using Directed Call Pickup must have a COR that allows this feature.

In this exercise, you will:

- Determine if Directed Call Pickup is enabled on your system.
- Create one or more Classes of Restriction (COR) that allow Directed Call Pickup.
- Assign the COR to individual extensions.
- Assign a Directed Call Pickup button to each extension that is assigned the COR.
- Assign a feature access code (FAC).

Before you can assign Directed Call Pickup to a user, you must ensure that Directed Call Pickup is available on your system.

To ensure that Directed Call Pickup is enabled on your system:

1. Type **`change system-parameters features`**. Press **Enter**.

   The system displays the **Feature-Related System Parameters** screen.
2. Click **Next** until you see the **Directed Call Pickup?** field (see Figure 348: Feature-Related System Parameters screen on page 1017).
3. Perform one of the following actions:
   - If the **Directed Call Pickup?** field is set to **y**, your system is set up for Directed Call Pickup. Press **Cancel**.
   - If the **Directed Call Pickup?** field is set to **n**:
     - Type **y** in the field.
     - Press **Enter** to save your changes.

Creating Classes of Restriction for Directed Call Pickup

You must create one or more Classes of Restriction (COR) that allow Directed Call Pickup. All users to whom you assign a COR can then use Directed Call Pickup.

There are three ways to set up a COR for Directed Call Pickup. You can create a COR where users can:

- Only have their extensions answered by Directed Call Pickup. Users with this COR cannot pick up other extensions.
- Only pick up other extensions using Directed Call Pickup. Users with this COR cannot have their extensions answered by other users.
- Both have their extensions answered by Directed Call Pickup and pick up other extensions.

To create a COR that allows Directed Call Pickup:

1. Type **`change COR n`**, where ***n*** is the COR that you want to change. Press **Enter**.

   The system displays the **Class of Restriction screen (page 1)** screen.

2. Perform one of the following actions:

   - To create one or more CORs where the extensions can only be picked up by the Directed Call Pickup feature, but not be able to pick up other extensions:
     - Type **y** in the **Can Be Picked Up By Directed Call Pickup?** field.
     - Leave the **Can Use Directed Call Pickup?** field set to **n**.

       Any extension to which you assign this COR can only be picked up by the Directed Call Pickup feature.
   - To create one or more CORs where the extensions can only use the Directed Call Pickup feature to pick up other extensions, but not be picked up by other extensions:
     - Leave the **Can Be Picked Up By Directed Call Pickup?** field set to **n**.
     - Type **y** in the **Can Use Directed Call Pickup?** field.

       Any extension to which you assign this COR can only use the Directed Call Pickup feature to pick up other extensions.
   - To create one or more CORs where the extensions can use the Directed Call Pickup feature both to pick up other extensions and be picked up by other extensions:
     - Type **y** in the **Can Be Picked Up By Directed Call Pickup?** field.
     - Type **y** in the **Can Use Directed Call Pickup?** field.

       Any extension to which you assign this COR can use the Directed Call Pickup feature both to pick up other extensions and be picked up by other extensions.

3. Press **Enter** to save your changes.

For more information on Class of Restriction (COR), see the "Class of Restriction" feature.

Assigning a Class of Restriction to a user

You must assign a COR to user extensions before anyone can use Directed Call Pickup.

To modify an extension to allow Directed Call Pickup:

1. Type `change station n`, where *n* is the extension that you want to change. Press **Enter**.

   The system displays the **Station** screen.
2. In the **COR** field, type the appropriate COR that allows Directed Call Pickup capabilities.
3. Press **Enter** to save your changes.

Assigning a Directed Call Pickup button

Assign a Directed Call Pickup button to all extensions that share a COR where the **Can Use Directed Call Pickup?** field is set to **y**.

To assign a Directed Call Pickup button for each extension in the COR:

1. Type `change station n`, where *n* is an extension to which you have assigned the Directed Call Pickup COR. Press **Enter**.

   The system displays the **Station** screen.
2. Click **Next** until you see the **BUTTON ASSIGNMENTS** area.
3. Move to the button number that you want to use for Directed Call Pickup. You can use any of the available buttons.
4. Type **dir-pkup** after the button number.
5. Press **Enter** to save your changes.
6. Repeat this procedure for each member of the COR who can pick up other extensions using Directed Call Pickup.
7. Notify each user what button to use for Directed Call Pickup.

Assigning a Directed Call Pickup feature access code

Also assign a Directed Call Pickup feature access code (FAC). Give the FAC to each user whose extension shares a COR where the **Can Use Directed Call Pickup?** field is set to **y**.

Instead of using the Directed Call Pickup button, users can dial the assigned FAC to answer calls using Directed Call Pickup.

To create a Directed Call Pickup feature access code (FAC):

1. Type `change feature-access-codes`. Press **Enter**.

   The system displays the **Feature Access Code (FAC)** screen.
2. Click **Next** until you see the **Directed Call Pickup Access Code** field.

3. Perform one of the following actions:
   - If the **Directed Call Pickup Access Code** field already contains a code**,** press **Cancel**.
   - If the **Directed Call Pickup Access Code** field does not contain a code:
     - Type a code in the field. Make sure that the code you type conforms to your dial plan.
     - Press **Enter** to save your change.
4. Communicate the FAC with each member of the COR that can pick up other extensions using Directed Call Pickup.

Removing Directed Call Pickup from a user

To remove Directed Call Pickup from a user:

1. Type **`change station`** ***`n`***, where ***`n`*** is the extension of the user. Press **Enter**.

   The system displays the **Station** screen.
2. In the **COR** field, type a different COR that does not have Directed Call Pickup permissions.
3. Click **Next** until you see the **BUTTON ASSIGNMENTS** section.
4. Move to the button number that contains **dir-pkup**.
5. Click **Clear** or **Delete**, depending on your system.
6. Press **Enter** to save your changes.

Managing Hunt Groups

This section shows you how to set up hunt groups. This section explains how calls to a hunt group are handled, and shows you different call distribution methods.

What are hunt groups?

A hunt group is a group of extensions that receive calls according to the call distribution method you choose. When a call is made to a certain telephone number, the system connects the call to an extension in the group.

Use hunt groups when you want more than one person to be able to answer calls to the same number. For example, set up a hunt group for:

- a benefits department within your company
- a travel reservations service

Setting up hunt groups

Let us set up a hunt group for an internal helpline. Before making changes to Communication Manager, we'll decide:

- the telephone number for the hunt group
- the number of people answering calls
- the way calls are answered

Our dial plan allows 4-digit internal numbers that begin with 1. The number 1200 is not in use. So, we'll set up a helpline hunt group so anyone within the company can call extension 1200 for help with a telephone.

We will assign 3 people (agents) and their extensions to our helpline. We want calls to go to the first available person.

Instructions

To set up our helpline hunt group:

1. Type **`add hunt-group next`**. Press **Enter**.

   The system displays the Hunt Group screen. The **Group Number** field is automatically filled in with the next hunt group number.

Figure 98: Hunt Group screen

```
change hunt-group n                                                 Page 1 of X
                                      HUNT GROUP

           Group Number: 4__                                           ACD? _
             Group Name: ____________________________                Queue? y
                                                               Queue Limit:____
        Group Extension: ________                                   Vector? _
             Group Type: _______                             Coverage Path: ____
                     TN: ___                        Night Service Destination: ____
                    COR: _                                 MM Early Answer? _
          Security Code: ____                       Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___    Port: x____   Extension: ____
 Time Warning Threshold: ___    Port: x____   Extension: ____
```

2. In the **Group Name** field, type the name of the group.

   In our example, type **internal helpline**.

3. In the **Group Extension** field, type the telephone number.

   We'll type **1200**.

4. In the **Group Type** field, type the code for the call distribution method you choose.

   We'll type **ucd-loa** so a call goes to the agent with the lowest percentage of work time since login.

   Note:

   The COS for all hunt groups defaults to 1. Therefore, any changes to COS 1 on the **Class of Service** screen changes the COS for all your hunt groups. A **COS** field does not appear on the **Hunt Group** screen.

5. Press **Next Page** to find the **Group Member Assignments** screen.

Figure 99: Group Member Assignments screen

```
change hunt-group n                                                 Page   4 of  xx
HUNT GROUP
          Group Number: 4 Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 1500   Administered Members (min/max): 1 /9
                                       Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)            Ext       Name (24
characters)

 1 :1011                               14 :
 2 :1012                               15 :
 3 :1013                               16 :
 4 :                                   17 :
 5 :                                   18 :
 6 :                                   19 :
 7 :                                   20 :
 8 :                                   21 :
 9 :                                   22 :
10 :                                   23 :
11 :                                   24 :
12 :                                   25 :
13 :                                   26 :

More Members Exist
```

6. In the **Ext** field, type the extensions of the agents you want in the hunt group.

   We'll type **1011**, **1012**, and **1013**.

   Tip:

   For a ddc group type (also known as "hot seat" selection), the call is sent to the extension listed in the first **Ext** field. The system uses this screen to determine the hunting sequence.

7. Press **Enter** to save your changes.

   The **Name** fields are display-only and do not appear until the next time you access this hunt group.

Dynamic hunt group queue slot allocation

The dynamic hunt group queue slot allocation feature eliminates the need to preallocate queue slots for hunt groups. The system dynamically allocates the queue slots from a common pool on an as-needed basis. All possible calls can be queued. There is no additional administration needed. This feature expands the capacities of your system by eliminating the potential of missed calls due to a full queue.

When the **Queue?** field on the **Hunt Group** screen is set to **y**, this feature applies to all uses of hunt groups:

- Automatic Call Distribution (ACD) non-vector/vector splits and skills
- Non-ACD hunt group
- Voice mail

Related topics

See Hunt Group on page 1119 for more information on an ACD and non-ACD hunt group.

Changing a hunt group

To make changes to a hunt group:

1. Type **`change hunt-group n`**, where ***n*** is the number of the hunt group. Press **Enter**.
2. Change the necessary fields.
3. Press **Enter** to save your changes.

Setting up a queue

You can tell your server running Communication Manager how to handle a hunt-group call when it cannot be answered right away. The call waits in "queue."

We will tell Communication Manager that as many as 10 calls can wait in the queue, but that you want to be notified if a call waits for more than 30 seconds.

You also want Communication Manager to send a warning when 5 or more calls are waiting in the queue. This warning flashes queue-status buttons on telephones that have a status button for this hunt group. When the buttons flash, everyone answering these calls can see that the help-line calls need more attention.

Instructions

To set up our helpline queue:

1. Type **`change hunt-group n`**, where ***n*** is the number of the hunt group to change. Press **Enter**.

   In our example, type **`change hunt-group 5`**. The system displays the Hunt Group screen.

Figure 100: Hunt Group screen

```
change hunt-group n                                              Page 1 of X
                                      HUNT GROUP

           Group Number: 4__                                        ACD? _
             Group Name: ____________________________             Queue? y
                                                            Queue Limit:____
        Group Extension: ________                                Vector? _
             Group Type: _______                          Coverage Path: ____
                     TN: ___                    Night Service Destination: ____
                    COR: _                               MM Early Answer? _
          Security Code: ____                      Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___   Port: x____  Extension: ____
 Time Warning Threshold: ___   Port: x____  Extension: ____
```

2. In the **Queue** field, type **y**.
3. In the **Queue Length** field, type the maximum number of calls that you want to wait in the queue.

   In our example, type **10**.
4. In the **Calls Waiting Threshold** field, type the maximum number of calls that can be in the queue before the system flashes the queue status buttons.

   In our example, type **5**.
5. In the **Time Warning Threshold** field, type the maximum number of seconds you want a call to wait in the queue before the system flashes the queue status buttons.

   In our example, type **30**.
6. Press **Enter** to save your changes.

Setting up hunt groups for TTY callers

Several laws, such as the Americans with Disabilities Act (ADA) of 1990 and Section 255 of the Telecommunications Act of 1996, require that "reasonable accommodation" be provided for people with disabilities. For this reason, your company might choose to offer support for callers who use TTYs. (These devices are also known as TDDs -- "Telecommunication Device for the Deaf" -- but the term TTY is generally preferred, in part because many users of these devices are hearing-impaired, but not deaf.)

TTY callers can be accommodated by creating a hunt group that includes TTY-equipped agents. The TTY itself looks a little like a laptop computer, except that it has a one- or two-line alphanumeric display instead of a computer screen. The cost of a typical TTY is approximately three hundred dollars. Although many TTYs can connect directly with the telephone network via analog RJ-11 jacks, Avaya recommends that agents be equipped with TTYs that include an acoustic coupler that can accommodate a standard telephone handset. One reason for this recommendation is that a large proportion of TTY users are hearing impaired, but still speak clearly. These individuals often prefer to receive calls on their TTYs and then speak in response. This requires the call center agent to alternate between listening on the telephone and then typing on the TTY, a process made considerably easier with an acoustically coupled configuration.

Although TTY-emulation software packages are available for PCs, most of these do not have the ability to intermix voice and TTY on the same call.

For a TTY hunt group, you can record TTY announcements and use them for the hunt group queue. To record announcements for TTY, simply follow the same steps as with voice recordings from your telephone (see Managing Announcements on page 511). However, instead of speaking into your telephone to record, you type the announcement with the TTY device.

Note:

For an alternative to simply creating a TTY hunt group, you can use vectors to process TTY calls. With vectors, you can allow TTY callers and voice callers to use the same telephone number. In this case, you can also record a single announcement that contains both TTY signaling and a voice recording.

Adding hunt group announcements

You can add recorded announcements to a hunt group queue. Use announcements to encourage callers to stay on the line or to provide callers with information. You can define how long a call remains in the queue before the caller hears an announcement.

For more information on how to record an announcement, see "Announcements" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Let us add an announcement to our internal helpline. We want the caller to hear an announcement after 20 seconds in the queue, or after approximately 4 or 5 rings. Our announcement is already recorded and assigned to extension 1234.

Tip:

You can use **`display announcements`** to find the extensions of your recorded announcements.

Instructions

To add an announcement to our helpline queue:

1. Type **`change hunt-group n`**. Press **Enter**, where n is the number of the hunt group to change.

   In our example, type **`change hunt-group 5`**.

   The system displays the **Hunt Group** screen.

2. Press **Next Page** to find the **First Announcement Extension** field.

Figure 101: Hunt Group screen

```
                                                              Page 2 of X
                               HUNT GROUP

                          Message Center: ________
                         AUDIX Extension: ________
               Message Center AUDIX Name: ______
                                Primary? _
     Calling Party Number to INTUITY AUDIX? _
                           LWC Reception: _______
                              AUDIX Name: _______
                    Messaging Server Name: _______

             First Announcement Extension: 1234 ___ Delay (sec): 20
            Second Announcement Extension: ________ Delay (sec): __ Recurring? _
```

3. In the **First Announcement Extension** field, type the extension of the announcement you want callers to hear.

   In this example, type **1234**.

4. In the **First Announcement Delay (sec)** field, type the number of seconds you want the caller to wait before hearing the first announcement.

   In our example, type **20**.

   Tip:

   If you set the delay announcement interval to 0, callers automatically hear the announcement before anything else. This is called a "forced first announcement."

5. Press **Enter** to save your changes.

You can use the same announcement for more than one hunt group.

Managing Vectors and VDNs

This section provides an introduction to vectors and Vector Directory Numbers (VDN). It gives you basic instructions for writing simple vectors.

SECURITY ALERT:

Vector fraud is one of the most common types of toll fraud because vectors route calls based on the Class of Restriction (COR) assigned to the VDN. See *BCS Products Security Handbook*, 555-025-600 for more information.

This section references announcements, hunt groups, queues, splits, and skills, which are covered in detail in other sections of this book. You can also find information about these topics in *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Note:

The **Client Room** field on the **Class of Service** screen will affect VDN displays. If a local station that has a COS with the **Client Room** field set to **y** calls a local VDN, the agent's display that receives the call will look as if it is a direct station call rather than the expected VDN display of "station name to vdn name."

What are vectors?

A vector is a series of commands that you design to tell the system how to handle incoming calls. A vector can contain up to 32 steps and allows customized and personalized call routing and treatment. Use call vectoring to:

- play multiple announcements
- route calls to internal and external destinations
- collect and respond to dialed information

Tip:

The vector follows the commands in each step in order. The vector "reads" the step and follows the command if the conditions are correct. If the command cannot be followed, the vector skips the step and reads the next step.

Your system can handle calls based on a number of conditions, including the number of calls in a queue, how long a call has been waiting, the time of day, day of the week, and changes in call traffic or staffing conditions.

Writing vectors

Writing vectors is easy, but Avaya recommends that you set up and test your vectors before you use them across the system.

We'll write a vector to handle calls to our main number. It is the first vector so we'll use number 1.

Tip:

Use `list vector` to see a list of existing vectors.

Before you start:

- On the **System Parameters Customer-Options (Optional Features)** screen, ensure that the **Basic Call Vectoring** field is **y**. If not, contact your Avaya representative.
- To provide announcements, you need an Announcement circuit pack. For more information on the circuit pack, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.
- Use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack.
  - Call Classifier - Detector circuit pack.
- Note on adding Meet-Me Conference vectors: If the vector for Meet-Me Conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller might not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

To write a vector:

1. Type `change vector 1`. Press **Enter**.

   The system displays the **Call Vector** screen.

Figure 102: Call Vector screen

```
change vector n                                                    Page 1 of x
                                CALL VECTOR
   Number: nnnn          Name: _______________

Multimedia? n     Attendant Vectoring? n     Meet-me Conf? y            Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 ____________
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

The vector **Number** field on the left side of the screen is filled in automatically.

2. In the **Name** field, type a description for the vector.

   In our example, type **main number calls**.

   Tip:

   The information in the heading of the **Call Vector** screen is display only. Use the **`display system-parameters customer-options`** command to see the features that are turned on for your Communication Manager server.

3. Type your vector steps in the numbered column on the left of the screen.

   Tip:

   When you type in your vector steps, Communication Manager automatically completes some of the vector step information for you. For example, if you type "q" in a vector step field, it fills in "queue-to." Also, additional fields appear when you complete a field. Press **Tab**. This makes it very easy to type in your vector steps.

Now that vector 1 is set up, we will add a vector step to it to tell Communication Manager how to handle the calls to our main number.

Putting a call in a queue

Write a vector so that calls that come into the main business number redirect to a queue.

We will use a vector-controlled hunt group for the main number queue. This hunt group was set up as main split 47. When calls first arrive, all calls to our main number should be queued as "pri 1" for low priority.

To queue calls, write the following vector (step 2). (Please note, we started our example on step 2 because step 1 is used later in this chapter.)

Figure 103: Call Vector screen

```
change vector 1                                                   page 1 of x
                                   CALL VECTOR
   Number:  1       Name: main number calls
                  Attendant Vectoring? n     Meet-me Conf? y            Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 queue-to main split 47 pri 1
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Tip:

Remember, Communication Manager automatically fills in some of the information when you type your vector step. Press **Tab**.

Playing an announcement

Write a vector to play an announcement for callers in a queue. Use the announcement to ask callers to wait. You need to record the announcement before the vector can use it.

Let us play our announcement 4001, asking the caller to wait, then play music for 60 seconds, then repeat the announcement and music until the call is answered. The `goto` command creates the loop to repeat the announcement and the music. **Unconditionally** means under all conditions.

Tip:

Rather than loop your vectors directly back to the announcement step, go to the previous queue-to step. This way, if for some reason the call does not queue the first time, Communication Manager can attempt to queue the call again. If the call successfully queued the first time though, it merely skips the queue-to step and plays the announcement. The system cannot queue a call more than once in the exact same priority level.

To play and repeat an announcement, write this vector (steps 3-5):

Figure 104: Call Vector screen

```
change vector 1                                                    page 1 of x
                                CALL VECTOR
   Number:  1       Name: main number calls
               Attendant Vectoring? n     Meet-me Conf? y           Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Routing based on time of day

Write a vector for calls that come in after your office closes.

Assume that your business is open 7 days a week, from 8:00 a.m. to 5:00 p.m. When calls come in after business hours, you want to play your announcement 4002, which states that the office is closed and asks callers to call back during normal hours. Write the vector so the call disconnects after the announcement is played.

For after hours treatment, write this vector (steps 1, 6, and 7):

Figure 105: Call Vector screen

```
change vector 1                                                    page 1 of x
                                CALL VECTOR
   Number:  1       Name: main number calls
               Attendant Vectoring? n     Meet-me Conf? y          Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 goto step 7 if time-of-day is all 17:00 to all 8:00
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 stop
  07 disconnect after announcement 4002 ("We're sorry, our office is closed...")
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

If the **goto** command in step 5 fails, Communication Manager goes to the next step. The **stop** in step 6 prevents callers from incorrectly hearing the "office is closed" announcement in step 7. **Stop** keeps the call in the state it was in before the command failed. In this case, if step 5 fails, the call remains in step 4 and the caller continues to hear music.

CAUTION:

Add a stop vector step only after calls are routed to a queue. If a stop vector is executed for a call not in queue, the call drops.

Allowing callers to leave a message

Write a vector that allows callers to leave messages. This type of vector uses a hunt group called a messaging split. For our example, we send after-hours calls to the voice mailbox at extension 2000 and use messaging split 99.

Once the vector routes a call to the mailbox, the caller hears a greeting (that was recorded with the voice mail for mailbox 2000) that tells them they can leave a message.

To let callers leave messages, write this vector (step 7):

Figure 106: Call Vector screen

```
change vector 1                                                  page 1 of x
                                CALL VECTOR
   Number:  1       Name: main number calls
               Attendant Vectoring? n     Meet-me Conf? y           Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 goto step 7 if time-of-day is all 17:00 to all 8:00
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 stop
  07 messaging split 99 for extension 2000
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Redirecting calls during an emergency or holiday

You can provide a quick way for a supervisor or agent to redirect calls during an emergency or holiday. Use a special mailbox where you can easily change announcements. This vector is also an alternative to making sure all agents log out before leaving their telephones.

In our example, no agents are normally logged in to split 10. We'll use split 10 for an emergency. We preset buttons on our agents' telephones so people with these telephones can log in at the touch of a button.

To quickly redirect calls:

1. Create a special mailbox with the appropriate announcement such as "We are unable to answer your call at this time" or "Today is a holiday, please call back tomorrow."

   In our example, we recorded the mailbox greeting for extension 2001.

2. Insert the following bold steps (steps 1, 10, and 11).

   See Inserting a step on page 290 for more information.

Figure 107: Call Vector screen

```
change vector 1                                                      page 1 of x
                                CALL VECTOR
   Number:  1       Name: main number calls
               Attendant Vectoring? n     Meet-me Conf? y           Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 goto step 10 if staff agents split 10 > 0
  02 goto step 8 if time-of-day is all 17:00 to all 8:00
  03 queue-to main split 47 pri 1
  04 announcement 4001 (All agents are busy, please wait...)
  05 wait-time 60 secs hearing music
  06 goto step 2 if unconditionally
  07 stop
  08 messaging split 99 for extension 2000
  09 stop
  10 messaging split 99 for extension 2001
  11 stop
```

When there is an emergency, fire drill, or holiday, the supervisor or agent logs into this split. When an agent logs into split 10, the system looks at vector step 1, sees that more than 0 people are logged into split 10, and sends calls to step 10 (which sends to messaging split 99). When your business returns to normal and the agent logs out of split 10, call handling returns to normal.

Giving callers additional choices

You can give your callers a list of options when they call. Your vector tells Communication Manager to play an announcement that contains the choices. Communication Manager collects the digits the caller dials in response to the announcement and routes the call accordingly.

We'll create a vector that plays an announcement, then lets callers dial an extension or wait in the queue for an attendant.

Please note, the following example of this "auto attendant" vector is a new vector and is not built on the vector we used in the previous example.

To let callers connect to an extension, write this kind of vector:

Figure 108: Call Vector screen

```
change vector 1                                                    page 1 of x
                                CALL VECTOR
   Number:  1       Name: main number calls
               Attendant Vectoring? n     Meet-me Conf? y            Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

  01 wait-time 0 seconds hearing music
  02 collect 4 digits after announcement 4004 (You have reached our company.
        Please dial a 4-digit extension or wait for the attendant.)
  03 route-to digits with coverage y
  04 route-to number 0 with cov n if unconditionally
  05 stop
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Inserting a step

It is easy to change a vector step and not have to retype the entire vector. We will add announcement 4005 between step 3 and step 4 in vector 20.

To insert a new vector step in vector 20:

1. Type **`change vector 20`**. Press **Enter**.

   The system displays the Call Vector screen.

2. Press **Edit**.
3. Type **i** followed by a space and the number of the step you want to add.

   In our example, type i 4.

4. Type the new vector step.

   We will type **announcement 4005 (Please wait...)**.

5. Press **Enter** to save your changes.

   Tip:

   When you insert a new vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. Communication Manager inserts a "*" when the numbering needs more attention.

Deleting a step

To delete vector step 5 from vector 20:

1. Type `change vector 20`. Press **Enter**.

   The system displays the Call Vector screen.

2. Press **Edit**.

3. Type **d** followed by a space and the number of the step you want to delete.

   In our example, type **d 5**.

   Tip:

   You can delete a range of vector steps. For example, to delete steps 2 through 5, type **d 2-5**. Press **Enter**.

4. Press **Enter** to save your changes.

   Tip:

   When you delete a vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. An asterisk (*) is inserted when the numbering needs more attention.

Using Variables in Vectors

Variables in Vectors (VIV) is a Call Vectoring feature that allows you to create variables that can be used in vector commands to:

- Improve the general efficiency of vector administration
- Provide increased manager and application control over call treatments
- Allow you to create more flexible vectors that better serve the needs of your customer and contact center operations

The vector variables are defined in a central variable administration table. Values assigned to some types of variables can also be quickly changed by means of special vectors, Vector Directory Numbers (VDNs), or Feature Access Codes (FACs) that you administer specifically for that purpose. Different types of variables are available to meet different types of call processing needs. Vector variables can be added to "consider location," "messaging," and "adjunct routing" vector steps when the Call Center Release is 3.0 or later. Depending on the variable type, variables can use either call-specific data or fixed values that are identical for all calls. In either case, an administered variable can be reused in many vectors. For a more detailed description of variable types and purposes, see *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Administering vector variables

Administering variables and implementing them in your vectors is a relatively simple process:

1. First, determine how you intend to use the new variable and identify its defining characteristics. Use this information to decide on an available variable type that meets your needs.
2. Type **change variables**. The Variables for Vectors screen appears.

Figure 109: Variables for Vectors screen

```
 change variables                                                       Page  1 of x
                                 VARIABLES FOR VECTORS

Var   Description      Type  Scope Length Start   Assignment        VAC
A:    _______________  _____ __    ____   ___
B:    _______________  _____ __    ____   ___
C:    _______________  _____ __    ____   ___
D:    _______________  _____ __    ____   ___
E:    _______________  _____ __    ____   ___
F:    _______________  _____ __    ____   ___
G:    _______________  _____ __    ____   ___
H:    _______________  _____ __    ____   ___
I:    _______________  _____ __    ____   ___
J:    _______________  _____ __    ____   ___
K:    _______________  _____ __    ____   ___
L:    _______________  _____ __    ____   ___
M:    _______________  _____ __    ____   ___
```

3. In the **Var** column, select an unused letter between **A** and **Z**. This letter is used to represent this variable in vector steps. Complete the editable fields in the row that you select. Depending on your entry in the **Type** field, some fields in the row may be pre-populated and display-only, or not applicable.
   - **Description** - a short description of your variable
   - **Type** - the variable type
   - **Scope** - local or global
   - **Length** - length of the digit string
   - **Start** - digit start position
   - **Assignment** - pre-assigned value
   - **VAC** - Variable Access Code (for **value** variable type only)
4. Press **Enter** to save your changes.

Note:

For more detailed descriptions of fields, see the **Variables for Vectors** screen in the Screen Reference chapter. For a more detailed description and examples of vectors and vector variables, see *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Handling TTY calls with vectors

Unlike fax machines and computer modems, a Tele-typewriter device (TTY) has no handshake tone and no carrier tone. A TTY is silent when not transmitting. This is why systems cannot identify TTY callers automatically. However, the absence of these special tones also means that voice and TTY tones can be intermixed in pre-recorded announcements. The ability to provide a hybrid voice-and-TTY announcement, when combined with the auto-attendant vectoring capability, can permit a single telephone number to accommodate both voice and TTY callers.

The sample vector that follows allows TTY callers to access a TTY agent. It begins with a step that plays a TTY announcement combined with a voice announcement. The announcement tells the TTY caller to enter a digit that will direct them to a TTY support person. The vector then processes the digit entered to connect the TTY caller to the TTY split (or hunt group). For more information on recording TTY announcements, see Managing Announcements on page 511.

In the following example, split 47 (hunt group 47) has already been established and consists of TTY-enabled agents.

If a TTY caller calls the number that connects to vector 33, the following occurs:

1. After a short burst of ringing, a quick burst of TTY tones is sent to the caller telling the caller to hold, "HD". Then, a voice announcement follows for callers using a normal telephone connection. The announcement tells them to stay on the line. Finally, another burst of TTY tones is sent to the TTY caller which displays on the caller’s TTY device as, "Dial 1."

   The TTY caller won’t hear the voice announcement, but because the step collects digits, it allows the caller to enter **1** on his or her touchtone telephone.

   Note:

   For voice callers, the burst of TTY tones lasts about one second and sounds like a bird chirping.

2. In vector step 3, since the TTY caller entered **1** in vector step 2, the TTY caller is sent to vector step 8, at which point the caller is put in queue for a TTY-enabled agent in split 47.

   Note:

   The voice caller is sent to vector step 3 also, but a voice caller does not go to vector step 8 because the caller did not enter **1** at vector step 2. Instead, voice callers continue on to vector step 4, where they connect to split 48.

3. While the TTY caller waits in queue, he or she hears silence from vector step 9, then the announcement in vector step 10, and is then looped back to wait with silence by vector step 11.

More information

See the *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for more information.

Automated Attendant competes with several features for ports on the Call Classifier — Detector circuit pack or equivalent. See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207 for more information on the circuit pack.

Fixing vector problems

If there is a problem with a vector, Communication Manager records the error as a vector event. Vector events occur for a number of reasons including problems with a trunk, full queue slots, or the vector reaching the maximum 1000 steps allowed.

Use **display events** to access the **Event Report** screen and see the event record. Use the event record to see why the vector failed.

To view the **Event Report**:

1. Type **`display events`**. Press **Enter**.

   The system displays the **Event Report** screen.

Figure 110: Event Report screen

```
display events                                                          page 1 of x
                                EVENT REPORT
         The following option control which events will be displayed:
            EVENT CATEGORY
                        Category: Vector
            REPORT PERIOD
                        Interval: _a_ From: __/__/__:__ To: __/__/__:__
            SEARCH OPTIONS
                        Vector Number: __
                        Event Type: ___
```

2. To see all current vector events, press **Enter**.

   OR

   Indicate the events that you want to see by completing the **Report Period** and **Search Option** fields. See *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for more information.

3. Press **Enter** to view the report.

   The system displays the **Event Report (detail)** screen.

Figure 111: Event Report screen

```
display events                                                      page 1 of x
                                 EVENT REPORT
Event Event                    Event    Event    First        Last         Event
Type  Description               Data 1   Data 2   Occur        Occur        Cnt
  20  Call not queued           12/5     B        09/28/13:43  09/28/13:43  21
 541  Not a messaging split     Split    4C       09/28/13:43  09/28/13:43  136
```

Look at the information in the **Event Data** field to diagnose the vector event. In this example, there was a problem with:

- Vector 12, step 5
- Split 89

Working with Vector Directory Numbers

A VDN is an extension that directs an incoming call to a specific vector. This number is a "soft" extension number not assigned to an equipment location. VDNs must follow your dial plan.

We will create VDN 5011 for our sales department. A call into 5011 routes to vector 11. This vector plays an announcement and queues calls to the sales department.

SECURITY ALERT:
Vector fraud is one of the most common types of toll fraud because vectors route calls based on the class of restriction (COR) assigned to the VDN. See the *Avaya Toll Fraud and Security Handbook*, 555-025-600 for more information.

Adding a vector directory number

To add a vector directory number:

1. Type **`add VDN 5011`**. Press **Enter**.

   You enter the VDN extension you want to add. The system displays the **Vector Directory Number** screen.

Figure 112: Vector Directory Number screen

```
change vdn nnnn                                                Page   1 of x
                       VECTOR DIRECTORY NUMBER

                             Extension: nnnn
                                  Name*:
                          Vector Number: xxxx
                    Attendant Vectoring: n
                   Meet-me Conferencing? n
                     Allow VDN Override? n
                                    COR: 59
                                    TN*: 1
                               Measured: none
         Acceptable Service Level (sec):
                Service Objective (sec):
         VDN of Origin Annc. Extension*: 301
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:

* Follows VDN Override Rules
```

2. Type a description for this VDN in the **Name** field.

   In our example, type **Sales Department**.

   The information in the VDN Name field appears on a display telephone. This allows the agent to recognize the nature of the call and respond accordingly.

   Tip:

   The **VDN Override** on the Vector Directory Number screen controls the operation of the display.

3. Enter the vector number.

   In our example, type **11**.

4. In the **Measured** field, indicate how you want to measure calls to his VDN.

   In our example, type **both** (for both CMS and BCMS).

   Tip:

   BCMS must be enabled to use **both**. Use **`display system-parameters customer-options`** to see if BCMS is enabled.

5. Press **Enter** to save your changes.

Viewing vector directory numbers

To see the VDNs already associated with your vectors:

1. Type **`list VDN`**. Press **Enter**.

   The system displays the Vector Directory Number screen.

Figure 113: Vector Directory Number screen

```
list vdn
                         VECTOR DIRECTORY NUMBER
                                                                      Event
                                    VDN            Vec          Orig   Notif  Skills
Name                             Ext    Ovrd   COR TN Num   Meas   Annc   Adj 1st 2nd 3rd
Tech Support                     5000   y      59  1  234   none   301
Customer Serv.                   5001   n      1   1    1   none   302
New Orders                       5002   y      23  1    5   none   303
```

2. Each VDN maps to one vector. Several VDNs can map to the same vector.

Understanding Automatic Call Distribution

Automatic Call Distribution (ACD) is an Avaya Communication Manager feature used in many contact centers. ACD gives you greater flexibility to control call flow and to measure the performance of agents.

ACD systems operate differently from non-ACD systems, and they can be much more complex. ACD systems can also be more powerful because they allow you to use features and products that are not available in non-ACD systems. See the *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779, for more information on ACD call centers.

Enhancing an ACD system

First, all call center management systems (such as Avaya's Basic Call Management System (BCMS), BCMSVu, and the sophisticated Avaya IP Agent Call Management System) require ACD. These management systems give you the ability to measure more aspects of your center's operation, and in more detail, than is possible with standard Avaya Communication Manager reports.

Call vectoring greatly enhances the flexibility of a call center, and most vectoring functions require ACD. Vectoring is a simple programming language that allows you to custom design every aspect of call processing. See What are vectors? on page 282 for more information.

Together, ACD and vectoring allow you to use Expert Agent Selection (EAS) For a variety of reasons, you might want certain agents to handle specific types of calls. For example, you might want only your most experienced agents to handle your most important customers. You might have multilingual agents who can serve callers in a variety of languages.

EAS allows you to classify agents according to their specific skills and then to rank them by ability or experience within each skill. Avaya Communication Manager uses these classifications to match each call with the best available agent. See *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for more information on call vectoring and EAS.

Assigning a Terminating Extension Group

A Terminating Extension Group (TEG) allows an incoming call to ring as many as 4 telephones at one time. Any user in the group can answer the call.

Once a member of the TEG has answered a group call, the TEG is considered busy. If a second call is directed to the group, it follows a coverage path if one has been assigned.

Instructions

Now assign a terminating extension group to the advertising department. For example, we will assign this TEG to extension 6725.

1. Type **`add term-ext-group next`**. Press **Enter**.

   The system displays the Terminating Extension Group screen.

Figure 114: Terminating Extension Group screen

```
change term-ext-group n                                        Page   1 of   1
                       TERMINATING EXTENSION GROUP

    Group Number: n                          Group Extension:
      Group Name:                              Coverage Path:
   Security Code:                                        COR: 1
                                                          TN: 1
ISDN Caller Disp:                                        LWC Reception: none
      AUDIX Name:

GROUP MEMBER ASSIGNMENTS

    Ext      Name (first 26 characters)    Ext       Name (first 26 characters)
 1: 51001    26 character name sta 51001   3:
 2:                                        4: 51002   26 character name sta 51002
```

2. In the **Group Extension** field, type **6725**.

   This is the extension for the advertising group.
3. In the **Group Name** field, type **advertising**.

   This is the name of the group.
4. In the **Coverage Path** field, type **5**.

   This is the number of the call coverage path for this group.

Chapter 8: Routing Outgoing Calls

World Class Routing

Your system uses Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) to direct outgoing calls.

- AAR routes calls within your company over your own private network.
- ARS routes calls that go outside your company over public networks. ARS also routes calls to remote company locations if you do not have a private network.

Automatic routing begins when a user dials a feature access code (FAC) followed by the number the user wants to call. Avaya Communication Manager analyzes the digits dialed, selects the route for the call, deletes and inserts digits if necessary, and routes the call over the trunks you specify in your routing tables. ARS and AAR can access the same trunk groups and share the same route patterns and other routing information. ARS calls can be converted to AAR calls and vice-versa.

The FAC for AAR is usually the digit 8. The FAC for ARS is usually the digit 9 in the US and 0 outside of the US. Your Avaya technician or business partner sets up AAR on your server running Communication Manager and usually assigns the AAR FAC at the same time. You can administer your own ARS FAC.

This section describes only ARS call routing.

Managing Calling Privileges

Each time you set up a telephone, you use the **Station** screen to assign a class of restriction (COR). You can create different CORs for different groups of users. For example, you might want executives in your company to have different calling privileges than receptionists.

When you set up a COR, you specify a facility restriction level (FRL) on the **Class of Restriction** screen. The FRL determines the calling privileges of the user. Facility Restriction Levels are ranked from 0 to 7, where 7 has the highest level of privileges.

You also assign an FRL to each route pattern preference in the **Route Pattern** screen. When a user makes a call, the system checks the user's COR. The call is allowed if the caller's FRL is higher than or equal to the route pattern preference's FRL.

Instructions

Let us say we are setting up a new telephone for an executive. The current translations assign COR 1, with outward restrictions and an FRL 0, which is the lowest permission level available. We want to assign a COR with the highest level of permissions, FRL 7, to station 1234.

To change station 1234 from COR 1 to COR 7:

1. Type **`change station 1234`**. Press **Enter**.

   The **Station screen** appears.
2. In the **COR** field, type **7**. Press **Enter** to save your changes.
3. To change from FRL 0 to FRL 7, type **`change cor 7`**. Press **Enter**.

   The **Class of Restriction** screen appears.
4. In the **FRL** field, type **7**. Press **Enter** to save your changes.

Now all users with COR 7 will have the highest level of calling permissions.

Assigning ARS FAC

Be sure the ARS feature access code (FAC) is set up on your system. In the U.S., 9 is usually the ARS FAC. Users dial 9 to make an outgoing call.

When a user dials 9 to access ARS and make an outgoing call, the ARS access code 9 is dropped before digit analysis takes place. will not be part of the digit analysis.

Instructions

To assign the ARS FAC:

1. Type `change dialplan`. Press **Enter**.

   The **DCS to QSIG TSC Gateway** appears.
2. Move to the 9 row and type **fac** in the first column. Press **Enter** to save your changes.
3. Type `change features`. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.
4. Type **9** in the **ARS - access code** field.
5. Press **Enter** to save your changes.

Location ARS FAC

The **Location ARS FAC** allows users in different locations to use the same "culturally significant" FAC they are accustomed to, such as dialing 9 for an outside line, and access the same feature. The Location ARS FAC is only accessible for calling numbers at locations administered with that ARS FAC (for details on setting up Location ARS FAC, see the **Locations** screen). If an attempt is made to use an ARS FAC at a location for which it is not valid, the attempt is denied. The ARS access code on the **Feature Access Code (FAC)** screen continues to be used when a location ARS does not exist. If a location ARS FAC exists, then the ARS access code on the **Feature Access Code (FAC)** screen is prohibited/denied from that location.

By using a local ARS code, the ability to administer two ARS codes on the **Feature Access Code (FAC)** screen is lost.

Displaying ARS Analysis Information

Instructions

You will want to become familiar with how your system currently routes outgoing calls. To display the ARS Digit Analysis Table that controls how the system routes calls that begin with 1:

1. Type **`display ars analysis 1`**. Press **Enter**.

   The ARS Digit Analysis Table for dialed strings that begin with 1 appears. Note that Communication Manager displays only as many dialed strings as can fit on one screen at a time.

   Note:

   Type **`display ars analysis`** and press **Enter** to display an all-location screen. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

To see all the dialed strings that are defined for your system, run an ARS Digit Analysis report:

1. Type **`list ars analysis`**. Press **Enter**.

The **ARS Digit Analysis Report** appears. You might want to print this report to keep in your paper records.

Understanding ARS Analysis

With ARS, Communication Manager checks the digits in the number called against the ARS Digit Analysis Table to determine how to handle the dialed digits. Communication Manager also uses Class of Restriction (COR) and Facility Restriction Level (FRL) to determine the calling privileges.

Let us look at a very simple AAR and ARS digit analysis table. Your system likely has more defined dialed strings than this example.

Figure 115: ARS Digit Analysis Table

```
change ars analysis                                              Page 1 of X

                             ARS DIGIT ANALYSIS TABLE
                                Location:  all         Percent Full:
     Dialed         Total    Route    Call   Node  ANI
     String        Min Max  Pattern   Type   Num   Reqd
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
_________________  __  __    _____    ____   ___    n
```

The far-left column of the **ARS Digit Analysis Table** lists the first digits in the dialed string. When a user makes an outgoing call, the system analyzes the digits, looks for a match in the table, and uses the information in the matching row to determine how to route the call.

Let us say a caller places a call to 1-303-233-1000. Communication Manager matches the dialed digits with those in the first column of the table. In this example, the dialed string matches the ‘1’. Then Communication Manager matches the length of the entire dialed string (11 digits) to the minimum and maximum length columns. In our example, the 11-digit call that started with 1 follows route pattern 30 as an fnpa call.

Tip:

The first dialed digit for an external call is often an access code. If ‘9’ is defined as the ARS access code, Communication Manager drops this digit and analyzes the remaining digits with the ARS Analysis Table.

The Route Pattern points to the route that handles the calls that match this dial string.

Call Type tells what kind of call is made with this dial string. Call type helps Communication Manager decide how to handle the dialed string.

Examples of digit conversion

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits can be inserted or deleted from the dialed number. For instance, you can tell Communication Manager to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

The table below reflects these values:

- ARS feature access code = 9
  AAR feature access code = 8
  Private Network Office Code (also known as Home RNX) = 222
  Prefix 1 is required on all long-distance DDD calls
  Dashes (-) are for readability only

Communication Manager maps the dialed digits to the matching pattern that most closely matches the dialed number. Example: If the dialed string is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on pattern 957-123.

Table 5: ARS digit conversion examples

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	Call routes via AAR for RNX 362
Long-distance call to specified carrier	9-10222+DDD	10222	(blank)	(blank)	Call routes as dialed with DDD # over private network
Terminating a local DDD call to an internal station	9-1-201-957-5567 or 9-957-5567	1-201-957-5 or 957-5	222-5	222-5567.	Call goes to home RNX 222, ext. 5567
Unauthorized call to intercept treatment	9-1-212-976-1616	1-XXX-976	#	(blank)	"#" means end of dialing. ARS ignores digits dialed after 976. User gets intercept treatment.
					1 of 2

Table 5: ARS digit conversion examples (continued)

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
International calls to an attendant	9-011-91-6725 30	011-91	222-0111#	222-0111	Call routes to local server (RNX 222), then to attendant (222-0111).
International call to announcement (This method can also be used to block unauthorized IDDD calls)	9-011-91-6725 30	011-91	222-1234#	222.1234-	Call routes to local server (RNX 222), then to announcement extension (222-1234).
International call from certain European countries needing dial tone detection	0-00-XXXXXX XX	00	+00+	00+XXXX	The first 0 denotes ARS, the second pair of 0s denotes an international call, the pluses denote "wait" for dial tone detection.
					2 of 2

Defining operator assisted calls

Here is an example of how Communication Manager routes an ARS call that begins with 0 and requires operator assistance. The user dials 9 to access ARS, then a 0, then the rest of the number.

To see how Communication Manager handles a call to an operator:

1. Type **`display ars analysis 0`**. Press **Enter**.

   The **AAR and ARS Digit Analysis Table** screen starting with 0 appears.

Figure 116: ARS Digit Analysis Table screen

```
change ars analysis                                                 Page 1 of X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6

          Dialed          Total      Route    Call  Node  ANI
          String         Min Max    Pattern  Type  Num   Req
   0____________________1   1       1        svcl  ___   n
   0____________________8__ 8 ____  1        op    ___   n
   0____________________11_ 11____  1        op    ___   n
   00___________________2 _ 2 ____  1        op    ___   n
   01___________________10_ 23 ___  deny     op    ___   n
   011__________________10_ 23 ___  deny     iop   ___   n
   1____________________11_ 11 ___  3        intl  ___   n
```

The table in this example shows 6 translations for calls that begin with 0.

Instructions

We will use the ARS digit analysis table shown above and follow the routing for an operator assisted a call to NJ.

- A user dials 9 0 908 956 1234.
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 0, and analyzes the number. Then it:
  - determines that more than 1 digit was dialed
  - rules out the plan for 00, 01, and 011
  - determines that 11 digits were dialed
- Communication Manager routes the call to route pattern 1 as an operator assisted call.

Defining Inter-exchange carrier calls

Here is an example of how Communication Manager routes an ARS call to an inter-exchange (long-distance) carrier (IXC). IXC numbers directly access your long-distance carrier lines.

IXC numbers begin with 1010, followed by three digits, plus the number as it is normally dialed including 0, 00, or 1+ 10 digits. These numbers are set up on your default translations.

Remember, the user dials 9 to access ARS, then the rest of the number.

Instructions

To see how Communication Manager handles a call to an IXC:

1. Type **`display ars analysis 1`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen starting with 1 appears.

Figure 117: ARS Digital Analysis Table screen

```
change ars analysis                                                  Page 1 of X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
          Dialed         Total      Route   Call  Node  ANI
          String        Min Max    Pattern Type  Num   Req
   1____________________11_ 11 ___ 3       intl  ___   n
   1010xxx______________7 _ 7 ____ 5       op    ___   n
   1010xxx0_____________8 _ 8 ____ 5       op    ___   n
   1010xxx0_____________18_ 18 ___ 5       op    ___   n
   1010xxx00____________9 _ 9 ____ 5       op    ___   n
   1010xxx01____________17_ 25 ___ 3       iop   ___   n
   2____________________7 _ 7 ____ 2       hnpa  ___   n
   3____________________7 _ 7 ____ 2       hnpa  ___   n
   4____________________7 _ 7 ____ 2       hnpa  ___   n
```

This table shows five translations for IXC calls.

When you use x in the **Dialed String** field, Communication Manager recognizes x as a wildcard. The x represents any digit, 0 - 9. If I dial 1010, the next 3 digits will always match the x wild cards in the dialed string.

Use the ARS digit analysis table shown above and follow the routing for an IXC call to AT&T. 1010288 is the carrier access code for AT&T.

- A user dials 9 1010288 plus a public network number.
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 1010, and analyzes the number.
- Then it matches 288 with xxx and sends the call over route pattern 5.

Restricted area codes and prefixes

Certain area code numbers are set aside in the North American Numbering Plan. These numbers are 200, 300, 400, 500, 600, 700, 800, 877, 888, 900. You need to specifically deny calls made to area codes 200 through 900 (except 800 and 888).

You can also deny access to the 976 prefix, which is set aside in each area code for pay-per call services, if you do not want to incur charges. You can block 976 or any other prefix in all NPAs with a single entry in the digit analysis table. See Using wild cards on page 311 for more information.

Instructions

Set the 200 area code apart from other area codes 201 through 209. We use the digit analysis table 120 because it defines long distance calls that begin with 1 and all area codes from 200 through 209.

To deny long distance calls to the 200 area code:

1. Type **`change ars analysis 120`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen beginning with 120 appears.

Figure 118: ARS Digit Analysis Table screen

```
change ars analysis                                                Page 1 of X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
          Dialed          Total      Route    Call   Node  ANI
          String         Min Max    Pattern  Type   Num   Req
   120__________________11   11      4        fnpa   ___   n
   1200_________________11_  11____  deny     fnpa   ___   n
```

The table in this example shows two translations for calls that begin with 120.

First, follow the routing for a long-distance call that begins with 120 and is allowed. The 120 translation handles all dial strings 1-201 through 1-209, and there are many matches.

- A user dials 9 120 plus 8 digits (the first of the 8 digits is not 0).
- Communication Manager drops the ARS FAC (9 in our example), looks at the **ARS Digit Analysis Table** for 120, and analyzes the number. It determines the call is long-distance and sends the call over route pattern 4.

Now we will follow a call that begins with the restricted area code 200. Only one string matches this translation.

- A user dials 9 1200 plus 7 digits.
- Communication Manager drops the ARS FAC (9), and looks at the **ARS Digit Analysis Table** for 1200. It determines that the call type is deny, and the call does not go through.

Using wild cards

You can use wild cards to help separate out calls to certain numbers. Remember, when you use the wild card x in the **Dialed String** field, Communication Manager recognizes x as any digit, 0 - 9. For example, you can restrict users from making calls to a 555 information operator where you might incur charges.

Instructions

To prevent callers from placing calls to long-distance 555 information numbers:

1. Type **`change ars analysis 1`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen beginning with 1 appears.

Figure 119: ARS Digit Analysis Table screen

```
change ars analysis                                                Page 1 of X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
          Dialed          Total     Route   Call  Node  ANI
          String         Min Max    Pattern Type  Num   Req
   1_____________________11_ 11 ___ 1       intl  ___   n
   1xxx555______________11_ 11 ___ 1       intl  ___   n
   1010xxx______________7 _ 7 ____ 1       op    ___   n
```

2. Use the arrow keys to move to a blank **Dialed String** field.
3. Enter **1xxx555** in the **Dialed String** field.
4. Enter **11** in the **Total Min** and **11** in **Total Max** fields.
5. Enter **deny** (denied) in the **Route Pattern** field.
6. Enter **fnhp** in the **Call Type** field.
7. Press **Enter** to save your changes.

Defining local information calls

You can set up Communication Manager to allow calls to local information, or in this example, 411.

Instructions

To allow 411 service calls:

1. Type **`change ars analysis 4`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen beginning with 4 appears.

Figure 120: ARS Digit Analysis Table screen

```
change ars analysis                                                Page 1 of X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
          Dialed         Total      Route   Call  Node  ANI
          String        Min Max    Pattern Type  Num   Req
   411__________________3_  3  ___ 1       svcl  ___   n
   5____________________7 _ 7 ____ 2       hnpa  ___   n
   6____________________7 _ 7 ____ 2       hnpa  ___   n
   7____________________7 _ 7 ____ 2       hnpa  ___   n
   911__________________3   3 ____ 1       svcl  ___   n
```

2. Use the arrow keys to move to a blank **Dialed String** field.
3. Enter **411** in the **Dialed String** field.
4. Enter **3** in the **Total Min** and **3** in **Total Max** fields.
5. Enter **1** in the **Route Pattern** field.
6. Enter **svcl** (service call) in the **Call Type** field.

Press **Enter** to save your changes.

Administering Call Type Digit Analysis

There must be at least one entry in the **Call Type Digit Analysis Table** for Call Type Digit Analysis to take place.

1. Enter **`change calltype analysis`**.

   The **Call Type Digit Analysis Table** appears.

2. In the **Match** field, enter the digits the system uses to match to the dialed string.

   The dialed string contains the digits that Communication Manager analyzes to determine how to process the call. For example, enter **303** to match any dialed number beginning with 303.

3. In the **length: Min Max** fields, enter the minimum and maximum number of dialed digits for the system to match.

4. Enter up to four digit manipulations for this Match string.

   Enter the number of digits to delete, the number of digits to insert, and the call type against which to test the modified digit string.

Call Type Digit Analysis example

In our example, this is the administered **Call Type Digit Analysis Table**.

```
display calltype analysis                                               Page 1 of x
                         CALL TYPE DIGIT ANALYSIS TABLE
                                      Location:   all
    Dialed String             Delete Insert      Type    Delete Insert        Type
Match: 303________________ 1: 0      __________ ars  2: 0__    1___________ ars
length: Min 10 Max 10   __ 3: 3      __________ ext  4: 0      011_________ ars
```

In our example, Communication Manager analyzes 3035554927 for routing.

1. Communication Manager deletes 0 digits, inserts nothing, and searches the resulting 3035554927 against the ARS tables.
2. If there are no matching entries, Communication Manager deletes 0 digits, inserts the digit 1, and searches the resulting 13035554927 against the ARS tables.
3. If there are no matching entries, Communication Manager deletes 3 digits, inserts nothing, and searches the resulting 5554927 against numbers of ext type in the dial plan.
4. If there are no matching entries, Communication Manager deletes 0 digits, inserts 011, and searches the resulting 0113035554927 against the ARS tables.

Setting up Multiple Locations

You can define a location number for:

- Remote Offices
- Media gateways
- IP network regions, used by IP stations and IP trunks

You can create numbering plans and time zone and daylight savings plans that are specific for each location. Choose your main location, and offset the local time for each location relative to the system clock time. The main location is typically set to have offset 0.

Before you start

Ensure that the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information. If you are setting up locations across international borders, you must ensure that the **Multinational Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is also set to **y**.

Be sure your daylight savings rules are administered. Daylight Savings Rule numbers are located on the **Daylight Savings Rules** screen.

Each cabinet in a server or switch and each port network in the cabinet must be assigned a location number. See the `add-cabinet` and `change-cabinet` commands in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers,* 03-300431.

Instructions

For example, we will set up multiple locations for Communication Manager server with cabinets in Chicago and New York. Location 1 is assigned to the cabinet in Chicago, our main office, so Central Standard Time is used for our main location. Location 2 is assigned to the cabinet in New York. We'll define the numbering plan area (NPA) for the Chicago and New York locations, and set the time zone offset for NY to show the difference in time between Eastern Standard Time and Central Standard Time.

Tip:

Type `list cabinets` to see the **Cabinet** screen and a list of cabinets and their locations.

To define locations for cabinets in Chicago and New York:

1. Type **`change locations`**. Press **Enter**.

   The Locations screen appears.

Figure 121: Locations screen

```
change locations                                                Page   1 of   x
                                   LOCATIONS

               ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name           Timezone  Rule      NPA    ARS Attd Loc.   Pre- Proxy Sel.
 No                   Offset                    FAC  FAC Parms. fix  Rte. Pat.
 1.  Main           + 00:00     1        312
 2.  Denver-01______ - 01:00     1        303    ____ ____ __    ____ _______
 3.  Lincroft-01____ + 01:00     1        953    ____ ____ __    ____ _______
 xxx _______________ _ __:__   __        ___    ____ ____ __    ____ _______
 xxx _______________ _ __:__   __        ___    ____ ____ __    ____ _______
```

2. Type **y** in the **ARS Prefix 1 required for 10-digit NANP calls** field.

   Our dial plan requires users to dial a 1 before all 10-digit (long distance) NANP calls.

3. Type **Chicago** in the **Name** field in the Number **1** row.

   Use this field to identify the location.

4. Type **+00:00** in the **TimeZone Offset** field in the Number **1** row.

   In our example, the system time and the Chicago location time are the same.

5. Type **1** in the **Daylight Savings Rule** field in the Number **1** row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time.

   Tip:

   Use the **`display daylight-savings-rules`** command to see what rules have been administered on Communication Manager.

6. Type **312** in the **Number Plan Area Code** field in the Number **1** row.

   In our example, 312 is the local area code for Chicago, location 1.

7. Type **New York** in the **Name** field in the Number **2** row.

8. Type **-01:00** in the **TimeZone Offset** field in the Number **2** row.

   In our example, subtract one hour from the system clock in Chicago to provide the correct time for the location in New York.

9. Type **1** in the **Daylight Savings Rule** field in the Number **2** row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time, and both locations use the same rule.

10. Type **212** in the **NANP** field in the Number **2** row.

    In our example, 212 is the local area code for New York, location 2.

11. Press **Enter** to save your changes.

Related topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on the Multiple Locations feature.

See Setting the System Date and Time on page 41 for more information about how to set your system clock and specify the daylight savings rule for the location.

See Establishing Daylight Savings Rules on page 32 for more information about how to specify the dates and times of daylight savings rules.

Routing with Multiple Locations

When you set up multiple locations, you can define call routing that covers all locations as well as call routing specific to each individual location. Use your routing tables to define local routing for 911, service operators, local operator access, and all local calls for each location. Leave long-distance and international numbers that apply across all locations on the routing tables with **Location** field set to **all**.

Before you start

Be sure the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information.

AAR or ARS must be administered.

- For AAR, verify that either the **Private Networking** field or the **Uniform Dialing Plan** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.
- For ARS, verify that the **ARS** field is **y** on the **System-Parameters Customer-Options (Optional Features)** screen.

You can define a location number for:

- Remote Offices
- Media gateways
- IP network regions, used by IP stations and IP trunks

Instructions

For example, we will use ARS to set up local call routing for two Communication Manager server locations. Our Chicago server is assigned to location 1, and our New York server is assigned to location 2.

Our example shows a simple local dialing plan. Each location already contains location-specific routing tables. We'll use route pattern 1 for local service calls and route pattern 2 for local HNPA calls in the Chicago location.

Tip:

Create location-specific routing by assigning different route patterns for each location.

To define local calls for servers in Chicago and New York:

1. Type **`change ars analysis location 1`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen for location 1 appears.

Figure 122: ARS Digital Analysis Table screen

```
change ars analysis                                                 Page 1 of X

                                     ARS DIGIT ANALYSIS TABLE
                                             Location: 1___      Percent Full:  ___

     Dialed          Total     Route     Call    Node   ANI
     String         Min Max   Pattern    Type    Num    Reqd
2________________    7   7     2____     hnpa    ___     n
3________________    7   7     2____     hnpa    ___     n
4________________    7   7     2____     hnpa    ___     n
411______________    3   3     1____     svcl    ___     n
5________________    7   7     2____     hnpa    ___     n
555______________    7   7     deny_     hnpa    ___     n
6________________    7   7     2____     hnpa    ___     n
611______________    3   3     1____     svcl    ___     n
7________________    7   7     2____     hnpa    ___     n
8________________    7   7     2____     hnpa    ___     n
811______________    3   3     1____     svcl    ___     n
9________________    7   7     2____     hnpa    ___     n
```

2. Type the information for local dialed strings and service calls in each row on the screen.

   In our example, for location 1 (Chicago) local HNPA calls:

   - Type the appropriate digit in the **Dialed String** field.
   - Type **7** in the **Total Min** field.
   - Type **7** in the **Total Max** field.
   - Type **2** in the **Route Pattern** field.
   - Type **hnpa** in the **Call Type** field.

   In our example, for location 1 (Chicago) local service calls:

   - Type the appropriate digits in the **Dialed String** field.
   - Type **3** in the **Total Min** field.
   - Type **3** in the **Total Max** field.
   - Type **1** in the **Route Pattern** field.
   - Type **svcl** in the **Call Type** field.

3. Press **Enter** to save your changes.
4. Type **`change ars analysis 4 location 2`**. Press **Enter**.

   The **ARS Digit Analysis Table** for location 2 appears.

5. Type in the local HNPA and service call routing information for New York.
6. Press **Enter** to save your changes.

Related topics

See "Automatic Routing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on ARS.

See AAR and ARS Digit Analysis Table on page 723, AAR and ARS Digit Conversion Table on page 730, and Toll Analysis on page 1659 for general information on ARS administration. You can define location specific entries in addition to the global entries on these screens.

See "Multiple Locations" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information on the Multiple Locations feature.

Modifying Call Routing

If your system uses ARS Digit Analysis to analyze dialed strings and select the best route for a call, you must change the digit analysis table to modify call routing. For example, you'll need to update this table to add new area codes or to restrict users from calling specific areas or countries.

Adding a new area code or prefix

A common task for system administrators is to configure their system to recognize new area codes or prefixes.

When you want to add a new area code or prefix, you look up the settings for the old area code or prefix and enter the same information for the new one.

> **Tip:**
>
> Use **`display toll xxx`**, where ***`xxx`*** is the prefix you want to add, to see if the new area code or prefix number is set up as a toll call (y) or not. Some users might not be allowed to dial toll call numbers.

Instructions

We will add a new area code. When the California area code, 415, splits and portions change to 650, you will need to add this new area code to your system.

> **Tip:**
>
> If you do not need to use 1 for area code calls, omit the **1** in steps 1, 3, and 5 in our example. Also, enter **10** in the **Total Min** and **Total Max** fields (instead of 11) in step 6.

To add this non-local area code:

1. Type **`list ars route-chosen 14152223333`**. Press **Enter**.

   You can use any 7-digit number after 1 and the old area code (415). We used 222-3333.

   The **ARS Route Chosen Report** screen appears.

Figure 123: ARS Route Chosen Report screen

```
list ars route-chosen 14152223333

                           ARS ROUTE CHOSEN REPORT

     Location: 1                      Partitioned Group Number:  1

      Dialed               Total      Route     Call      Node      Location
      String             Min  Max     Pattern   Type      Number
      141                11   11      30        fnpa                all
```

2. Write down the **Total Min**, **Total Max**, **Route Pattern**, and **Call Type** values from this screen.

   In this example, the **Total Min** is **11**, **Total Max** is **11**, **Route Pattern** is **30**, and the **Call Type** is **fnpa**.
3. Type **`change ars analysis 1650`**. Press **Enter**.

   The **ARS Digit Analysis Table** screen appears.

Figure 124: ARS Digit Analysis Table screen

```
change ars analysis                                                 Page 1 of  X
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
          Dialed         Total      Route   Call  Node  ANI
          String        Min Max     Pattern Type  Num   Req
   1____________________11   11     30      fnpa  ___   n
   167__________________11   11     30      fnpa  ___   n
   1800_________________11   11     30      fnpa  ___   n
   2___________________ 7    7      2       hnpa  ___   n
   3___________________ 7    7      2       hnpa  ___   n
   4___________________ 7    7      2       hnpa  ___   n
   5___________________ 7    7      2       hnpa  ___   n
   7___________________ 7    7      2       hnpa  ___   n
   8___________________ 7    7      2       hnpa  ___   n
   911_________________ 3    3      1       emer  ___   n
   976________________ _11   11     deny    hnpa  ___   n
```

4. Move to a blank **Dialed String** field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate **Dialed String** field, where you can make changes.
5. Enter **1650** in the **Dialed String** field.

6. Enter the minimum and maximum values from step 2 in the **Total Mn** and **Total Mx** fields.

   In our example, enter **11** in each field.

7. Enter the route pattern from step 2 in the **Route Pattern** field.

   In our example, enter **30**.

8. Enter **fnpa** in the **Call Type** field.

9. Enter the node number from step 2 in the **Node Num** field.

   For our example, leave the node number blank.

10. Press **ENTER** to save your changes.

To add a new prefix, follow the same directions, except use a shorter dial string (such as `list ars route-chosen 2223333`, where `222` is the old prefix) and a dial type of **hnpa**.

Tip:

If you change an existing area code for a network with multiple locations, be sure to change the **Number Plan Area Code** field on the **Locations** screen. See Setting up Multiple Locations on page 314.

Using ARS to restrict outgoing calls

ARS allows you to block outgoing calls to specific dialed strings. For example, you can restrict users from making international calls to countries where you do not do business, or in the U.S. you can restrict access to 900 and 976 pay-per-call numbers.

SECURITY ALERT:

To prevent toll fraud, deny calls to countries where you do not do business. The following countries are currently concerns for fraudulent calling.

country	code	country	code
Colombia	57	Pakistan	92
Ivory Coast	225	Peru	51
Mali	23	Senegal	221
Nigeria	234	Yemen	967

Instructions

To prevent callers from placing calls to Colombia (57):

1. Type **`change ars analysis 01157`**. Press **Enter**.

   Enter 011 (international access) and the country code (57). The **ARS Digit Analysis Table** screen appears.

2. Move to a blank **Dialed String** field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate **Dialed String** field. Skip to Step 5 to deny calls to this dialed string.

3. Enter **01157** in the **Dialed String** field.
4. Enter **10** in the **Total Min** and **23** in **Total Max** fields.
5. Enter **deny** (denied) in the **Route Pattern** field.
6. Enter **intl** in the **Call Type** field.
7. Press **Enter** to save your changes.

Overriding Call Restrictions

You can use authorization codes to enable callers to override a station's calling privileges. For example, you can give a supervisor an authorization code so they can make calls from a telephone that is usually restricted for these calls. Since each authorization code has its own COR, the system uses the COR assigned to the authorization code (and FRL assigned to the COR) to override the privileges associated with the employee's telephone.

Note that authorization codes do not override dialed strings that are denied. For example, if your ARS tables restrict users from placing calls to Colombia, a caller cannot override the restriction with an authorization code.

Before you start

Verify that the **Authorization Codes** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**.

SECURITY ALERT:
You should make authorization codes as long as possible to increase the level of security. You can set the length of authorization codes on the **Feature-Related System Parameters** screen.

Instructions

We will create an authorization code 4395721with a COR of 2.

1. Type **`change authorization-code 4395721`**. Press **Enter**.

   The **Authorization Code - COR Mapping screen** appears.
2. In the **AC** field, type **4395721**.
3. In the **COR** field, enter **2**.
4. Press **Enter** to save your changes.

Figure 125: Authorization Code - COR Mapping

```
change authorization-code                                          Page 1 of 1
                      Authorization Code - COR Mapping
     Note:    XX codes administered.  Use "list" to display all codes.
    AC     COR   AC     COR   AC     COR   AC     COR   AC      COR    AC     COR

   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
   _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
```

Defining ARS Partitions

Most companies want all their users to be able to make the same calls and follow the same route patterns. However, you might find it helpful to provide special calling permissions or restrictions to a group of users or to particular telephones.

ARS partitioning allows you to provide different call routing for a group of users or for specific telephones.

Note:

If you used partitioning on a prior release of Avaya Communication Manager and you want to continue to use partitioning, please read this section carefully. In this release of Avaya Communication Manager, partition groups are defined on the Partition Route Table. If you want to define routing based on partition groups, use the Partition Route Table. Partition groups are no longer defined on the Digit Analysis Table.

Before you start

- Ensure that the **Tenant Partitioning** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**.
- Ensure that the **Time of Day Routing** field on the **System Parameters Customer-Options (Optional Features)** screen is **n**.

Setting up partition groups

Let us say you allow your employees to make local, long distance, and emergency calls. However, you have a lobby telephone for visitors and you want to allow users to make only local, toll-free, and emergency calls from this telephone.

To restrict the lobby telephone, you modify the routing for a partition group to enable only specific calls, such as U.S. based toll-free 1-800 calls, and then assign this partition group to the lobby telephone.

Instructions

To enable 1-800 calls for partition group 2:

1. Type **`list ars route-chosen 18002221000`**. Press **Enter.**

   You can use any 7-digit number following the 1800 to create an example of the dialed string.

   The **ARS Route Chosen Report** screen for partition group 1 appears.

Figure 126: ARS Route Chosen Report screen

```
list ars route-chosen 18002221000

                            ARS ROUTE CHOSEN REPORT

      Location :  1                     Partitioned Group Number: 1

          Dialed                Total     Route      Call       Node     Location
          String               Min  Max  Pattern     Type       Number

1800___________________         11  11     p1___       fnpa       _____      all
```

2. Record the route pattern for the selected dialed string.

   In our example, the route pattern for 1800 is p1. This indicates that the system uses the Partition Routing Table to determine which route pattern to use for each partition.

   Note:

   If there was a number (with no p) under **Route Pattern** on the **Route Chosen Report**, then all partitions use the same route pattern. You need to use the Partition Routing Table only if you want to use different route patterns for different partition groups.

3. Press **Cancel** to return to the command prompt.

4. Type **`change partition-route-table index 1`**. Press **Enter.**

   The **Partition Routing Table** screen appears. In our example, partition group 1 can make 1800 calls and these calls use route pattern 30.

Figure 127: Partition Routing Table screen

```
change partition route-table                                    Page 1 of X
                             Partition Routing Table

                                Routing Patterns
 Route
 Index        PGN 1   PGN 2   PGN 3   PGN 4   PGN 5   PGN 6   PGN 7   PGN 8
 -----        -----   -----   -----   -----   -----   -----   -----   -----
 196          ____    ____    ____    ____    ____    ____    ____    ____
 197          ____    ____    ____    ____    ____    ____    ____    ____
 198          ____    ____    ____    ____    ____    ____    ____    ____
 199          ____    ____    ____    ____    ____    ____    ____    ____
 200          ____    ____    ____    ____    ____    ____    ____    ____
 201          ____    ____    ____    ____    ____    ____    ____    ____
 202          ____    ____    ____    ____    ____    ____    ____    ____
 203          ____    ____    ____    ____    ____    ____    ____    ____
 204          ____    ____    ____    ____    ____    ____    ____    ____
 205          ____    ____    ____    ____    ____    ____    ____    ____
 206          ____    ____    ____    ____    ____    ____    ____    ____
 207          ____    ____    ____    ____    ____    ____    ____    ____
 208          ____    ____    ____    ____    ____    ____    ____    ____
 209          ____    ____    ____    ____    ____    ____    ____    ____
 210          ____    ____    ____    ____    ____    ____    ____    ____
```

5. In the **PGN2** column that corresponds to Route Index 1, type **30**. Press **Enter**.

   This tells the system to use route pattern 30 for partition group 2 and allow partition group 2 members to make calls to 1800 numbers.

Assigning a telephone to a partition group

To assign an extension to a partition group, first assign the partition group to a COR, and then assign that COR to the extension.

Instructions

To assign a Class of Restriction (COR) to partition group 2:

1. Type **`list cor`**. Press **Enter**.

   The **Class of Restriction Information** screen appears.

Figure 128: Class of Restriction Information screen

```
list cor                                                            page 1 of x
                   CLASS OF RESTRICTION INFORMATION

                  COR   COR Description
                   0
                   0    supervisor
                   2    telecommuting
                   3
```

2. Choose a COR that has not been used. Press **Cancel**.

   In our example, select **3**.

3. Type **`change cor 3`**. Press **Enter**.

   The **Class of Restriction** screen appears.

Figure 129: Class of Restriction screen

```
change cor n                                                    Page 1 of x
                              CLASS OF RESTRICTION
               COR Number: n
          COR Description: supervisor

                      FRL: 0                                    APLT? y
  Can Be Service Observed? n              Calling Party Restriction: none
Can Be A Service Observer? y               Called Party Restriction: none
 Partitioned Group Number: 1           Forced Entry of Account Codes? n
         Priority Queuing? n                     Direct Agent Calling? y
     Restriction Override: none             Facility Access Trunk Test? n
     Restricted Call List? n                      Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y               Fully Restricted Service? n
Group II Category For MFC: 7               Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_               Add/Remove Agent Skills? y
            MF ANI Prefix: ______          Automatic Charge Display? n
Hear System Music on Hold? y     PASTE(Display PBX Data on telephone)? n
                              Can Be Picked Up By Directed Call Pickup? n
                                          Can Use Directed Call Pickup? n
                                          Group Controlled Restriction: inactive
```

4. Type a name for this COR in the **COR Description** field.

   In our example, type **lobby**.

5. Enter **2** in the **Partitioned Group Number** field.
6. Press **Enter** to save your changes.

Now assign COR 3 to the lobby telephone at extension 1234:

1. Type `change station 1234`. Press **Enter**.

   The Station screen for 1234 appears.
2. In the **COR** field, enter **3**.
3. Press **Enter** to save your changes.

Setting up Time of Day Routing

Time of Day Routing lets you redirect calls to coverage paths according to the time of day and day of the week. You need to define the coverage paths you want to use before you define the time of day coverage plan.

You can route calls based on the least expensive route according to the time of day and day of the week the call is made. You can also deny outgoing long-distance calls after business hours to help prevent toll fraud. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks used for call forwarding to external numbers.

Before you start

AAR or ARS must be administered on Communication Manager before you use Time of Day Routing.

- For AAR, verify that either the **Private Networking** field or the **Uniform Dialing Plan** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.
- For ARS, verify that the **ARS** field is **y** and the **Time of Day Routing** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Instructions

As an example, we will allow our executives to make long distance calls during business hours. Let us look at the Time of Day Routing Plan before we make any changes.

To display your Time of Day Routing Plan:

1. Type `display time-of-day 1`. Press **Enter**.

   The Time Of Day Routing Plan screen for plan **1** appears.

Figure 130: Time of Day Routing Plan 1 screen

```
change time-of-day
                          TIME OF DAY ROUTING PLAN ___               Page 1 of 1
     Act      PGN   Act     PGN   Act      PGN   Act      PGN   Act      PGN   Act      PGN
     Time     #     Time    #     Time     #     Time     #     Time     #     Time     #
Sun  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Mon  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Tue  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Wed  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Thu  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Fri  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Sat  00:00    1     __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
```

Make a note of the routing plan that is currently in effect. In our example, this plan is for employees who can only make local calls.

You can see that in our example, two partition group numbers control time of day routing. PGN 1 begins one minute after midnight (00:01) every day of the week, and is used for after-business hours and all day Saturday and Sunday. PGN 2 is assigned to office hours Monday through Friday, not including noon (12:00) to 1:00 p.m. (13:00).

2. Press **Cancel** to clear the screen.

Now we will create a new time of day routing plan for long-distance calls for our executives.

1. Type **`change time-of-day 2`**. Press **Enter**.

   The **Time of Day Routing Plan 2** screen appears.

2. Type **1** in each field as shown on **Time of Day Routing Plan 1**.

   In our example, this is the PGN used for after hours and the lunch hour.

3. Type **3** in all other fields.

   In our example, PGN 3 uses the route pattern for long-distance calls during business hours. We can save money by using the trunk lines provided by our new long-distance carrier.

4. Press **Enter** to save your changes.

Now assign your new Time of Day Routing Plan 2 to the COR assigned to your executives. See Class of Restriction on page 834 to see where to assign this field.

Example

For this example, assume the following:

- Jim is the user at extension 1234.
- Extension 1234 is assigned a COR of 2.
- COR 2 is assigned a Time of Day Plan Number of 1.
- The Time of Day Routing Plan 1 is administered as shown in the example above.

When Jim comes into work on Monday morning at 8:30 and makes an ARS call (dials the ARS access code followed by the number of the person he is calling), the system checks the Time of Day Plan Number assigned to Jim's COR.

Because Jim has a COR of 2 with Time of Day Plan Number 1, the system uses Time of Day Routing Plan 1 to route the call.

According to Time of Day Routing Plan 1, calls made between 8:00 a.m. and 11:59 a.m. route according to the route pattern set up on PGN 1.

If Jim makes a call between 12:00 p.m. and 1:00 p.m. on Monday, the Time of Day Routing Plan 1 is used again. However, this time the call is routed according to PGN 2.

Related topics

See Route Pattern on page 1444 screens for more information.

See Defining ARS Partitions on page 324 to see how to set up partition groups.

Location by Region

Location by Region provides a way to administer location by IP network region. This allows for the correct date and time information and trunk routing based on the IP network region.

Setting up a remote user by network region and time zone

Instructions

With your system located in New York and a remote user located in Germany, to create the correct time zone settings:

1. Type **`change locations`**. Press **Enter**.

   The Locations screen displays.

Figure 131: Locations screen

```
change locations                                                  Page   1 of   1
                                   LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name                Timezone  Rule       NPA     ARS Attd Loc.   Pre- Proxy Sel.
 No                        Offset                      FAC  FAC  Parms. fix  Rte. Pat.
 1.   Main              + 00:00    1          312
 2.   Denver-01______   - 01:00    1          303     ____ ____ __     ____ _______
 3.   Lincroft-01____   + 01:00    1          953     ____ ____ __     ____ _______
 xxx _______________ _ __:__    __          ___     ____ ____ __     ____ _______
 xxx _______________ _ __:__    __          ___     ____ ____ __     ____ _______
```

2. On line 11, in the **Name** field, enter the Communication Manager server associated with the location (for instance, **Germany**).
3. In the first **Timezone Offset** field, enter **+** to indicate the time is ahead of the system time.
4. In the second **Timezone Offset** field, enter **08** for the number of hours difference between this location and system time.
5. In the **Daylight Savings** field, enter **1** if this country has daylight savings.
6. Press **Enter** to save your changes.
7. Type **`change ip-network-map`**. Press **Enter**.

   The IP Address Mapping screen displays.

Figure 132: IP Address Mapping screen

```
change ip-network-map                                            Page 1 of X

                            IP ADDRESS MAPPING
                                                                   Emergency
                                   Subnet                 802.1Q   Location
FROM IP Address   (TO IP Address or Mask)    Region         VLAN   Extension
  1.__2.__3.__0    1.__2.__3.255    24        __1          ___3    ________
  1.__2.__4.__4    1.__2.__4.__4    32        __2          ___0    ________
  1.__2.__4.__5    1.__2.__4.__5    __        __3          ___0    ________
  1.__2.__4.__6    1.__2.__4.__9    __        __4          ___4    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
___.___.___.___  ___.___.___.___    __        ___          ____    ________
```

8. In the **From IP Address** field, enter the IP address for the remote station in Germany.
9. In the **To IP Address** field, enter the IP address of your system.
10. In the **Subnet or Mask** field, enter the subnet mask value of your network.
11. In the **Region** field, enter a number that is not being used. In this example, enter **3**.
12. Press **Enter** to save your changes.
13. Type **`change ip-network-region 3`**. Press **Enter.**

    The IP Network Region screen displays.

Figure 133: IP Network Region screen

```
change ip-network-region n                                     Page 1 of x
                              IP NETWORK REGION
  Region: n
Location:                  Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                            IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value:                      Server IP Address:   .   .   .
        Audio PHB Value:                            Server Port: 5005
        Video PHB Value:
802.1P/Q PARAMETERS                     RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? y
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20     Retry upon RSVP Failure Enabled? y
     Keep-Alive Interval (sec): 6              RSVP Profile: guaranteed-service
             Keep-Alive Count: 5      RSVP unreserved (BBE) PHB Value: 40
```

14. In the **Name** field, enter the location name for familiarity.
15. In the **Location** field, enter the number from the **Locations** screen. In this example, it was **11**.
16. Press **Next Page** until you get to page 3, the **Inter Network Region Connection Management screen**.

Figure 134: Inter Network Region Connection Management screen

```
change ip-network-region n                                         Page   3 of x

                   Inter Network Region Connection Management

 src dst codec direct   Total          Video                              Dyn
 rgn rgn  set  WAN   WAN-BW limits Norm Prio Shr Intervening-regions   CAC IGAR
 3   1    1       y    256:Kbits             y                               n
 3   2    1       n       :NoLimit           n                               n
 3   3    1               :NoLimit                                           n
 3   4    1       n       :NoLimit           n                               n
 3   5    1       n       :NoLimit           n                               n
 3   6    1       y       :NoLimit           y                               n
 3   7    1       y     10:Calls             y                               n
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

17. Notice in the **src rgn** column that a **3** displays, and under **dst rgn** a **1**, indicating that Network Region 3 (Germany) is connected to Network Region 1 (New York) using Codec Set 1.
18. Press **Enter** to save your changes.

Related Topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on the Multiple Locations feature.

Chapter 9: Managing Multimedia Calling

Multimedia Applications Server Interface

The Multimedia Applications Server Interface (MASI) defines a protocol and a set of operations that are used to extend Avaya Communication Manager feature functionality to a Multimedia Communications Exchange (MMCX) system. MASI architecture fits the client/server model, where Avaya Communication Manager functions as a server for MMCX clients. Examples of features supported by MASI include call detail recording (CDR), AUDIX/INTUITY voice mail integration, and Automatic Alternate Routing (AAR)/Automatic Route Selection (ARS).

MMCX can make use of both MASI features and MMCX autonomous features. Autonomous features are those that MMCX provides, even if MASI is not enabled. This document does not discuss them unless there is a consideration for MASI administration.

Some autonomous MMCX features:

- Basic Call (Place/Drop)
- Call Coverage
- Conference
- Transfer

Avaya Communication Manager/MASI features:

- Basic Call (Place/Drop) - Avaya Communication Manager tracks the status of all calls placed to or from a MASI terminal.
- Call Detail Recording - Avaya Communication Manager tracks calls to and from MASI terminals and can produce call records that indicate if a call uses MASI.
- Call Coverage - Avaya Communication Manager tracks MMCX calls that are sent to coverage. A Communication Manager coverage path can contain both MASI terminals and Communication Manager stations.
- Conference - Avaya Communication Manager tracks conference calls that involve MASI terminals, if a Communication Manager station originates the conference. Conferences that involve MASI terminals and Communication Manager stations are voice-only. If the Communication Manager station originates the call, the caller can use the consultative form of conference or transfer.
- World Class Routing (AAR or ARS) - Calls from MASI terminals can take advantage of Avaya Communication Manager World Class Routing capabilities.

- Voice messaging access to AUDIX/INTUITY - MMCX users can take advantage of AUDIX voice messaging, and receive message waiting indication.
- MMCX trunking - By assigning trunk access codes to interfaces from the MMCX to other MMCXs or the PSTN, Avaya Communication Manager can monitor traffic over those interfaces.

Before you start

CAUTION:

Avaya Communication Manager offers a wide range of features, and MMCX users might want to take advantage of this. In some cases, these features will operate as expected. However, some features are not supported for use over the MASI link, and their behavior is unpredictable. You might cause harm to your system by attempting to use these features. The Interactions section contains a list of features, and lists those features that are absolutely not supported for use with MASI. If you administer features on the DO NOT ADMINISTER list, Avaya cannot be responsible for the result.

For purposes of administration, there are feature buttons and groups of users that you must not administer with MASI terminal extensions. There are also features that you simply cannot administer for a MASI terminal, because the software does not allow it.

About this section

The following section describes the Multimedia Applications Server Interface (MASI), and provides instructions on how to set it up, including administration and monitoring. It also includes a section on troubleshooting.

You need to use both the DEFINITY Server system administration terminal (SAT) and the MMCX administration terminal to administer MASI. This document describes what you need to do at the DEFINITY Server SAT. It also occasionally mentions administration that you must do at the MMCX administration terminal. For more detailed MMCX information, see your MMCX documentation.

List of terms

This is a list of terms that are specific to MASI, or that have meanings in the context of MASI that are not standard.

- **chgmasi** - The command you use to administer MASI at the MMCX administration terminal.
- **Interserver** - Connections between MMCX terminals on different MMCX servers/nodes.
- **MASI domain** - A MASI domain consists of Avaya Communication Manager and one or more MASI nodes that share the same dial plan. That is, the extension numbers on the MMCX are known to Communication Manager, and fit in the Communication Manager dial plan.
- **MASI interworking** - MASI interworking refers to the completion of a voice connection within Avaya Communication Manager, involving at least one MASI terminal and a MASI path.
- **MASI link** - The connection between the MMCX and Avaya Communication Manager.
- **MASI node** - A single MMCX server. You can connect more than one MASI node to a Communication Manager. Each node has a separate number. This node number needs to be consistent whenever referring to a specific MMCX server.
- **MASI non-interworking** - MASI non-interworking refers to the completion of a call by MMCX, not involving a MASI path.
- **MASI path** - The Integrated Services Digital Network (ISDN) B-channels between MMCX and Avaya Communication Manager in a MASI environment. Paths are used for voice and data connections between Avaya Communication Manager and MMCX.
- **MASI signaling link** - ISDN D-channel used to transport a new ISO protocol called the MASI protocol between Avaya Communication Manager and the MMCX.
- **MASI terminal** - The representation in Avaya Communication Manager of MMCX terminals in a MASI environment.
- **MMCX interface** - PRI interface for connecting an MMCX server to other public, private or wide area network (WAN) switching systems or equipment that is part of the public network. Similar to a Communication Manager trunk group. These can include non-MASI trunks connecting Avaya Communication Manager and the MMCX.
- **MMCX trunk** - The representation in Avaya Communication Manager of trunk or network facilities terminating on MMCX. For purposes of MASI, they are called "interfaces."

Planning for MASI

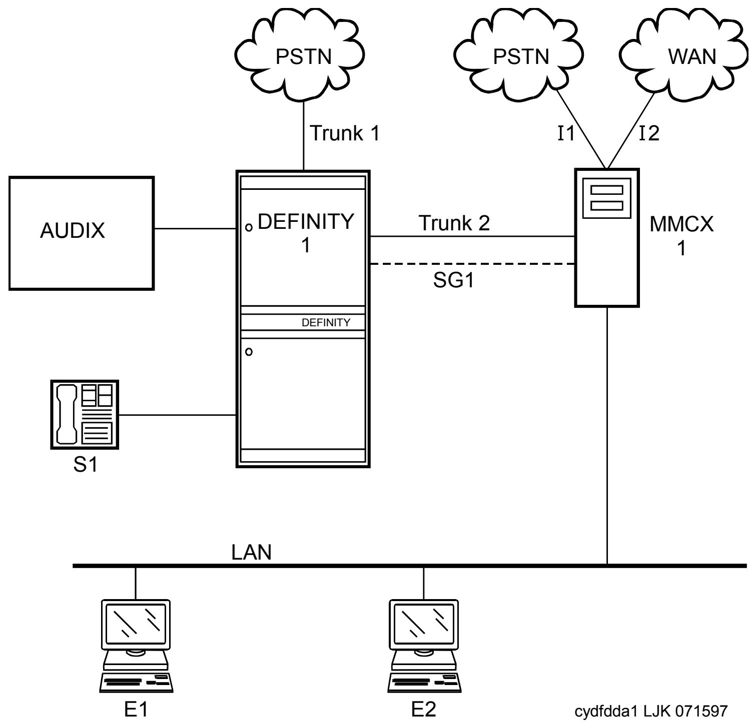
Before you start to administer MASI, you should make a plan for how to do it. Among the configurations on the following pages, there is probably one that matches the configuration of your system fairly closely. You might want to either write on these pages, or draw up your own configuration. It might help you if you have already determined trunk group and signaling group numbers, unused extensions, and so on. The following are things you need to consider:

- Establish the dial plan on the MMCX to agree with that of Avaya Communication Manager. If you use Universal Dial Plan and MMCX, you might need to make adjustments for the MMCX dial plan.
- Find unused extensions and trunk group numbers. You need:
  - one trunk group number for each ISDN-PRI connection to the MMCX.
  - one signaling group number for each MASI node and an unused Communication Manager extension for the signaling group.
  - one unused Communication Manager extension for the Near-End Path Termination number for all MASI Paths to this ECS. You can use the same number for all MASI nodes in the domain.
  - two unused MMCX extensions for the ***nearpath*** and ***tscnum*** arguments to the **chgmasi** command. This is the command you use to administer MASI on the MMCX.

MASI configurations

There are several ways to set up combinations of MASI nodes and DEFINITY servers.The following figures depict several possible configurations.

Figure 135: MASI domain of Avaya Communication Manager running on one DEFINITY Server and one MMCX



The parts of this drawing, for MASI, are as follows:

- Trunk 1 — This is any type of trunk connection to the public network.
- Trunk 2 — This is the link between the Avaya Communication Manager solution and the MMCX, and requires a TN464C or later DS1 circuit pack. You administer this link as an ISDN-PRI trunk group, a MASI path and an NCA-TSC.
- I1 and I2 — These are MMCX interfaces to destinations other than Avaya Communication Manager. Administer as MASI trunks.
- E1 and E2 — Endpoints (terminals) belonging to the MMCX. Administer as MASI terminals.
- MMCX — Determine a node number for each MMCX server. This can be any number from 1 to 15. Once the node number is established, Avaya Communication Manager informs the MMCX of its node number.
- S1 — Avaya Communication Manager station.

Figure 136: MASI domain of Avaya Communication Manager running on one DEFINITY Server and two (or more) MMCXs

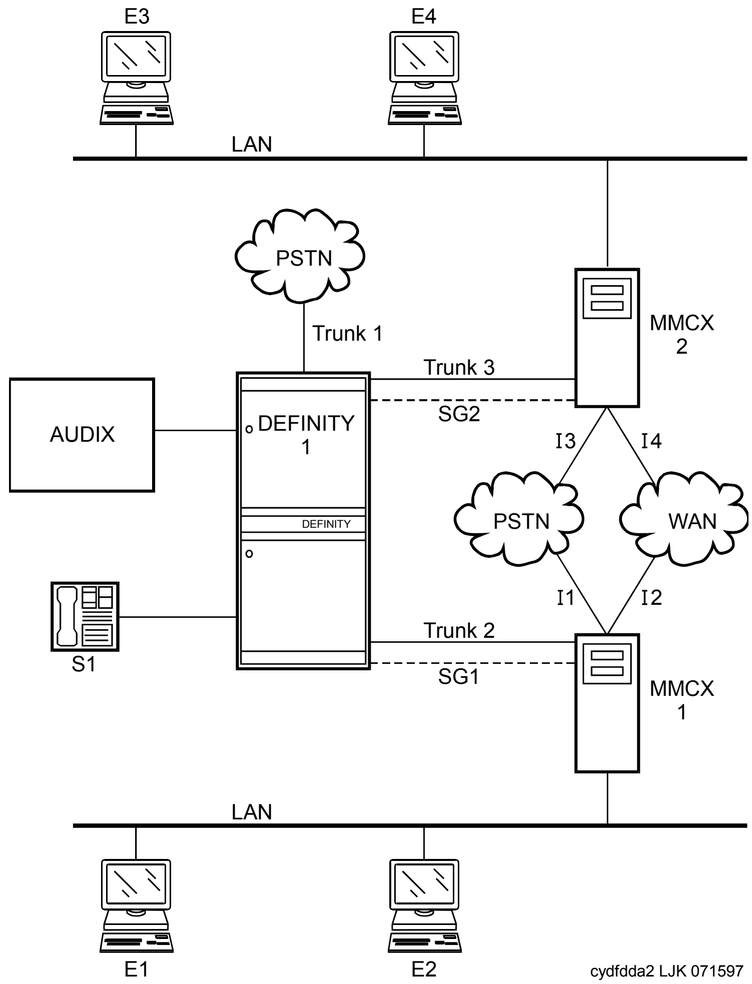


Figure 137: Two separate MASI domains

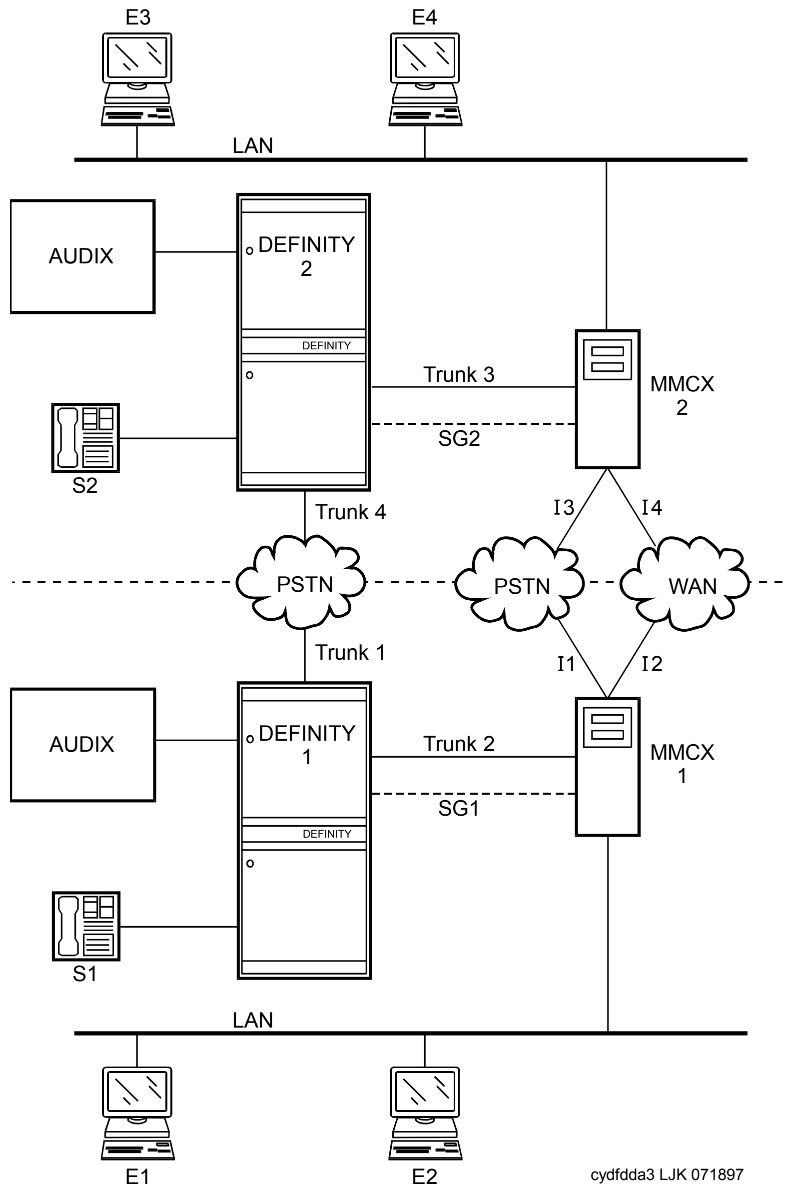
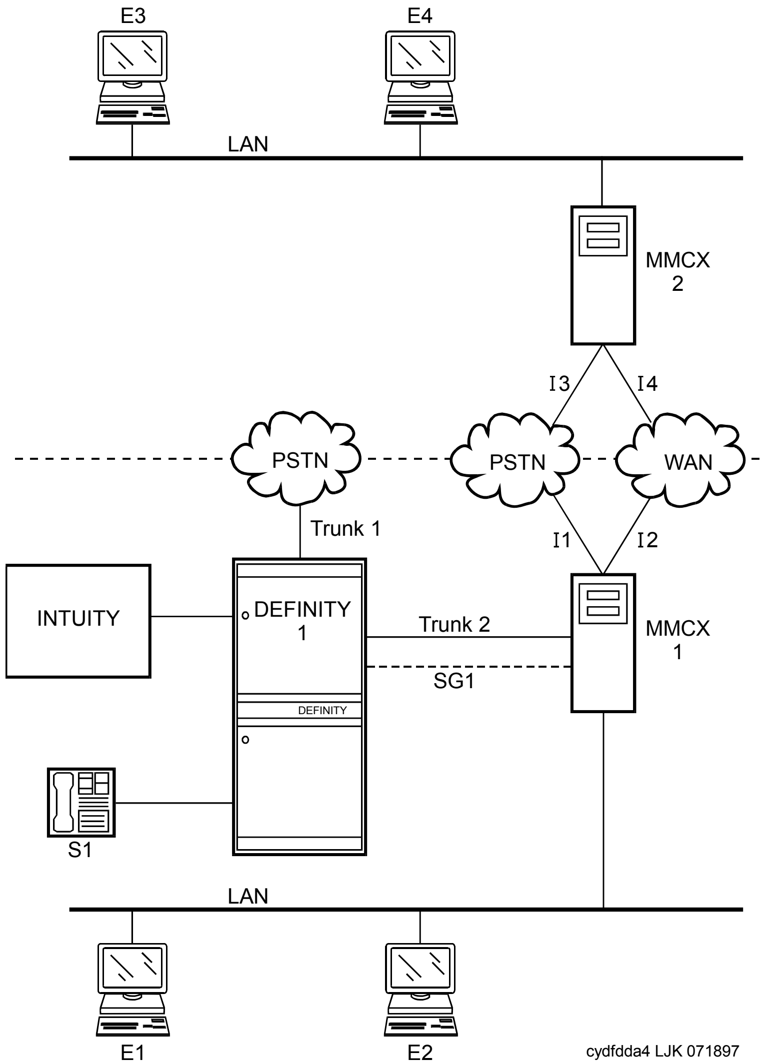


Figure 138: One MASI domain, and one non-MASI MMCX



The MASI node must be directly connected to the Avaya DEFINITY Server for MASI features to work. In this configuration, terminals that belong to MMCX 2 (E3 and E4) do not take advantage of MASI capabilities.

How to administer MASI

This section discusses the administration required to make MASI work. You perform most of this administration from the DEFINITY Server administration terminal. However, there are a few things you must do at the MMCX administration terminal. This section sometimes refers to the **`chgmasi`** command. This is the command you use to administer MASI parameters on the MMCX. For more information about using the **`chgmasi`** command, see your MMCX documentation.

Step 1 — Establishing customer options (Avaya)

An Avaya technical support representative must activate MASI using the **System-Parameters Customer-Options (Optional Features)** screen. The technical support representative should also verify that ISDN-PRI over PACCON (for DEFINITY Server CSI configurations), and AAR/ARS are enabled.

On the MMCX, MASI must be enabled using the **`chgmasi`** command.

Step 2 — Establishing maintenance parameters and alarming options (Avaya)

Ensure that on the **Maintenance-Related System Parameters** screen, the **Packet Bus Activated** field is **y**.

Using the **`set options`** command (Avaya *init* or *inads* logins only), set **MASI alarming options**. For more information, see *Maintenance Procedures for Avaya Communication Manager, Media Gateways and Servers*, 03-300432.

Step 3 — Establishing the physical connection

Establish the physical connection between the Avaya DEFINITY Server and the MMCX.

Step 4 — Administering circuit pack

Using the **DS1 Circuit Pack** screen, verify that the DS1 circuit pack you use to establish the MASI link is administered as follows:

- Bit Rate = 1.544
- Line Coding = b8zs
- Line Compensation = 1
- Signaling Mode = isdn-pri
- Interface = network
- Country Protocol = 1
- Protocol Version = a

Step 5 — Administering a signaling group

For each MASI node, you need to establish a unique signaling group. Use the command **`add signaling-group xxx`** to access the **Signaling Group** screen.

Figure 139: Signaling Group (Page 1 of 5)

```
add signaling-group nnn                                          Page 1 of x
                               SIGNALING GROUP

Group Number  ___            Group Type:  atm___       Name:
                                                Max Number of NCA TSC: ___
                              D-Channel:         Max number of CA TSC:  ___
                                              Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
    TSC Supplementary Service Protocol: _      Network Call Transfer? n

CIRCUIT PARAMETERS
     Virtual Path Identifier: 0
  Virtual Channel Identifier: 0

              Signaling Mode: isdn-pri            Circuit Type: T1
                   Idle Code: 11111111                 Connect: network
        Interface Companding: mulaw
            Country Protocol: 1
            Protocol Version: d

                                   DCP/Analog Bearer Capability:
                                           Interworking Message:
```

For each link, establish a Non-Call Associated Temporary Signaling Connection (NCA-TSC) with the following attributes:

- **Associated Signaling** - MASI requires Facility Associated Signaling, so this field must be set to **y**.
- **Primary D-channel** - Enter a 6- to 7-character port number associated with the DS1 Interface circuit pack port. The port address of the PRI that carries D-channel signaling.

  The port number is used to assign the primary D-channel in the Signaling Group. For 24-channel facilities, the 24th port is assigned as the D-channel. For 32-channel facilities, the 16th port is assigned as the D-channel.
- **Max Number of NCA TSC** - For MASI, this must be 1.
- **Max number of CA TSC** - Leave the default of 0.
- **Trunk Group For NCA TSC** - This can be left blank.
- **Trunk Group for Channel Selection** - This can be left blank.
- **Supplemental Service Protocol** - Values are **a** (AT&T) and **b** (Qsig).
- **Network Call Transfer?** - Values are **y** (yes) and **n** (no).

Figure 140: Administered NCA TSC Assignment page of the Signaling Group screen

```
                                                                    Page 2 of x
                             ADMINISTERED NCA TSC ASSIGNMENT

Service/Feature: _________________     As-needed Inactivity Time-out (min):
  TSC    Local                                                 Adj.      Mach.
 Index    Ext.   Enabled  Established   Dest. Digits     Appl.    Name       ID
  1:     _____      __     _________    _____________   _______  _______     __
  2:     _____      __     _________    _____________   _______  _______     __
  3:     _____      __     _________    _____________   _______  _______     __
  4:     _____      __     _________    _____________   _______  _______     __
  5:     _____      __     _________    _____________   _______  _______     __
  6:     _____      __     _________    _____________   _______  _______     __
  7:     _____      __     _________    _____________   _______  _______     __
  8:     _____      __     _________    _____________   _______  _______     __
  9:     _____      __     _________    _____________   _______  _______     __
 10:     _____      __     _________    _____________   _______  _______     __
 11:     _____      __     _________    _____________   _______  _______     __
 12:     _____      __     _________    _____________   _______  _______     __
 13:     _____      __     _________    _____________   _______  _______     __
 14:     _____      __     _________    _____________   _______  _______     __
 15:     _____      __     _________    _____________   _______  _______     __
```

- **Service/Feature** - Leave blank.
- **As-needed Inactivity Time-out (min)** - This field only applies to as-needed NCA-TSCs. Since MASI requires a permanent connection, leave blank.

- **TSC Index** - This display-only field specifies the administered NCA-TSCs assigned.
- **Local Ext** - Enter a valid, unassigned Avaya Communication Manager extension. This extension does not need a port assignment and does not need to correspond to any other administration.
- **Enabled** - Enter **y** to enable the administered NCA-TSC. You might want to wait to enable this link until all other administration is in place. If this is **y**, Avaya Communication Manager attempts to establish the connection as soon as you submit the form. This might cause your system to alarm, if other administration is not finished.
- **Establish** - Used to indicate the strategy for establishing this administered NCA-TSC. Enter **permanent** for MASI.
- **Dest. Digits** - A valid MMCX extension. This must correspond to the value of the ***`tscnum`*** argument to the **`chgmasi`** command.

  Note:

  These digits are sent as entered to the destination MMCX; no routing or other digit manipulation is performed.
- **Appl.** - Specifies the application this administered NCA-TSC is going to be used for. Enter **masi**.
- **Machine ID** - Used to indicate the MASI node to which this administered NCA-TSC is connected. This number should be the same as the MASI node number found on other screens.

Listing or determining status of TSCs

To determine which TSCs are designated for MASI, use the **`list masi tsc`** command.

Figure 141: MASI Temporary Signaling Connections (TSC) Display

```
                      MASI TEMPORARY SIGNALING CONNECTIONS (TSC)

Sig.  Primary   TSC      Local                                               Mach.
Grp   D-Chan    Index    Ext.   Enabled  Established   Dest. Digits          ID

xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
```

This command displays the following:

- **Sig Grp** — The number of the signaling group to which this TSC belongs
- **Primary D-Channel** — Port location of the Primary D-channel
- **TSC Index** — The number of the MASI TSC within the signaling group

- **Local Ext.** — Avaya Communication Manager extension associated with the TSC
- **Enabled** — Indicates the state of the connection - enabled (y/n)
- **Established** — Value of established flag (as-needed/permanent)
- **Dest. Digits** — The MMCX extension that indicates the TSC destination
- **Mach. ID** — MASI node number

Once you establish and enable the signaling group, you need to verify that it is active. Use the command **`status signaling-group signaling-group#`** or **`status tsc-administered signaling-group# [/tsc-index] [print]`** to determine if the link is active.

Step 6 — Administer ISDN-PRI trunk group

Use the command **`add trunk-group`** ***`xxx`*** to access the **Trunk Group** screen. For a more detailed description of the ISDN-PRI trunk group, see Trunk Group on page 1669.

Establish an ISDN-PRI trunk group with the following attributes:

Page 1

- Group Type = **isdn-pri**
- TAC = valid TAC that conforms to your existing dial plan
- Direction = **two-way**
- Service Type = **tie**
- CDR Reports = **n**

You must also administer the PRI link from the MMCX to the ECS, using the MMCX administration terminal. See your MMCX documentation for information on the **`addpri`** command.

Step 7 — Administer MASI Path Parameters

Use the **`change masi path-parameters`** command to access the **MASI Path Parameters** screen.

Figure 142: MASI Path Parameters screen

```
                          MASI PATH PARAMETERS

Near-End Path Extension: _____

    MASI Node      Trunk Group        Far-End Path Termination Number
        1              __                   __________________
        2              __                   __________________
        3              __                   __________________
        4              __                   __________________
        5              __                   __________________
        6              __                   __________________
        7              __                   __________________
        8              __                   __________________
        9              __                   __________________
       10              __                   __________________
       11              __                   __________________
       12              __                   __________________
       13              __                   __________________
       14              __                   __________________
       15              __                   __________________
```

Establish a MASI Path with the following attributes:

- **Near-End Path Extension** — An unassigned Communication Manager extension. When using the **chgmasi** command to administer the MMCX, this is the farpath extension. See your MMCX documentation for more information.
- **MASI Node** — The node number for the MMCX. For each MMCX/MASI node, this number must be the same everywhere it occurs (**Signaling Group**, **MASI Trunk Group**, and **MASI Terminal** screens).
- **Trunk Group** — This is the trunk group number in Communication Manager for the ISDN-PRI trunk that will be used to establish call paths.
- **Far-End Path Termination Number** — This is an unassigned MMCX extension. When using the **chgmasi** command to administer the MMCX, this is the nearpath extension. See your MMCX documentation for more information.

Step 8 — Administer MASI trunk groups

Use the **MASI Trunk Group** screen to define MMCX interfaces that interconnect MASI nodes, or that connect MMCX nodes to another private switch or central office. Examples of MMCX interfaces include:

- PRI trunks linking MMCX servers
- PRI trunks linking MMCX to the PSTN

- PRI trunks from MMCX to Avaya Communication Manager that are used for purposes other than MASI
- LAN interfaces linking MMCX servers

Use the command **`add masi trunk-group xxx (or 'next')`** to access the **MASI Trunk Group** screen. The trunk group number must not be assigned, and you cannot exceed the maximum total trunks for your system. Valid values for xxx are unused trunk group numbers in Avaya Communication Manager between 1 to 96 for DEFINITY Server CSI configurations.

Figure 143: MASI Trunk Group screen

```
                            MASI TRUNK GROUP

Group Number: 15                                           CDR Reports? y
  Group Name: INCOMING CALL______________ COR: 1_     TN: 1        TAC: 915_

                             MASI Node Number: __  Remote Group Number: _
```

- **Group Number** - This field displays the MASI trunk group number. This is the number assigned when executing the **`add masi trunk-group`** command.
- **CDR Reports** - Valid entries are **y**, **n**, and **r**. Default is **y**.
  - If you enter **y**, Call Detail Recording (CDR) records will be generated by completed outgoing calls terminated on this trunk group. If incoming calls are being recorded (the **Record Outgoing Calls Only** field on the **CDR System Parameters** screen is set to **n**), then a single CDR record will be generated for answered calls with the call duration.
  - If you enter **n**, no CDR records will be generated by calls originated by or terminated on this trunk group.
- **Group Name** - Enter a unique name that identifies the trunk group. Up to 27 characters can be used; default is "**INCOMING CALL**."
- **COR** - Enter a Class of Restriction (COR) number (0 to 995) that reflects the desired restriction; default is **1**.
- **TN** - This field displays the Tenant Partition number. All MASI trunks are associated with Tenant 1.
- **TAC** - Enter the trunk access code (TAC) that identifies the trunk group on CDR reports. You must assign a different TAC to each MMCX interface. Valid entries conform to the dial plan (1 to 4 digits, * and # are valid first digits).
- **MASI Node Number** — The node number assigned to this MMCX machine.
- **Remote Group Number** — This is the number of the remote trunk group. For ISDN-PRI interfaces, valid values are any number 1 to 8; for local area network (LAN) or WAN calling interfaces, the value must be 9. The combination of MASI Node Number and Remote Group Number must be unique. Remote group number corresponds to the group number on the MASI node.

Viewing a list of all MASI trunk groups

To view a list of all the MASI trunks administered on the ECS, use the command **`list masi trunk-group`**.

Figure 144: List masi trunk-group output

```
                              MASI TRUNK GROUP

Group                                          Node    Remote
Number  TAC   Group Name                       Number  Grp No.  CDR  COR  TN

xxx     xxxx  xxxxxxxxxxxxxxxxxxxxxxxxxxx      xx      x        x    xx   xxx
```

Determining the status of MASI trunk groups

To determine the status of a specific MASI trunk, use the command **`status masi trunk-group`** ***`xxx`***, where ***`xxx`*** is the trunk group number. This command provides descriptive information about the trunk, and the number of currently active trunk calls.

Figure 145: Status masi trunk-group output

```
                             MASI TRUNK GROUP STATUS

        Group Number: xxx       Number of Active MMCX Trunk Calls: xxx
    MASI Node Number: xx
 Remote Group Number: xxx
```

Step 9 — Administer MASI terminals

Use the **`add masi terminal`** ***`xxxxx`*** or **`next`** command to administer each MASI terminal as a MASI terminal. You use available extensions on the ECS, so they need to conform to the Avaya Communication Manager dial plan. The extension must match the Communication Manager dial plan, and for the **`add`** command, the extension must not already be in use.The extension of the MASI terminal must match the number of the MASI terminal.

Avaya Communication Manager users dial the MASI Terminal Extension to reach MMCX users.

Note:

Anytime you add a terminal or other extension to the MMCX, you must administer a corresponding MASI terminal on Avaya Communication Manager. If you do not, you will not be able to dial this extension from Avaya Communication Manager.

Figure 146: MASI Terminal screen — page 1

```
add masi terminal next

                           MASI TERMINAL

 Extension: 1000                                                   BCC: 0
                                              MASI Node Number: __ TN: 1__
                                                                  COR: 1_
      Name: ___________________________


 TERMINAL OPTIONS

        Send Display Info? y
```

- **Extension** — This field displays the extension that you entered on the command line.
- **BCC** — This field displays the bearer capability class of the terminal, and identifies the type of traffic the terminal supports. For MASI, this is always 0, for voice or voice-grade data.
- **MASI Node Number** — The number of the node on which this terminal resides.
- **TN** — The tenant partition in which this terminal resides. At present, all MASI terminals must reside within tenant 1. This field is display-only, and always 1.
- **COR** — The class of restriction associated with this terminal.
- **Name** — The name associated with the terminal. This can be any alphanumeric string up to 27 characters.
- **Send Display Info** — Indicates whether Avaya Communication Manager should forward display information associated with a call. Set to **y**.

Figure 147: MASI Terminal screen — page 2

```
                            MASI TERMINAL

 FEATURE OPTIONS
            LWC Reception: none___

              CDR Privacy? n
```

- **LWC Reception** — This field indicates whether the terminal can receive Leave Word Calling (LWC) messages. Valid values are **none**, **audix**, and **mas-spe** (for DEFINITY Server CSI configurations). SPE-based LWC is not supported for MASI terminals. However, if embedded AUDIX is used without a Data Control Link, you must administer MASI terminals to receive SPE-based LWC messages. For such cases, the LWC feature is used by AUDIX messaging systems to activate and deactivate message waiting lamps on MASI terminals.
- **CDR Privacy** — Indicates whether CDR Privacy is supported for this terminal. See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for more information.

Figure 148: MASI Terminal screen — page 3

```
                            MASI TERMINAL
SITE DATA
       Room: _______
       Jack: _____
      Cable: _____
      Floor: _______
   Building: _______







BUTTON ASSIGNMENTS
 1: call-appr
```

- **Room** - Enter up to 10 characters to identify the MASI terminal location. This field can be blank.
- **Jack** - Enter up to 5 characters to identify the location where the MASI terminal is connected. This field can be left blank.
- **Cable** - Enter up to 5 characters to identify the cable that connects the MASI terminal to the system. This field can be left blank.
- **Floor** - Enter up to 7 characters to identify the floor where the MASI terminal is located.
- **Building** - Enter up to 7 characters to identify the building where the MASI terminal is located. Valid entries are listed in the site table.
- **BUTTON ASSIGNMENTS** — This field contains a call appearance button and is display only.

Duplicating MASI terminals

Once you have one MASI terminal administered to your liking, you can use the **`duplicate masi terminal`** command to administer other stations with the same characteristics.

Figure 149: Duplicate MASI Terminal screen

```
                      MASI TERMINAL


Ext        Name                           Room      Jack  Cable
77777      ___________________________    _________ _____ _____
77778      ___________________________    _________ _____ _____
77779      ___________________________    _________ _____ _____
77781      ___________________________    _________ _____ _____
77782      ___________________________    _________ _____ _____
77783      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
_____      ___________________________    _________ _____ _____
```

Listing MASI terminals

To view a list of all the MASI terminals administered on a server, use the command **`list masi terminals`**.This command only lists terminals within the domain of the Avaya DEFINITY Server from whose SAT you issue the command.

Figure 150: List MASI Terminal output

```
                         MASI TERMINALS


                                     Node
       Ext   Name                    Number CDR COR TN

       xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
       xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
       xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
       xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
```

To view the active or idle status of a specific MASI terminal, use the command **`status masi terminal (extension)`**.

Figure 151: Status MASI terminal command

```
                            GENERAL STATUS

                  TYPE: MASI        Service State: active
             Extension: 54001
       MASI Node Number: 14
```

To determine which extension you assigned as the MASI Near-End Path Termination extension, use the command **`list extension-type`**. This command displays the extension number and type (attendant, masi-terminal, etc.), as well as other information about the extension.

Figure 152: List extension type

```
                           EXTENSION TYPE

                                                             Cv1/
 Ext   Type                 Name                   COR  TN   COS  Cv2
 ---   ----                 ----                   ---  --   ---  ----
 1234  masi-terminal                               1    1    1
 4077  term-masi-path-call
```

Step 10 — Administer features

AAR/ARS

1. AAR/ARS is an optional feature on Avaya Communication Manager, and you need to purchase this option to use it with MMCX. To verify that this feature is enabled, use the command **`display system-parameters customer-options`**. If it is not enabled, contact your Avaya representative.
2. The MMCX dial plan must use the same feature access codes as Avaya Communication Manager. If this is not already the case, modify the MMCX dial plan using the **`chgdp`** command. See your MMCX documentation for more information.
3. Include this feature access code in the **`chgmasi`** command.

CDR

1. To get call detail records for calls over MMCX interfaces, set **CDR Reports = y** on the **MASI Trunk Group** screen.
2. To get call records for calls over the ISDN-PRI trunk group, set **CDR Reports = y** on the **ISDN-PRI Trunk Group** screen.
3. To track calls between a MASI terminal and other MASI terminals or Communication Manager stations, enter the MASI terminal extension on the **Intra-switch CDR** screen.
4. Enter **n** in the **Record Non-Call Assoc TSC** field on the **CDR System Parameters** screen.

Note:

If you use the same PRI trunks for MASI and non-MASI calls, Avaya strongly recommends that you do not enable CDR for these calls. Establish a separate trunk group for non-MASI calls and set CDR Reports = **n**.

Coverage

To establish coverage from a MASI terminal to AUDIX:

1. Use the MMCX user interface to enter the AUDIX hunt group extension as the coverage point. You cannot use Avaya Communication Manager coverage administration for MASI terminals.
2. If AUDIX ports are not administered in Avaya Communication Manager, you must administer them.
3. Set up the MASI terminal as an AUDIX subscriber. Enter the MASI terminal extension in the **Extension** field on the **Subscriber Administration** screen.

To establish coverage from a MASI terminal to another MMCX terminal or Avaya Communication Manager station:

1. Use the MMCX user interface to enter the desired extension as the coverage point for the MASI terminal. You cannot use Avaya Communication Manager coverage administration for MASI terminals.

Step 11 — Verify administration

You should make test calls from Avaya Communication Manager to MMCX, to ensure that you can indeed place and receive calls.

Call an unattended MASI terminal. Verify that the call goes to AUDIX. Retrieve the call from the MASI terminal. Verify that all works as expected.

Setting MASI command permissions

If you are the super-user for your system, you can restrict other administrative logins from changing MASI administration. To do this, use the `change permissions (login-ID)` command. Enter **y** in the **Additional Restrictions** field, then move to the **Restricted Object List** page of the screen.

You can restrict the following MASI-related objects:

- masi-path-parameters
- masi-terminal
- masi-trunk-group
- masi-tsc

Using MASI with Communication Manager features

AAR/ARS

MMCX can take advantage of advanced routing features for voice-only calls to the public switched telephone network (PSTN) or an Avaya private network. Users must enter the AAR/ARS access code before the rest of the dialed digits. MASI will route the call over the Communication Manager private network (AAR) or the public network (ARS), based on digits supplied by the MMCX user.

Routing patterns must contain only trunk groups that actually terminate to Avaya Communication Manager. Calls from one MMCX to another MMCX do not use AAR/ARS. Authorization codes are not supported.

Call Detail Recording

Using the MASI link, Avaya Communication Manager is able to track call detail information for calls made using MMCX terminals and interfaces. CDR records all calls originating from or terminating at a MASI terminal. MASI CDR does not record ineffective call attempts when all MASI paths are busy.

The **Resource Flag** value of **8** indicates a MASI call. This field appears in unformatted, int-isdn, expanded and customized CDR formats. For formats other than these, you can determine that a call involves a MASI terminal or trunk by the trunk access code (TAC), dialed number or calling number fields.

The following are the CDR capabilities of MASI. Administration information is under the heading How to administer MASI on page 343.

- Incoming/Outgoing Trunk Call Splitting

  Call splitting does not produce separate records for MMCX calls that are transferred or conferenced.

- intra-switch CDR

  You can administer intra-switch CDR to monitor MASI terminals. To do this, simply add the MASI terminal extension on the **Intra-switch CDR** screen. Avaya Communication Manager then monitors calls from MASI terminals to other MASI terminals, and calls between MASI terminals and Communication Manager stations.

- CDR Privacy

  You can administer a MASI terminal for CDR Privacy.

- Account Code Dialing and Forced Entry of Account Codes

  This is not supported for MASI terminals. Therefore, make sure the COR you assign does not force entry of account codes.

- Trunk CDR

  You can get call detail records for all incoming and outgoing calls made over MMCX interfaces.

Call redirection / Voice-messaging access

MMCX users can enter an Avaya Communication Manager extension, including an AUDIX hunt group, Callmaster agent, attendant console or telephone as their coverage point.

If AUDIX is established as the MASI terminal's coverage point, the MASI terminal receives message waiting indication, and dials the AUDIX hunt group extension to retrieve messages. Once connected to AUDIX, operation for the MMCX user is the same as for a Communication Manager station user, including use of # to identify the extension, if desired.

Note:

It is not possible to determine the call coverage status of a MASI terminal.

Avaya Communication Manager tracks calls to MASI terminals that follow the autonomous coverage path from the MASI terminal. MMCX calls redirected to Communication Manager stations contain display information.

MASI terminals that dial AUDIX directly, or that place calls to MASI terminals that cover to AUDIX, do not receive ringback if all AUDIX ports are busy. Instead, these callers see a message the called party is busy, and the call drops.

Transfer

MASI terminals cannot transfer calls to Communication Manager stations, and cannot transfer a call to another MASI terminal if the call involves a Communication Manager station.

Conferencing

Conferences can involve both MASI terminals and Avaya Communication Manager stations, and either one can initiate the conference. Communication Manager stations participate in such conferences in voice-only mode. If an MMCX user initiates a conference that involves Communication Manager stations, the conference will drop when the initiator drops from the call. If a Communication Manager station initiates the conference, that station can drop without affecting the other conferees.

Status tracking - terminals and trunks

Avaya Communication Manager tracks the active/idle status of all MASI terminals, and monitors traffic over MMCX interfaces.

Trunk groups

For MASI purposes, there are two kinds of trunk groups: the ISDN-PRI trunk groups that serve as paths for establishing calls between Avaya Communication Manager stations or trunks and MASI terminals or interfaces, and the remote trunks that are interfaces from the MMCX to other entities. Each MASI remote trunk group appears to Communication Manager as a single unit, with no concept of members within the group.

Note:

You cannot test, busy out, or release MASI remote trunk groups, since you cannot dial a MASI remote trunk TAC from the Avaya DEFINITY Server. The TAC merely identifies the trunk to Avaya Communication Manager for purposes of status and CDR records.

You cannot administer MASI trunks as part of Communication Manager route patterns.

Unsupported Communication Manager Features

We can generalize feature interactions to some extent. For example, since there are no buttons available to a MASI terminal, any feature that requires a button is also not available. MASI cannot support features that require the user to dial a trunk access code for a MASI remote trunk, or a feature access code other than AAR/ARS. The MMCX dial plan can contain only those feature access codes that are supported. Note the following CAUTION.

CAUTION:

DO NOT ADMINISTER the following features! The following features are not supported for use over the MASI link, and Avaya cannot be responsible for the results if you attempt to administer them.

Unsupported Call Center features

- ASAI — You must not administer a MASI terminal in an ASAI domain. MASI terminals and MMCX trunks are not monitored by ASAI. It might be possible for a MASI terminal to place a call to a Communication Manager station that is part of an ASAI domain. ASAI will not be blocked from controlling this call, but there can be unpredictable results. The same is true for calls originating from an ASAI domain terminating at MASI terminals, and for ASAI-monitored hunt groups that contain MASI terminals.
- Automatic Call Distribution — You must not include a MASI terminal extension as part of an ACD hunt group. You must not mix MASI administration with anything related to ACD, including Outbound Call Management and PASTE.
- Call Vectoring — You must not include MASI terminal extensions in any step of a vector.

Unsupported Basic features

- Bridged Call Appearances — You must not administer a bridged appearance that involves a MASI terminal.
- Call Coverage — You must not administer a MASI terminal in the coverage path of an Avaya Communication Manager station.
- Call Forwarding — You must not forward a Communication Manager station to a MASI terminal.
- Call Pickup — You must not administer a MASI terminal as part of a pickup group.
- Intercom — You must not administer MASI terminals as members of any type of intercom group.
- Manual Message Waiting — You must not administer a manual message waiting button (**man-msg-wt**) with a MASI terminal as the referenced extension.
- Manual Signaling — You must not administer a manual signaling button (signal) with a MASI terminal as the referenced extension.

- Night Service — You must not administer a MASI terminal as a night service destination.
- Pull transfer — MASI terminals cannot perform a pull transfer operation. You must not administer this feature on an Avaya DEFINITY Server where MASI is active. This applies only in Italy.
- Station Hunting — You must not administer a MASI terminal as part of a station hunting path.
- Terminating Extension Groups — You must not administer a MASI terminal as part of a TEG.

Other interactions

The following section describes feature behaviors that might not be as expected, but that are not likely to be destructive.

Attendant features

- Dial Access to the Attendant — MASI terminals will be able to dial the attendant access code, if it is administered in the MMCX dial plan.
- Attendant Direct Extension Selection — Attendants are able to access MASI terminals via DXS buttons and busy lamp indicates status of the MASI terminal.
- Emergency Access to the Attendant — MASI terminals have emergency access using the attendant access code, if it is administered in the MMCX dial plan. However, off-hook alerting is not administrable.
- Attendant Intrusion — Attendants are able to activate intrusion towards MASI terminals.
- Attendant Override — Attendants are not able to activate override towards MASI terminals.
- Attendant Recall — MASI terminals cannot activate attendant recall.
- Attendant Remote Trunk Group Select — Attendants cannot use this feature to select MASI remote trunks.
- Attendant Return Call — Operates normally if a MASI terminal is the called party.
- Attendant Serial Call — Serial calls are denied if the calling party is an MMCX interface.
- Attendant Straightforward Outward Completion — The attendant is able to complete calls to Communication Manager trunks for MASI terminals.
- Attendant Through Dialing —The attendant can use Through Dialing to pass dial tone to MASI terminals.
- Attendant Timers — Attendant timers work the same no matter what kind of terminal is involved.

- Attendant Trunk Group Busy/Warning Indicators — You cannot administer Busy/Warning indicators for MASI trunks because they are not standard Avaya Communication Manager trunks. However, you can administer these indicators for the trunk group administered for MASI paths.
- Attendant Trunk Identification — The attendant is not able to identify the trunk name via button pushes.

Basic features

- Abbreviated Dialing — A Communication Manager station can enter an MMCX extension in an AD list. However, MASI terminals cannot use AD.
- Administered Connections — MASI terminals must not be the originator nor the destination of an administered connection.
- Automatic Callback — Automatic callback does not work towards a MASI terminal.
- Automatic Circuit Assurance — You must not administer a MASI terminal as an ACA referral destination. You cannot administer ACA for MASI remote trunks.
- Busy Verification of Terminals and Trunks — You cannot use Busy Verification for MASI terminals or remote trunks.
- Call Detail Recording — CDR Account Code Dialing and Forced Entry of Account Codes are not supported for MASI terminals.
- Call Park — The attendant can park calls at the extension of a MASI terminal, but users can only retrieve these calls from a Communication Manager station, since MASI terminals cannot dial the Answer Back FAC.
- Data Call Setup — Avaya Communication Manager users cannot place data calls to MASI terminals.
- Facility Busy Indication — You can use FBI to track the status of MASI terminals. The FBI button and indicator lamp must be on a Communication Manager station. You cannot use FBI to track MMCX interfaces.
- Facility Test Calls — Avaya Communication Manager users cannot make test calls to MMCX interfaces.
- Go to Cover — MASI terminals cannot activate this feature.
- Leave Word Calling — The only valid LWC destination for a MASI terminal is AUDIX. You cannot administer SPE-based LWC. MASI terminals cannot send LWC messages to Avaya Communication Manager stations or to MASI terminals.
- Loudspeaker paging — You can administer a MASI terminal as a code calling extension.
- Malicious Call Trace — MASI terminals cannot initiate malicious call trace.
- Message Retrieval — MMCX users can only retrieve messages through AUDIX messaging.

- Music on Hold — Music on hold will only be available if an Avaya Communication Manager station has placed the call on hold.
- Override — Executive override does not work towards MASI terminals.
- Priority Calling — Priority calling is not supported for calls to or from MASI terminals.
- Ringback Queueing — Ringback Queueing is not supported for MASI terminals.
- Send All Calls — MMCX has an autonomous SAC function.
- Tenant Partitioning — All MASI terminals exist in tenant 1, and you cannot change the tenant number.
- Time of Day coverage — As with all coverage, Avaya Communication Manager does not control coverage of the MASI terminal.
- Transfer out of AUDIX — A MASI terminal cannot use *T to transfer from AUDIX to another MASI terminal.

Hospitality features

- Do Not Disturb — MASI terminals cannot activate Do Not Disturb.

Multimedia features

- Multimedia Call Handling — Avaya MMCH users are not able to make H.320 calls to MASI terminals over the MASI link. Calls between MMCX terminals and MMCH terminals are voice only.

Troubleshooting

Verify proper operation using the following commands and follow normal escalation procedures to resolve any failures detected by the demand test.

1. Verify the DS1 trunk using the **`test board <board location> long`** command.
2. Verify the ISDN Signaling Group using the **`test signaling-group <group number>`** command. Also verify proper administration.
3. Verify the temporary signaling connection using the **`test tsc-administered <group number>`** command. Also verify proper administration.

Common error conditions

If the cable from an Avaya DEFINITY Server to the MMCX becomes disconnected, you should see alarms raised against ISDN-SGRP and UDS1-BD. In particular, you should observe ISDN-SGRP errors such as 769, 1793, and 257. To resolve, reconnect the cable and follow normal test procedures.

If the far-end path termination number is incorrect, you should observe MASI-PTH error 513. To resolve, correct administration using the **MASI Path Parameters** screen.

If the Layer 3 TSC is not administered properly or is out of service, you should observe errors (but no alarms) raised against TSC-ADM. Verify the signaling group administration and follow normal escalation procedures for TSC-ADM.

If the TSC fails to come up even through Layer 2 Signaling Group and below pass tests, you can run **`test tsc-administered <group number>`** to force a server heartbeat test, or simply wait 5 to 10 minutes for the link to recover. This situation might occur if the server running Communication Manager is rebooted or if the MASI interface is administered before the MMCX is properly administered.

You might want to use the **`busy port`** and **`release port`** commands to unlock things if features are not working.

Avaya Video Telephony Solution

Use the Avaya Video Telephony Solution (AVTS) to enable videoconferencing for your desktop and group video communications.

Note:

AVTS is Avaya's newest, and currently available H.323 video solution. Some older systems may still use the older technology H.320 video solution, Multi-Media Call Handling (MMCH). For more information on MMCH, see Multimedia Call Handling on page 375.

Detailed description of Avaya Video Telephony Solution

The Avaya Video Telephony Solution enables Avaya Communication Manager to merge a set of enterprise features with Polycom's videoconferencing adjuncts. It unifies Voice over IP with video, web applications, Avaya's video enabled IP softphone, third party gatekeepers and other H.323 endpoints.

The following components are part of the Avaya Video Telephony Solution feature:

- Polycom VSX3000, VSX7000 and VSX8000 conferencing systems with Release 8.03 or later
- Polycom V500 video calling systems
- Polycom MGC video conferencing bridge platforms with Release 8.0.1.
  Release 7.5 of the MGC is not supported.
- Third party gatekeepers, including Polycom Path Navigator

You also need a system running Avaya Communication Manager Release 3.0.1, and Avaya IP Softphone release 5.2 with video integrator.

Starting with Communication Manager Release 3.1.2, you can use cumulative bandwidth management to set video bandwidth for the Avaya Video Telephony Solution.The Audio Call Admission Control (CAC) capability allows you to set maximum bandwidth between multiple network regions for audio calls. Video bandwidth can also be controlled in a similar way.

For more information, see also:

Avaya Video Telephony Solution Release 3.0 Networking Guide, 16-601423, Issue 1
Video Telephony Solution Release 3.0 Quick Setup, 16-300310, Issue 3
IP Softphone and Video Integrator Getting Started, 16-600748, Issue 2

Note:

To configure the Polycom MGC-25 Video Conferencing Bridge Platforms with Avaya S8300, S8500 and S87xx Servers, see the procedures stated in the *Video Telephone Solution R3.0 Quick Set Up Guide*, 16-300310, Issue 3, February 2007.

Administering the Avaya Video Telephony Solution

The following steps are part of the administration for the Avaya Video Telephony Solution:

- Configuring the Polycom VSX Video Conferencing Systems and V500 Video Calling Systems
- Configuring Polycom PathNavigator Gatekeepers
- Configuring video trunks between two Avaya Communication Manager systems
- Configuring the Maximum Bandwidth for Inter-Network Regions
- Checking bandwidth usage
- Administering Ad-hoc Video Conferencing

This section describes:

- Any prerequisites for administering the Avaya Video Telephony Solution
- Complete administration procedures for the Avaya Video Telephony Solution

Prerequisites for administering Avaya Video Telephony Solution

You must complete the following actions before you can administer the Avaya Video Telephony Solution:

- Type **`display system-parameters customer-options`** to view the **System Parameters Customer-Options (Optional Features)** screen. Page down till you see the **Maximum Video Capable Stations** field and the **Maximum Video Capable IP Softphones** field. These two fields show up only if your system is licensed for the Avaya Video Telephony feature. Your Avaya license file must contain the RTUs that were purchased for **Maximum Video Capable Stations** field and the **Maximum Video Capable IP Softphones** fields.

  Note:

  You must make sure that the value of the **Maximum Video Capable Stations** field allows for each station that you use. In addition, each single-point VSX system is considered to be *one* station, and each multipoint VSX system is considered to be *three* stations.

- Type **`change ip-network-region`** # to view the **IP Network Region** screen. The following fields must be set to **y** on this screen:

- **Intra-region IP-IP Direct Audio**
- **Inter-region IP-IP Direct Audio**
- **IP Audio Hairpinning**

Configuring Video-Enabled Avaya IP Softphone Endpoints

To configure video-enabled Avaya IP Softphone endpoints:

1. Type the `display system-parameters customer-options` command and verify number on the **Maximum Video Capable IP Softphones**. This number is provided by the Communication Manager license file.
2. Type `change ip-codec-set` ***x*** command (where ***x*** is the chosen IP codec set) to set the following parameters:
   - **Allow Direct-IP Multimedia** to **y**.
   - **Maximum Call Rate for Direct-IP Multimedia** - the Call Rate is the combined audio and video transmit rate or receive rate. You can use this setting to limit the amount of bandwidth used for calls. For example, if you select 768 Kbits, a maximum of 768 Kbits will be used to transmit and to receive audio and video.
   - **Maximum Call Rate for Priority Direct-IP Multimedia** allows you to set the maximum call rate per call for priority users
   - Repeat this step for each IP codec set that will be used for video.
3. Type `change cos` and page down till you find the **Priority IP Video** field. This must be set to **y** for each class of station that is given a Priority status.
4. Type `change ip-network-region` ***x*** command (where ***x*** is the chosen IP network region) to set the following parameters:
   - **Intra-region IP-IP Direct Audio** to **yes**.
   - **Inter-region IP-IP Direct Audio** to **yes**.
   - **Security Procedures 1** to **any-auth**.
   - Repeat this step for each IP network region that will be used for video.
5. Type `add station` command to add an Avaya IP Softphone station, and set the following parameters for that station:
   - **IP Softphone** to **y**.
   - **IP Video Softphone** to **y**.
   - **IP Audio Hairpinning** to **y**.
   - Repeat Step 5 for each video-enabled Avaya IP Softphone endpoint you want to configure.

Configuring the Polycom VSX Video Conferencing Systems and V500 Video Calling Systems

To configure the Polycom VSX Video Conferencing systems and the V500 Video Calling Systems:

1. You must know the following information:
   - Maximum number of VSX and V500 systems on your network
   - PIN for each VSX/V500 system. The default is the unit's serial number.
   - Polycom software key for each system
   - Avaya option key for each system
   - Whether the VSX system has the multipoint option or IMCU option
   - IP address of the voice system
2. Use the **`display system-parameters customer-options`** command to verify the **Maximum Video Capable Stations**. This number is provided by the Communication Manager license file. The Maximum Video Capable Stations is determined by using the following criteria:
   - Each V500 system is considered to be **one** station.
   - Each single-point VSX system is considered to be **one** station.
   - Each VSX multipoint system is considered to be **three** stations.
3. Use the **`change ip-codec-set`** ***x*** command (where ***x*** is the chosen IP codec set) to define the following wideband codecs:
   - SIREN14-S96K (1 fpp, 20 ms)
   - G722.1-32K (1 fpp, 20 ms)
   - G.726A-32K (no silence suppression, 2 fpp, 20 ms)
   - G.711MU (no silence suppression, 2 fpp, 20 ms)
   - G.729A (no silence suppression, 2 fpp, 20 ms)
   - Set **Allow Direct-IP Multimedia** to **y**
   - Set **Maximum Call Rate for Direct-IP Multimedia -** the Call Rate is the combined audio and video transmit rate or receive rate. You can use this setting to limit the amount of bandwidth used for calls. For example, if you select 768 Kbits, a maximum of 768 Kbits will be used to transmit and receive audio and video. Repeat this step for each IP codec set that will be used for video.
   - **Maximum Call Rate for Priority Direct-IP Multimedia** allows you to set the maximum call rate per call for priority users
4. Use the **`change ip-network-region`** ***x*** command (where ***x*** is the chosen IP network region) to set the following parameters:

- **Intra-region IP-IP Direct Audio** to **yes**
- **Inter-region IP-IP Direct Audio** to **yes**
- **Security Procedures 1** to **any-auth**
- Repeat this step for each IP network region that will be used for video.

5. Use the **`add station`** command to add a station for the Polycom system to set the following parameters:
   - **Type** to **H.323**
   - **Security Code** to the “pin” on the VSX or V500 system
   - **IP Video** to **y**
   - **IP Audio Hairpinning** to **y**.
6. If the VSX system has the multipoint option or IMCU option, perform the following steps:
   a. Use the **add station** command to add a second station for the Polycom system.
   b. Set **Type** to **H.323**.
   c. Set **Security Code** to the “pin” on the VSX. Make sure the security code is the same as the previous station. All three stations must have the same security code.
   d. Set **IP Video** to **y**.
   e. Repeat Steps a through e to create the third consecutive station.
   f. Use the **`change station`** ***`xx`*** command (where ***`xx`*** is the first station you added for the Polycom system) to set **Hunt-to Station** to the second station you added for the Polycom system.
   g. Use the **`change station xx`** command (where ***`xx`*** is the second station you added for the Polycom system) to set **Hunt-to Station** to the third station you added for the Polycom system.
   h. Use the **`change station`** ***`xx`*** command (where ***`xx`*** is the third station you added for the Polycom system) to set **Hunt-to Station** to the first station you added for the Polycom system. All three stations must be in a circular hunt.
7. Install the Polycom system and connect it to your network.
8. Upgrade the Polycom system software.
9. Using a web browser, access the Polycom home page for the unit, and select **Admin Settings**>**Network**>**IP Network**.
10. Select the **Enable IP H.323** check box.
11. Select the **Display H.323 Extension** check box.
12. In the H.323 Extension (E.164) box, enter the station number you specified for this system on the Avaya Communication Manager system.
13. From the Use Gatekeeper box, select **Specify with PIN**.

14. In the Gatekeeper IP Address box, enter the IP address of the CLAN or PCLAN followed by ***:1719*** to specify the correct port that must be used.
15. In the Authentication PIN box, enter the security code you entered in Step 4.
16. In the Number box in the Gateway area, enter the extension you specified in Step 10.
17. In the Type of Service box in the Quality of Section area, select IP Precedence.
18. In the Type of Service Value boxes (Video, Audio, and Far End Camera Control), enter the QoS values for the IP Network Region settings in which the VSX station belongs.
19. Select the **Enabled PVEC** check box.
20. Select the **Enable RSVP** check box.
21. Select the **Dynamic Bandwidth** check box.
22. From the Maximum Transmit Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Avaya Communication Manager system.
23. From the Maximum Receive Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Avaya Communication Manager system.
24. Complete the Firewall and Streaming sections as necessary.
25. When finished, click the **Update** button.
26. Repeat the steps for each Polycom system.

Configuring Polycom PathNavigator Gatekeepers

To configure a Polycom PathNavigator gatekeeper:

1. Use the **change ip-codec-set 1** command to set the following parameters:
   - **Allow Direct-IP Multimedia** to **y** (page 2 of screen).
   - **Maximum Call Rate for Direct-IP Multimedia**. This setting is the combined audio and video transmit rate or receive rate for non-priority (normal) video calls. You can use this setting to limit the amount of bandwidth used for normal video calls. For example, if you select 384 Kbits, a maximum of 384 Kbits will be used to transmit *and* to receive audio/ video.
   - **Maximum Call Rate for Priority Direct-IP Multimedia**. This setting is the combined audio and video transmit rate or receive rate for priority video calls. You can use this setting to limit the amount of bandwidth used for priority video calls. For example, if you select 384 Kbits, a maximum of 384 Kbits will be used to transmit *and* to receive audio/ video.
2. Use the **change ip-network-region** command to put the gatekeeper in its own network region. Set the following parameters:
   - **Intra-region IP-IP Direct Audio** to **no**.

- **Inter-region IP-IP Direct Audio** to **no**.
- **Security Procedures 1** to **any-auth** (page 2 of screen).
- **Video Norm** (page 3 of screen) to the amount of bandwidth that you want to allocate for the normal video pool to each IP network region.
- **Video Prio** (page 3 of screen) to the amount of bandwidth that you want to allocate for the priority video pool to each IP network region.
- **Video Shr** (page 3 of screen). Specify whether the normal video pool can be shared for each link between IP network regions.

Note:

If one of the video bandwidth limits is in Kbits, and another video bandwidth limit is in Mbits, all of the video bandwidth limits will be converted to the same unit (that is, Kbits or Mbits).

3. Use the **change node-names ip** command to add an entry for the Polycom PathNavigator gatekeeper. Be sure to enter the IP address of the IP board for the gatekeeper.
4. Use the **add signaling-group** command to add a signaling group for the gatekeeper. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Near-end Listen Port** to **1719**.
   - **LRQ Required** to **y**.
   - **Incoming Priority Video.** If you want all incoming calls to receive priority video transmissions, select **y**.
   - **Far-end Node Name** to the name you entered for the gatekeeper in Step 3.
   - **Far-end Listen Port** to **1719**.
   - **Far-end Network Region** to the IP network region you specified in Step 2.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
5. Use the **add trunk-group** command to add a trunk group for the gatekeeper. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Carrier Medium** to **IP**.
   - Add members to this trunk group.
6. Use the **change signaling-group *xx*** command (where ***xx*** is the signaling group you added in Step 4) to set **Trunk Group for Channel Selection** to the trunk group you added in Step 5.
7. Create a route pattern to the gatekeeper.

8. Configure the gatekeeper.

Configuring video trunks between two Avaya Communication Manager systems

To configure video trunks between two Communication Manager systems:

1. Use the `change ip-codec-set 1` command to set the following parameters:
   - Set **Allow Direct-IP Multimedia** to **y** (page 2 of screen).
   - Set **Maximum Call Rate for Direct-IP Multimedia -** the Call Rate is the combined audio and video transmit rate or receive rate. You can use this setting to limit the amount of bandwidth used for calls.
   - **Maximum Call Rate for Priority Direct-IP Multimedia** allows you to set the maximum call rate per call for priority users
2. Type **`display route-pattern`** ***`xxx`***, where ***`xxx`*** is the number for the route pattern.

   To enable multimedia, the **M** field under **BCC** value must be set to **y**. This will allow you to send multimedia calls over a specific trunk.
   It is possible to have video over trunks that do not have **M** field set for the **BCC**. Setting **M** on the **BCC** enables you to select the route that the route pattern that you should use.
3. Use the `change node-names ip` command to add an entry for the trunk. Be sure to enter the IP address of the CLAN or PCLAN of the other Communication Manager system.
4. Use the `add signaling-group` command to add a signaling group for the video trunk. Set the following parameters:
   - **Group Type** to **h.323** or **sip**.
   - **Priority Video** to **y**.
   - **IP Video** to **y**.
   - Near-end Listen Port
   - **LRQ Required** to **n**.
   - **Far-end Node Name**
   - Far-end Listen Port
   - Far-end Network Region
   - **Calls Share IP Signaling Connection** to **n**.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
5. Use the `add trunk-group` command to add a trunk group for the video trunk. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Carrier Medium** to **IP**.

- Add members to this trunk group.

6. Use the **`change signaling-group`** ***`xx`*** command (where ***`xx`*** is the signaling group you added in Step 3) to set **Trunk Group for Channel Selection** to the trunk group you added in Step 4.
7. Create a route pattern for the trunk group.

Configuring the Maximum Bandwidth for Inter-Network Regions

To configure the maximum bandwidth for inter-network regions:

1. Type **`change ip-network region 1`**.
   The system displays the **IP Network Region** screen.
   Page down till you see the page titled **Inter Network Region Connection Management**.

Figure 153: IP Network Region screen

```
change ip-network-region 1                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total            Video                            Dyn
 rgn rgn  set   WAN  WAN-BW-limits  Norm Prio  Shr Intervening-regions  CAC IGAR
 1   1    1
 1   2    2     y      10:Mbits        2     3  y                              n
 1   3    3     y       5:Mbits        3     2  y                              n
 1   4    4     y       3:Mbits        1     1  y                              n
 1   5    5     y        :NoLimit               y                              n
 1   6    4     y      10:Mbits        1     1  y                              n
 1   7    4     y        :NoLimit               y                              n
 1   8
 1   9
 1   10
 1   11
 1   12
```

2. In the column named **Total**, you can specify the bandwidth across the network regions. In the column named **Video**, you specify how much of the total bandwidth is to be used by video calls. The following are the available options:
   - To support audio only and no video, set the **Video** field to 0 and audio to a very high number as shown in Figure 153.
   - To support audio and video with no bandwidth management, set both the **Total** and **Video** fields to No Limit.
   - To restrict audio bandwidth, and allow unlimited video bandwidth, set the **Total** field to the desired bandwidth. Set the **Video** field to No Limit.

- To control both audio and video bandwidth, set the **Total** field to the total bandwidth available between network regions. Set the **Video** field to the maximum bandwidth that can be used by video. The **Video** field must be set to a value less than or equal to the Total.
- Set priority video to the maximum bandwidth that can be used exclusively by priority video users

3. Figure 154 shows one possible usage scenario for Intervening-regions for network that are not directly connected by WAN. The values that you see in Figure 154 are shown as an example.

Figure 154: Inter Network Region Connection Management screen

```
display ip-network-region 2                                    Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total            Video                           Dyn
 rgn rgn  set   WAN  WAN-BW-limits  Norm Prio  Shr Intervening-regions  CAC IGA
 2   1    2     y      10:Mbits        2     3  y                             n
 2   2    2
 2   3    2     n                                  4:  1:   :                 n
 2   4    4     y      10:Mbits        2     2  y                             n
 2   5    5     y      10:Mbits        0     0  y                             n
 2   6    4     n                                  4:  1:  3:  7              n
 2   7
 2   8
 2   9
 2   10
```

Checking bandwidth usage

To check the status of bandwidth usage:

1. Type **`status ip-network-region`**.
   The system displays the **Inter Network Region Bandwidth Status** screen for a call that is up.

Figure 155: Inter Network Region Bandwidth Status screen

```
status ip-network-region 2/4

                     Inter Network Region Bandwidth Status
                                                Number of    # Times
Src Dst  Conn     Conn  BW-limit    BW-Used(Kbits) Connections BW-Limit  IGAR
Rgn Rgn  Type     Stat                 Tx     Rx      Tx      Rx Hit Today Now/Today

  2   4 direct    pass   10 Mbits       0      0       0       0       0       0/  0
               Video:     2 Mbits       0      0       0       0       0
            Priority:     2 Mbits       0      0       0       0       0
```

2. You can view the audio bandwidth usage on the first row.
   You can view the normal video bandwidth usage on the second row.
   You can view the priority video bandwidth usage on the third row.

Administering Ad-hoc Video Conferencing

Administer the Ad-hoc Video Conferencing feature to allow users to create video conference calls. From a two-party video call, a user can press the **Conference** button on their telephone, dial the number of a third party, and press **Conference** again to add the party to the video conference call. Additional parties, up to a maximum of six, can be added in the same way. If the originator or any party who joins the conference call has administered COS permissions for Ad-hoc Video Conferencing, the video feature is enabled for the call. The call is moved from a Communication Manager hosted audio-only conference to an external bridge multimedia conference.

Administering ad-hoc video conferencing involves the following steps:

1. On page 2 of the **System Parameters Customer-Options (Optional Features) screen**, ensure that the **Maximum Administered Ad-hoc Video Conferencing Ports** field is set to the number of ports available for Ad-hoc Video Conferencing.
2. On the **Class of Service** screen, ensure that **Ad-hoc Video Conferencing** is set to **y** for each class of user with Ad-hoc Video Conferencing privileges. Then assign the COS on the Station screen for the appropriate users.
3. On the **Video Bridge** screen, configure video bridges for Ad-hoc Video Conferencing.

For more detailed information on Ad-hoc Video Conferencing, see *Avaya Video Telephony Solution Networking Guide*, 16-601423.

Multimedia Call Handling

Multimedia Call Handling (MMCH) enables users to control voice, video, and data transmissions using a telephone and PC. Users can conduct video conferences and route calls like a standard voice call. They can also share PC applications to collaborate with others working from remote sites.

Note:

MMCH is Avaya's older technology H.320 video solution. Avaya Video Telephony Solution is Avaya's newer, and preferred H.323 video solution. For more information on AVTS, see Avaya Video Telephony Solution on page 364.

Operations in Basic or Enhanced modes

There are two distinct levels of functionality: Basic and Enhanced. The Basic mode of operation treats a standard-protocol H.320 multimedia call as a data call. If the call is redirected, it is converted to a voice call. As a voice call, certain features are enabled, such as coverage, voice mail, and multiparty video conferencing.

The Enhanced mode of operation allows a multifunction telephone to control a multimedia call as if it were a standard voice call. Spontaneous video conferencing, call forwarding, coverage, hold, transfer and park, along with many routing features, are available to multimedia calls. Both modes of operation allow data collaboration between multiple parties using the T.120 standard protocol.

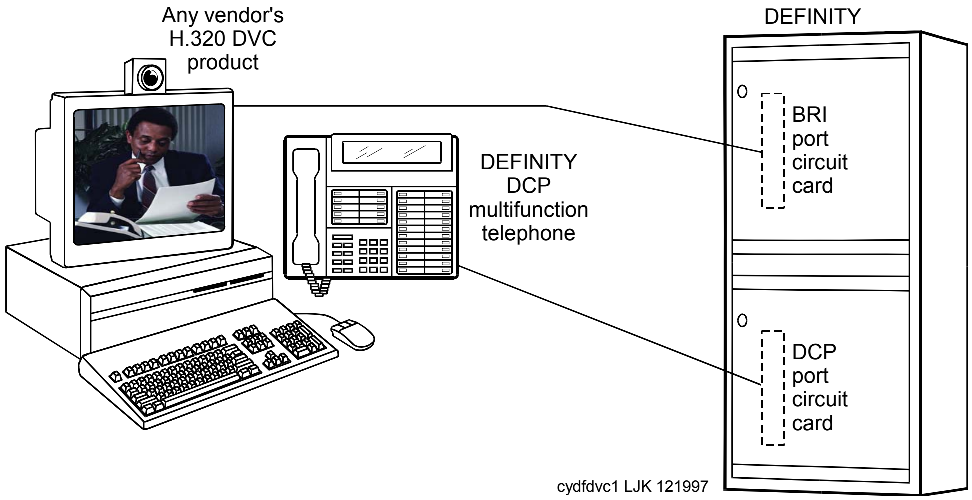
Definitions: MMCH features and components

Multimedia call

A multimedia call, for MMCH, is one that conforms to the H.320 and T.120 suite of protocol standards. These standards allow video-conferencing packages from different vendors to communicate with one another. The capabilities of the individual multimedia-endpoint package can vary, however.

- An H.320 call can contain voice, video and data.
- The bandwidth for MMCH calls is limited to 2 B-channels.

Figure 156: MMCH multimedia complex



Basic multimedia complex

A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Basic mode. With a Basic multimedia complex, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and, if unanswered, will next alert at the voice station. A Basic multimedia complex provides a loose integration of the voice station and H.320 DVC system.

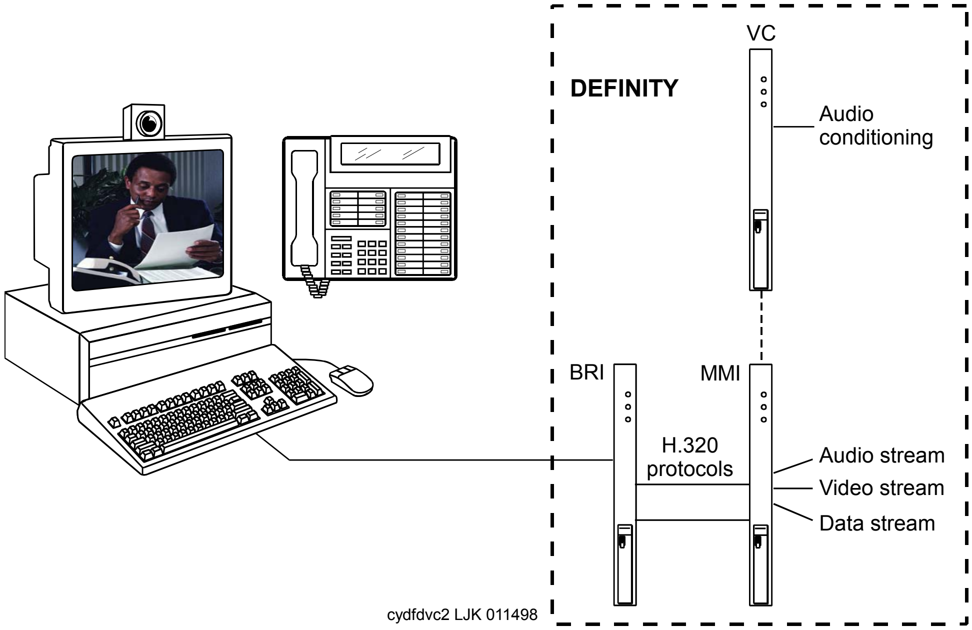
Enhanced multimedia complex

An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Enhanced mode. The Enhanced multimedia complex acts as though the PC were directly connected to the multifunction telephone. Thus, voice call control, multimedia call control and call status are enabled at the telephone. An Enhanced multimedia complex provides a tight integration of the voice station and H.320 DVC system.

Multimedia endpoint

The multimedia endpoint is a user’s PC that has been equipped with an H.320 multimedia package. The PC is physically connected to Avaya Communication Manager with a BRI line.

Figure 157: Enhanced MMCH service link



Enhanced mode service link

The service link is the combined hardware and software multimedia connection between the user's multimedia endpoint and the Avaya DEFINITY Server which terminates the H.320 protocol. The service link provides video, data, and, optionally, voice streams to augment the capabilities of the telephone and PC. A service link only applies to an Enhanced multimedia complex, never to a Basic multimedia complex. The service link is administered on the **Station** screen and can be either "permanent" or "as-needed."

Feature Description

MMCH's two levels of functionality for a multimedia complex, Basic and Enhanced mode, are enabled either by administration on Communication Manager or by an mm-basic feature button or FAC.

Basic Mode Operation

In Basic Mode:

- All voice-only calls originate at the voice station.
- All multimedia calls originate with the H.320 DVC system.

- All incoming voice calls attempt to alert at the voice station and receive all standard voice call treatment.
- All incoming H.320 multimedia calls attempt to alert on the H.320 DVC system initially. If answered, a 2-way video call will result. The Basic multimedia complex voice station will not be involved in the call in any way.

  If the H.320 multimedia call is not answered at the H.320 DVC system and the Basic multimedia complex voice station has the **H.320** field administered to **y**, the call will:
  - Time out at the DVC system.
  - Alert at the associated voice station set as a voice-only call.
  - Receive all standard voice call treatment.
- Call control depends on what type of call is being originated.
  - Video is received and controlled at the PC.
  - Voice is received and controlled at the telephone set.
- The voice station of a Basic multimedia complex must manually add their multimedia endpoint to a multimedia conference. There is limited support for multimedia feature interactions. A specific set of voice features work for multimedia calls.
- Service Links are not used by Basic mode complexes.
- A single number can be used to reach the Basic multimedia complex for voice or H.320 multimedia calls.

Enhanced Mode Operation

The Enhanced multimedia complex provides a much more tightly coupled integration of the complex voice station and H.320 DVC system. In Enhanced Mode:

- Both multimedia and voice calls must originate at the telephone set.
- Voice and multimedia calls can be controlled at the telephone set.
- Conferencing is spontaneous and established just like a voice-only conference call.
- There is extensive support for multimedia feature interaction. Most voice features work the same for multimedia calls.
- Service Links can be either "permanent" or "as-needed."

Physical Installation

The physical components necessary to utilize MMCH capabilities include:

- H.320 DVC systems that are BRI connected to the Avaya DEFINITY Server.
- Non-BRI multifunction telephones.
- Avaya TN787 MultiMedia Interface (MMI) and TN788 Voice Conditioner (VC) boards.
- A T.120 Extended Services Module (ESM) server (necessary only if you plan to do T.120 data collaboration). Connectivity of the ESM requires an additional TN787 along with a TN2207 DS1 circuit pack.

Dual Port Desktop

Both Basic and Enhanced multimedia complexes are dual-port desktops that consist of:

- A BRI-connected multimedia-equipped PC that supports the H.320 protocol.
- A non-BRI-connected multifunction telephone set.

The PC and the multifunction telephone are individually wired to the Avaya DEFINITY Server. These two pieces of equipment can be administratively associated to form a Basic or ENHANCED multimedia complex

MMCH works with any H.320 system that is fully H.320 compliant and operates at the 2B or 128K rate.

Note:

If you intend to share applications among users or whiteboard capabilities, the endpoint software you choose must also support the T.120 protocol.

The following endpoint-software packages have been tested:

- PictureTel PCS 50 & PCS 100, Release 1.6T.
- Proshare 2.0a, 2.1.
- Zydacron Z250 Ver. 2.02, Z350 Ver. 1.2 (With Netmeeting 2.0).

MMI & VC hardware

The MMCH feature requires the use of two additional circuit packs:

- Multi Media Interface (MMI) TN787J.
- Voice Conditioner (VC) TN788B.

The TN787 and TN788 are service circuit packs. The TN787 supports simultaneous operation of 16 2B H.320 calls. The TN788 supports the voice processing for 4 H.320 endpoints.

- These service circuit packs can be located in any Port Network.
- These packs do not require any translations as part of their implementation.
- The MMI and VC circuit packs are resource circuit packs akin to the Tone Detector circuit packs.
- These circuit packs require no administration on Communication Manager and can be located in multiple port networks.

T.120 Data Collaboration Server

The Extended Services Module (ESM) provides T.120 data collaboration capability on a MMCH multipoint H.320 video conference.

- Each person in the conference who wants to participate in the data collaboration session, must have a personal computer with an H.320 video application that supports the T.120 protocol.
- The Avaya DEFINITY Server must have an ESM installed.

ESM Installation

Figure 158: Typical Multimedia Call handling ESM Connections

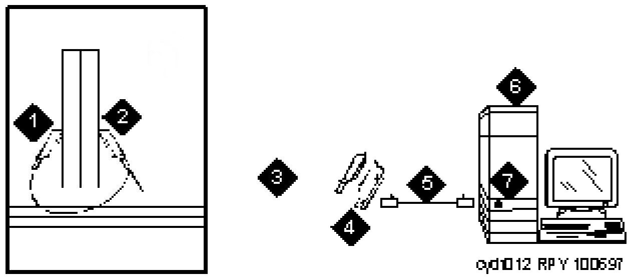


Figure notes:

1. **Port B Y-cable connector to a TN787 multimedia interface (MMI) circuit pack**
2. **Port A Y-cable connector to a TN2207 PRI circuit pack**
3. **25-pair Y-cable**
4. **356A adapter**
5. **D8W cord connected to 356A adapter S/B port 8**
6. **Extended services module (ESM)**
7. **Port B on compatible primary rate interface (PRI) card**

Use the following procedure and Typical Multimedia Call handling ESM Connections on page 381 to connect to the ESM equipment:

1. Install the TN2207 primary rate interface (PRI) circuit pack and the TN787 multimedia interface (MMI) circuit pack in the port carrier of the server for Avaya Communication Manager.

   Note:

   These two circuit packs should be co-located in the cabinet since they must be connected by a Y-cable on the back plane of the Avaya DEFINITY Server.

2. Record the circuit pack locations.
3. Connect the ESM Y-cable as shown.
4. Administer the **DS1 Circuit Pack** screen and the **Signaling Group** screen for the ESM (see ESM T.120 Server Administration on page 391).
5. Configure the ESM adjunct.

Planning for MMCH

The following are some of the tasks you perform in planning and administering MMCH.

Planning the system

Questions to help you use Avaya Communication Manager for multimedia are:

- How many MMCH users are you going to have?
- How many multimedia calls do you expect to have at any given time?

With the information above you can determine how many Voice Conditioner (VC) and Multimedia Interface (MMI) circuit packs you need.

- Will users need data collaboration capabilities? If so, you need to install the Extended Services Module (ESM).
- Which stations, hunt groups or vectors need early answer?
- Do you have ISDN-PRI trunks? It is possible to use separate DS1 trunks for data, but ISDN-PRI trunks are recommended.

Installation checklist

1. Purchase MMCH right-to-use.
2. Avaya — enable MMCH on **System Parameters Customer-Options (Optional Features)** screen.
3. Administer default multimedia outgoing trunk parameter selection on the **Feature-Related System-Parameters Features** screen.
4. Administer MMCH related feature access codes on the **Feature Access Code (FAC)** screen.
5. Install and administer hardware:
   - Install MMIs, VCs and the ESM.
   - Administer the ESM to ECS connection — **DS1 Circuit Pack** and **Signaling Group** screens.
   - Establish maintenance parameters — **Maintenance-Related System Parameters** screen.
6. Administer multimedia complexes:
   - Administer data modules — **Data Module** screen, or **Data Module** page of the **Station** screen.

- Administer stations as part of a multimedia complex, assign associated data module extension, multimedia mode, service link mode and appropriate multimedia buttons — **Station** screen.

7. Administer early answer and H.320 flag for stations, the early answer flag for hunt groups, and the multimedia flag for vectors as appropriate.
8. Train end users.
9. Monitor traffic and performance.

Related screens

- **System Parameters Customer-Options (Optional Features)**
  - Multimedia Call Handling (Basic)
  - Multimedia Call Handling (Enhanced)
- **Feature Related System-Parameters**
  - Default Multimedia Outgoing Trunk Parameter Selection (p.2)
- **Maintenance-Related System Parameters**
  - Packet Bus Activated = **y**
  - Minimum Maintenance Thresholds - MMIs, VCs
- **Data Module** (type = 7500 or WCBRI)
  - Multimedia (p. 1) = **y**
  - XID (p. 2) = **n**
  - MIM Support (p. 2) = **n**
- **Station**
  - MM Complex Data Ext (p. 1)
  - H.320 Conversion (p. 2)
  - Multimedia Early Answer (p. 2)
  - Multimedia Mode (p.2)
  - Service Link Mode (p.2)
  - Feature Buttons (p.3) (optional)
- **Hunt Group**
  - MM Early Answer (optional)
- **Call Vector**
  - Multimedia (optional)

- **Feature Access Code (FAC)**
  - Basic Mode Activation (p.5)
  - Enhanced Mode Activation (p.5)
  - Multimedia Call Access Code (p.5)
  - Multimedia Data Conference Activation & Deactivation (p.5)

  The Multimedia Data Conference Deactivation FAC must be entered after you are active on a multimedia call. To enter the FAC:

  1. Select **Transfer**.
  2. Receive a dialtone.
  3. Dial the FAC.
  4. Receive a confirmation tone.
  5. Re-select the call appearance for the held multimedia call.

  - Multimedia Multi-Address Access Code (p.5).
  - Multimedia Parameter Access Code (p.5).
- **DS1 Circuit Pack** (ESM Only)
  - Bit Rate=**2.048**.
  - Line Coding=**hdb3**.
  - Signaling Mode=**isdn-pri**.
  - Connect=**pbx**.
  - Interface=**network**.
  - Country Protocol=**1**.
  - CRC=**y**.
  - MMI Cabling Board.
- **Signaling group** (ESM Only)
  - Primary D-Channel.

Administering MMCH settings

System Parameters Customer-Options (Optional Features) screen

Ensure that the **Multimedia Call Handling (Basic)** field is **y**. This feature is provided via license file. To enable this feature, contact your Avaya representative.

Feature-Related System Parameters screen

The default bandwidth for MMCH calls is defined on the **Feature-Related System Parameters** screen.

Note:

Originating a multimedia call with the **mm-call** button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen.

- This default parameter will be either 2x56 or 2x64.
- The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Data Module screen

The H.320 DVC system should contain a BRI interface. You must connect this BRI interface to a port on a TN556 BRI circuit pack and administer it as a BRI data module.

- You can administer the data endpoint type as 7500 (recommended) or WCBRI.
- The fields for multimedia are the same on either screen.
- The administration for a Basic mode and an Enhanced mode data module are exactly the same.

Figure 159: Data Module screen (Page 1 of 2)

```
change data-module nn                                          Page   1 of x
                                DATA MODULE

   Data Extension: 30            Name: 27                             BCC:
             Type: data-line___   COS: 1       Remote Loop-Around Test?
             Port: _______        COR: 1         Secondary data module?
              ITC: restricted__    TN: 1                  Connected to: dte


 ABBREVIATED DIALING
   List1:

 SPECIAL DIALING OPTION:



 ASSIGNED MEMBER (Station with a data extension button for this data module)

           Ext     Name
       1: 1002    27 character   station name
```

Figure 160: Data Module screen (Page 2 of 2)

```
change data-module nn                                            Page 2 of x
                              DATA MODULE
        CAPABILITIES
                       KYBD Dialing? y                    Configuration? n
                           Busy Out? n
        SPEEDS
              Low? y           1200? y           4800? y          19200? y
              300? y           2400? y           9600? y     Autoadjust? n
        OPTIONS
                    Permit Mismatch? n                     Dial Echoing? y
                Disconnect Sequence: two-breaks             Answer Text? y
                             Parity: even         Connected Indication? y
```

- **Type** — Set the data module type to 7500 or WCBRI.
- **Multimedia** — This field appears on the **Data Module** screen only if **MM** is set to **y** on the **System-Parameters Customer-Options (Optional Features)** screen. Enter **y** to enable this data module to be multimedia compliant.

- **MM Complex Voice Ext:** (display only) — This field contains the number of the associated telephone in the complex. This is a display-only field, and is blank until you enter the data module extension in the **Station** screen **MM Complex Data Ext** field. Once you have done that, these two extensions are associated as two parts of a multimedia complex.
- **XID and MIM Support** — Valid entries are **y** (default) and **n**. These fields must be set to **n**.

Station screen

After you have administered the BRI data module, use the **Station** screen to associate it with a voice station to screen a multimedia complex. This is a one-to-one relationship: you can administer only one station and one data endpoint per multimedia complex. Neither the voice station, nor the data endpoint can be a member of another multimedia complex.

Note:
A BRI station cannot be part of a multimedia complex.

- **H.320 Conversion** — Valid entries are **y** and **n** (default). This field is optional for non-multimedia complex voice stations and for Basic multimedia complex voice stations. It is mandatory for Enhanced multimedia complex voice stations. Because the system can only handle a limited number of conversion calls, you might need to limit the number of telephones with H.320 conversion. Enhanced multimedia complexes must have this flag set to **y**.

  For non-multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the stand-alone voice station. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the **H.320** field enabled. The **H.320** field is only needed at the *first* station that might receive the H.320 call.

  For Basic multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the Basic multimedia complex voice station *after* an attempt has been made to offer the call to the H.320 DVC system. If the call is unanswered at the H.320 DVC system, the call will alert at the voice station after 5 seconds or after the administered number of rings as specified in the voice station's coverage path. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the **H.320** field enabled. The H.320 field is only needed at the *first* station that might receive the H.320 call.

- **Service Link Mode** - The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and the Avaya DEFINITY Server which terminates the H.320 protocol. A service link is never used by a Basic mode complex H.320 DVC system. Connecting a service link will take several seconds. When the service link is connected, it uses MMI, VC and system timeslot resources. When the service link is disconnected it does not tie up any resources. The Service Link Mode can be administered as either 'as-needed' or 'permanent' as described below:
  - As-Needed - Most non-call center multimedia users will be administered with this service link mode. The as-needed mode provides the Enhanced multimedia complex with a connected service link whenever a multimedia call is answered by the station and for a period of 10 seconds after the last multimedia call on the station has been disconnected. Having the service link stay connected for 10 seconds allows a user to disconnect a multimedia call and then make another multimedia call without having to wait for the service link to disconnect and re-establish.
  - Permanent - Multimedia call center agents and other users who are constantly making or receiving multimedia calls might want to be administered with this service link mode. The permanent mode service link will be connected during the station's first multimedia call and will remain in a connected state until the user disconnects from their PC's multimedia application or the Avaya DEFINITY Server restarts. This provides a multimedia user with a much quicker video cut-through when answering a multimedia call from another permanent mode station or a multimedia call that has been early answered.
- **Multimedia Mode** - There are two multimedia modes, Basic and Enhanced, as described below:
  - Basic - A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone set. When in Basic mode, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and if unanswered will next alert at the voice station if it is administered with H.320 = **y**. A Basic mode complex has limited multimedia feature capability as described in Basic Mode Operation on page 377).
  - Enhanced - An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone. The Enhanced mode station acts as though the PC were directly connected to the multifunction telephone; the service link provides the actual connection between the Avaya DEFINITY Server and the PC. Thus, voice and multimedia calls are originated and received at the telephone set. Voice and multimedia call status are also displayed at the telephone set. An Enhanced mode station allows multimedia calls to take full advantage of most call control features as described in Enhanced Mode Operation on page 379.
- **Multimedia Early Answer** — Valid entries are **y** and **n** (default). This field lets you set this telephone for early answer of multimedia calls. The system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established to the caller, the call will then alert at the voice station.

The station can then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party (see Hourglass Tone on page 398).

Example: An administrative assistant who does not have a multimedia PC, but might get multimedia mode calls from forwarding or coverage, might want to set the H.320 flag to **y** and the early answer flag to **y** on their voice station. This allows any multimedia call to be presented to the station with immediate voice path rather than hourglass tone. The answered call could then be transferred as voice to voice mail or transferred as multimedia to a user equipped with a multimedia endpoint.

Assigning Multimedia Buttons

There are six new multimedia specific buttons that can be added to a voice station. Most of them can be placed on any voice station, whether it is part of a Basic multimedia complex, an Enhanced multimedia complex or not part of any multimedia complex. Two feature buttons, **mm-basic** and **mm-pcaudio**, can only be placed on stations which are part of an Enhanced multimedia complex.

All of the multimedia specific feature buttons have a corresponding feature access code except mm-pcaudio and mm-cfwd.

- The mm-pcaudio feature can only be used via button.
- The **mm-cfwd** button can be replaced by the standard "*call forward*" FAC followed by the "*multimedia call*" FAC.
- **mm-call** -This button can exist on any voice station. Most multimedia enabled users will want an **mm-call** button. This button (or its corresponding FAC) must be used to indicate that the user is placing a multimedia mode call. To place a multimedia mode call the user would go off-hook, select an idle call appearance. Press the **mm-call** button followed by the destination extension digits. If the user has a speakerphone the user can simply press the **mm-call** button, which preselects an idle call appearance, followed by the destination extension digits.

  The **mm-call** button lamp lights when you press this button during call origination. The lamp also lights to indicate that the selected call appearance is a multimedia mode call.
- **mm-basic** - This button is only allowed on the voice station of a multimedia complex. The mm-basic button toggles a station between Basic and Enhanced modes. This button can NOT be used to change the station's multimedia mode when the station has an active multimedia call appearance.

  Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the **Station** screen will show **basic**. When in Enhanced mode this field on the **Station** screen will show **enhanced**. The current station Multimedia mode will be saved to translation when a `save translation` command is executed.

- **mm-pcaudio** - This button only works for an Enhanced multimedia complex voice station. When originating or receiving a multimedia call, the audio path is connected to the voice station's handset or speakerphone device. The **mm-pcaudio** button allows a user to switch the audio portion of any call to their PC's audio input/output device (if available). To switch the audio path to the PC while active on a call, the user presses the **mm-pcaudio** button (if off-hook you can now hang up the handset). The **mm-pcaudio** button's status lamp will light up when the button is pushed to move the audio path to the PC and remain lit while the audio path is at the PC device.

  Note:

  If you are on a voice only call, the voice path will switch to the PC device but you will get muted or loopback video depending on the multimedia endpoint software.

  A user can simply go off-hook on their voice station or press the speakerphone button to move the audio path of a multimedia call from the PC back to the voice station. Pressing the **mm-pcaudio** button while the status lamp is lit and the voice station's handset is on-hook will disconnect the user from the active call.

- **mm-datacnf** - Pressing the **mm-datacnf** button from any voice station that is participating in a multimedia call will light the status lamp and alert the Avaya DEFINITY Server that you want to enable T.120 data collaboration with the other parties on the call. The button status lamp will also light for other participants in the multimedia call who have **mm-datacnf** buttons.

  Pressing this button from the voice station that enabled data collaboration on a multimedia mode call will deactivate the data session and revert to a voice and video call. If you are participating on a multimedia call with data collaboration, but did not initiate the data collaboration, and you press this button, the status lamp led will flash momentarily and the T.120 data services will not be terminated, (only the station that activated the collaboration session can deactivate it). This button only works for stations connected to an Avaya DEFINITY Server equipped with an ESM adjunct.

- **mm-cfwd** - The **mm-cfwd** button allows a user to indicate that multimedia mode calls will be forwarded as multimedia mode calls to a specific forwarded-to destination. If voice call forwarding is active and multimedia call forwarding is not active then multimedia calls going off of the Avaya DEFINITY Server will be forwarded as voice only calls.

  The **mm-cfwd** button status lamp will be lit to indicate that multimedia call forwarding is activated. Pressing the **mm-cfwd** button when the lamp is lit will deactivate multimedia call forwarding.

  Note - pressing the **mm-cfwd** button is the same as dialing the regular call-fwd FAC followed by the mm-call button or FAC followed by the desired forwarded-to extension digits.

- **mm-multinbr** - The **mm-multinbr** call button is similar to the **mm-call** button. It allows origination of a multimedia call from any voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of address must be entered.

  Originating a multimedia call with the **mm-multinbr** button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

ESM T.120 Server Administration

Figure 161: DS1 Circuit Pack screen

```
add ds1 nnnn                                                      Page 1 of x
                                 DS1 CIRCUIT PACK

             Location: _____                           Name: _______________
             Bit Rate: _____                    Line Coding: ____
    Line Compensation: _                       Framing Mode: ___
       Signaling Mode: ___________                D-Channel: __
              Connect: _________                  Interface: ___________
         Interconnect: ___                    Peer Protocol: ________
                                           Country Protocol: ____
                                           Protocol Version: _
Interface Companding: _____                            CRC? _
            Idle Code: ________
                                DCP/Analog Bearer Capability: ________
                                           T303  Timer(sec):

    MMI Cabling Board: _____  MMI Interface: ESM

MAINTENANCE PARAMETERS

       Slip Detection? _                  Near-end CSU Type: ________
                                    Block Progress Indicator? n
```

From the system administration terminal:

1. Type **`list configuration all`**, and a list of the installed carriers, circuit packs, and ports appears.
2. Record the location (board number) of the MMI board cabled to the TN2207 slot and verify that all other required circuit packs are present.
3. Enter **`add DS1`** ***`xxxxx`***, (where ***`xxxxx`*** is the location of the TN2207 PRI circuit pack recorded in step 2), and the **DS1 Circuit Pack** screen appears.

4. Set the **Name** field to **ESM DS1**.
5. Set the **Bit Rate** field to **2.048**.

   The TN2207 DS1 must have a bit rate of 2.048, even if all other DS1 boards in the system are operating at 1.544. Verify the 24/32 channel switch on the circuit pack is in the 32 channel position.
6. Set the **Line Coding** field to **hdb3**.
7. Set the **Signaling Mode** field to **isdn-pri**.
8. Set the **Connect** field to **pbx**.
9. Set the **Interface** field to **network**.
10. Set the **Country Protocol** `field` to **1**.
11. Set the **CRC** field to **y**.
12. The **Idle Code** default is **11111111**.
13. The **DCP/Analog Bearer Capability** default is **3.1 kHz**.
14. Set the **MMI Cabling Board** field to ***xxxxx*** (where ***xxxxx*** is the location of the TN787 MMI circuit pack recorded in step 2). This must be the slot for port B of the Y-cable.
15. The **MMI Interface** field **ESM** appears.
16. Enter **`add signaling-group next`**. The **Signaling Group** screen appears.

Figure 162: Signaling Group screen

```
add signaling-group n                                       Page 1 of x
                              SIGNALING GROUP
Group Number  ___            Group Type:  isdn-pri
                  Associated Signaling?        Max Number of NCA TSC: ___
                     Primary D-Channel:         Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___ X-Mobility/Wireless Type: NONE
    TSC Supplementary Service Protocol: _
```

17. Set the **Associated Signaling** field to **y**.
18. Set the **Primary D-Channel Port** field to ***xxxx*16** (where ***xxxx*** is the address of the TN2207 PRI circuit pack, for example: 1B0516).
19. The **Max Number of NCA TSC** default is **0**.
20. The **Max Number of CA TSC** default is **0**.
21. **Trunk Group for NCA TSC** ____ (leave blank).
22. **Trunk Group for Channel Selection**____ (leave blank).
23. Logoff the terminal and then log back on the terminal to view your changes.

Troubleshooting

To determine ESM link status, enter the following commands from the system administration terminal:

1. **`Status esm`**
2. **`Status signaling-group`**
3. **`List MMI`**

Note:

When you move ESM circuit packs, you MUST remove the DS1 and signaling group translations. You cannot use the **`change circuit pack`** command.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector *must* have the **Multimedia** field set to **n**. This field causes multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent and starts network billing for the incoming call when vector processing begins.

Understanding the Multimedia Complex

1-number access

1-number access permits originating users to make voice or multimedia calls to a Basic multimedia complex by dialing the same number for either type of call. The number might be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya Communication Manager directs it to the telephone. If the incoming call is 56K or 64K data call, Avaya Communication Manager recognizes it as such and sends it to the multimedia endpoint. Likewise, if a voice call is addressed to the data extension, the system recognizes this and directs the call to the voice station.

Calls originating on the same server as the Basic mode complex destination can always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate data non-PRI digital facilities multimedia calls must be made to the data extension.

AVD (alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

Originating voice calls

All voice calls are originated at the voice station.

Originating multimedia calls

For a Basic mode complex, multimedia calls are normally originated at the user's multimedia equipped PC. These multimedia calls use the associated station's COR/COS.

The voice station of a Basic multimedia complex can also use the **mm-call** button or FAC, and the **mm-multinbr** button or FAC to originate multimedia calls. When these methods are used, a multimedia call is originated from the voice station. In order for the Basic multimedia complex to receive video, the user must make a call from the H.320 DVC system to the voice station of the complex or must make a multimedia call from the voice station to the H.320 DVC. This allows the station to spontaneously add themselves or other parties to a multimedia conference.

1. **H.320 DVC system GUI**. The normal way for a Basic multimedia complex endpoint to originate a multimedia call is through the vendor provided user interface. Generally, digits to dial are entered, speed is selected and the call originates from the DVC system. The voice station is not involved in such as origination.

   Any voice station can use the following mechanisms to originate a multimedia call from the voice station. For stations that are not part of a multimedia complex, video cannot be provided. For voice stations that are part of a Basic multimedia complex, video is not provided until a multimedia call is made from the complex's H.320 DVC system to the voice station or a multimedia call is made from the voice station to the H.320 DVC system. Video is automatically included for Enhanced multimedia complexes.

2. **mm-call (Multimedia Call) button**. If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user can select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user can originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green LED) should become solid.

   The user now dials the destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the **mm-multinbr** button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

Note:

The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a multimedia call.

3. **Multimedia Call feature access code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code can be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

   Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the **mm-multinbr** button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

   Note:

   The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a multimedia call.

4. **mm-multinbr (Multimedia Multi-number) button**. The **mm-multinbr** button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the **mm-multinbr** button. The user can select the **mm-multinbr button** and then go off-hook. If the user has a speakerphone on the station, the user can originate the call simply by selecting the **mm-multinbr** button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the **mm-multinbr** button, the **mm-multinbr** and **mm-call** (if present) status lamp (green led) should light steadily. The user now dials the first destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the 2nd address has been collected the **mm-multinbr** status lamp will go off.

   Originating a multimedia call with the **mm-multinbr** button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Note:

The **mm-multinbr** feature button is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

5. **Multimedia Multi-number Call feature access code**. For stations that do not have an administered **mm-multinbr** button, the Multimedia Multi-number call feature access code can be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the **mm-call** (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   Note:

   The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

6. **Multimedia parameter selection feature access code**. This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

   The MM parameter selection FAC can be used in conjunction with the **mm-multinbr** button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user can indicate a dual-address call at this point with the **mm-multinbr** button or FAC. The user now dials one or two sets of destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

Note:

The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

7. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLs
   1. Single address with TAC
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** TAC, **Dial** destination digits
   2. Dual address with TAC
      - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
      - ***Dial*** TAC, **Dial** 1st dest. digits, ***Hear*** dialtone
      - ***Dial*** TAC, **Dial** 2nd dest. digits
   3. Single address with AAR/ARS
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** AAR/ARS, ***Dial*** destination digits
   4. Dual address with AAR/ARS
      - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
      - ***Dial*** AAR/ARS, **Dial** 1st dest. digits, ***Hear*** dialtone
      - ***Dial*** AAR/ARS, **Dial** 2nd dest. digits
   5. Single address with AAR/ARS and authorization code
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** AAR/ARS FAC, **Dial** destination digits, ***Hear*** stutter dialtone
      - ***Dial*** authorization code
   6. Dual address with AAR/ARS and authorization code
      - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
      - ***Dial*** AAR/ARS FAC, **Dial** 1st dest. digits, ***Hear*** dialtone
      - ***Dial*** AAR/ARS FAC, **Dial** 2nd dest. digits, ***Hear*** stutter dialtone
      - ***Dial*** authorization code
   7. Single address with TAC or AAR/ARS and CDR account code
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** CDR FAC, ***Hear*** dialtone
      - ***Dial*** CDR account code, ***Hear*** dialtone
      - ***Dial*** TAC or AAR/ARS, ***Hear*** destination digits

8. Dual address with TAC or AAR/ARS and CDR account code
   - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
   - ***Dial*** CDR FAC, ***Hear*** dialtone
   - ***Dial*** CDR account code, ***Hear*** dialtone
   - ***Dial*** TAC or AAR/ARS, ***Dial*** 1st dest. digits
   - ***Dial*** TAC or AAR/ARS, ***Dial*** 2nd dest. digits

Receiving voice calls

Any voice calls directed to the voice or data extension of a Basic multimedia complex will ring at the voice station.

Receiving multimedia calls

Any data calls directed to the voice or data extension of a Basic multimedia complex will ring at the multimedia equipped PC if it is available. You can answer the multimedia call at the PC and voice and video will connect to the PC. If the data endpoint is unavailable, the system verifies that the telephone of the complex is administered with the **H.320** field set to **y**. If so, the system converts the call to voice and sends it to the telephone of the multimedia complex, where the call then alerts.

Hourglass Tone

When a voice station answers a converted multimedia call, the answering party might hear different things depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer

The answering party can administer their station to avoid hearing hourglass tone. With the **Station** screen **Early Answer** field set to **y**, the system answers the incoming multimedia call on behalf of the station and establishes the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Basic complex destination. The station can then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

If the **H.320** field is not set to **y** for the telephone of a Basic multimedia complex, H.320 calls alert at the multimedia endpoint until the caller drops. If an H.320 call is directed to a telephone with H.320 set to **n**, the system denies the call.

You can assign H.320 conversion to any voice station.

Authorization

Multimedia complexes require the same types of authorization (COR/COS) as standard telephones. If a call is addressed to the voice extension, the system checks the COR/COS of the telephone, whether the call is voice-only or multimedia. If a call is addressed to the data extension, the system checks the COR/COS of the data endpoint. If the call is subsequently redirected to the voice station, the system does a second COR/COS check for the authorization of the voice station. Calls originated from the PC use the COR/COS of the voice station.

Adjunct Switch Applications Interface

ASAI is not expected to support call-association for data calls. Therefore Avaya does not recommend that you use ASAI for multimedia.

Administered Connections

Basic Multimedia endpoints can serve as the origination point or destination of an administered connection.

Authorization and Barrier Codes

Basic Mode multimedia users or off-premises PC users might not be able to respond to prompts for authorization or barrier codes. Multimedia endpoints do not recognize the prompts.

An on-premises user might be able to use Remote Access and enter the entire digit string at once before launching the call, but it would be better to eliminate the need for such codes for multimedia users who need to call off premises.

Bridged Appearances

Voice users can bridge onto a call if the user has a bridged appearance of a voice member of the call.

Call redirection

Calls directed to either member of the Basic multimedia complex are subject to redirection (coverage, forwarding). Avaya Communication Manager converts calls to voice before sending them to coverage. Calls redirected through call forwarding maintain multimedia status if forwarded from the data endpoint.

Conferencing

A multimedia conference can consist of multimedia and voice-only conferees. All multimedia conferees are added to a multimedia conference by a voice-terminal user on Communication Manager, who acts as the controller of the multimedia conference. When the controller is a Basic complex voice station, the controller must remain on the conference until all parties have joined. Once all endpoints are on the conference, the voice-terminal user can put the call on hold or drop, if the user wishes.

Video conferees can see only their local video and one other party. If more than two people are involved in a video conference, the person who is speaking is the one whose video appears to other conferees. The speaker's video shows the previous speaker. This changes dynamically as the speaker changes.

Creating a multi-party video conference

All multimedia conferences must be controlled by a voice telephone. Multimedia conferees can be added by calling the voice telephone or by having the voice telephone make a multimedia call to other DVC endpoints. The controller can then conference together individual parties to create a multimedia conference.

To set up a multimedia conference:

1. Determine who is going to be the conference controller.

2. At the appointed time, the conference controller calls his or her telephone from the multimedia endpoint by dialing the 1-number extension. Once this call is established, the controller conferences in other calls as if this were a voice conference. The controller continues to add conferees in this manner until all conferees have joined, or until the number of conferees reaches the administered limit.
3. The conference controller can also add voice or multimedia parties to the conference spontaneously. The controller presses **CONFERENCE**, makes a voice or multimedia call to a new party. To make a multimedia call, the controller must originate a call using the mm-call button or FAC or the **mm-multinbr** button or FAC. After the new party begins alerting, the controller can press **CONFERENCE** to add the party to the existing conference call on hold.

Coverage

Multimedia calls to a Basic mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice station of the Basic multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the **Station** screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the **Station** screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Coverage: Multimedia calls and off-net call coverage

If the principal station’s coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Coverage: Multimedia calls and coverage to voice mail

Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the **Early Answer** field on the hunt group set to **y**. This field will have no effect on voice calls to the voice mail system.

Call Detail Recording

Each channel of a 2-channel call generates a separate CDR record.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration can be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Extended Services Module (ESM) server, which is an adjunct to Avaya Communication Manager. Up to six parties can participate in a single data conference, and up to 24 parties can use the ESM facilities for data collaboration at any given time.

Adding data sharing to a video conference

1. Set up a multimedia conference.
2. Once a multimedia call is active, any voice station in the conference, can initiate data collaboration by pressing the **mm-datacnf** button. Or, to use the feature access code to initiate a data conference, press the Transfer button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. Avaya Communication Manager will select a data rate which is acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM server resources available for all parties currently in the call, the activation of T.120 data sharing will be denied. The **mm-datacnf** status lamp will flash denial or the **mm-datacnf** FAC will produce reorder.
3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI can prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.
4. The user must now use the PC’s GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users can control the shared application, see the vendor provided documentation for your specific H.320 DVC system.

6. To end the data collaboration session and retain the voice/video conference, the station that selected the **mm-datacnf** button or FAC can press the **mm-datacnf** button or hit transfer and dial the **mm-datacnf** deactivation FAC.

   Note:

   As of this writing, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. Avaya recommends that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled.

If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM server resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the multi-layer protocol (MLP) data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing MLP data rate, the new endpoint will only receive voice and video.

Single server or switch data collaboration.

When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server or Avaya S8XXX Server, there is no restriction on the type of user. The parties can be any combination of Enhanced multimedia complexes, Basic multimedia complexes, or stand-alone H.320 DVC systems.

Multi-switch data collaboration.

When all parties involved in data collaboration conference are *not* located on the same physical Avaya S8XXX Server, the parties located on the Avaya server hosting the data conference (i.e. the server which activated **mm-datacnf**) can be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems. *All parties on remote servers must not be Enhanced multimedia complexes:* they must be Basic multimedia complexes or stand-alone H.320 DVC systems. Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Forwarding voice/multimedia calls

In Basic mode you can forward calls from either the telephone or the multimedia endpoint. To forward a call from the multimedia endpoint:

1. At the PC's multimedia application, enter the call-forwarding feature access code (FAC)
2. Enter the forward-to number in the **Dialed Number** field on the endpoint software
3. Click the **Dial** button (or equivalent)

Note:

The PC multimedia software will probably respond with a message that the call failed, since it does not recognize the FAC. In fact, Avaya Communication Manager *does receive* the message, and forwards all multimedia calls addressed to the 1-number.

If a call is forwarded from the telephone, the call converts to voice first. If using the multimedia endpoint to forward, the calls arrive at the forwarded-to extension as a data call. Such calls continue to ring until answered or abandoned, rather than follow a coverage path.

Users can forward calls from the multimedia endpoint using the call forward FAC. You can also assign a call-forward button at the voice station to forward calls for the data endpoint. If a Basic multimedia complex has console permissions, that user can forward calls for others by dialing the FAC, the data extension, and then the forwarded-to number.

Call Park

A voice-terminal user can park any active call, voice or multimedia, and unpark the call from another telephone. Users cannot park or unpark calls using multimedia endpoints.

Call Pickup

Members of a pickup group can answer an H.320 call using a telephone after the call has been converted to voice. This is true for standard or directed call pickup.

Consult

After a call is converted to voice, consult can be used when transferring or conferencing the call.

COR/COS

The Class of Restriction and Class of Service for H.320 calls originated from a 1-number complex are the same as those of the telephone in the complex.

Data Call Setup

Basic complex multimedia endpoints are BRI data endpoints, and can use data call-setup procedures as provided by the software vendor.

Data Hotline

If endpoint software allows users to select the dial function without entering a number, the endpoint can be used for hotline dialing.

Dial Access to Attendant

Access to Attendant is blocked for a data call from a Basic mode multimedia endpoint.

Data Trunk Groups

Data trunk groups can be used to carry H.320 calls of a fixed (administered) bearer capability.

Hold

The voice station and multimedia endpoint of a Basic complex are each independent devices with respect to call control. When a Basic multimedia complex voice station executes hold only the voice station is held. If the user has conferenced their multimedia endpoint into a multimedia conference, activating hold will *not* disconnect the multimedia endpoint from the conference, it will only disconnect the Basic multimedia complex voice station. Executing hold with an Enhanced mode complex will fully disconnect voice and video from the current active call.

Hunt Groups using Basic Mode complexes

Since Basic mode complexes can receive point to point multimedia calls at the DVC system and voice calls to the station simultaneously, the voice station extension can be placed in any normal voice hunt group or ACD skill and the data extension can be placed in a simple hunt group made up of only data extensions.

Basic mode complex data extensions or stand-alone data extensions can be used to create simple data hunt groups. Data extensions are not allowed in ACD hunt groups. Avaya recommends that you do not mix voice and data stations in a hunt group.

If you want multimedia calls to hunt to multimedia endpoints (i.e. 2B point to point data hunting), put the data extension in the hunt group. If you place the voice extension in a hunt group, only voice calls hunt to that extension. Multimedia calls to a hunt group with a Basic mode voice station as the hunt group member will *not* be offered to the DVC system of the Basic mode complex. If either the voice or data extension of a Basic mode complex is busy, the entire complex is considered busy for hunting purposes.

In order to guarantee that all members of a voice hunt group or skill can receive voice or multimedia calls, all members should have the **H.320** field on the **Station** screen set to **y**. Simple voice stations and Basic complex mode voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The **MM Early Answer** field (on the **Hunt Group** screen) tells the system to answer the incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately. This is not necessary for hunt groups comprised of data extensions.

Hunting, Other considerations

Agents that are part of a Basic mode complex can dial a feature access code to remove themselves from availability (and to indicate that they are available again) from both the multimedia endpoint and the telephone independently. This allows the voice member or the data member to be individually made unavailable. To make the data extension unavailable, the agent must dial the FAC from the DVC system.

CMS measurements can indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Hunting Call association (routing)

Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya Communication Manager attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member. In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group must have ANI information. The ANI information can be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same Avaya S8XXX Server as the hunt group are easily associated. If multimedia calls into a hunt group have incorrect ANI information (i.e. all calls from server X to server Y include the LDN for server X), then as the volume of calls increases, the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, Communication Manager will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Hunting with Multimedia vectors

Calls are often routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls that will terminate to voice stations, you must set the **Multimedia** field on the **Call Vector** screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

Note:

Vectors which have the **Multimedia** field set to **y** must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the **Multimedia** field set to **y** should never be set up to route to a hunt group or number which is a data extension.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector *must* have the **Multimedia** field set to **n**.

Intercept Treatment

H.320 calls that receive intercept treatment are treated like other data calls. H.320 calls cannot be directed to an attendant for service because the attendant cannot have H.320 conversion service.

ISDN Trunk Groups

Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis which can be used to distinguish voice calls from multimedia calls.

Malicious Call Trace

If a malicious call terminates at a Basic multimedia complex endpoint, the user can dial the feature access code from the telephone to activate malicious call trace, followed by the extension of the multimedia endpoint. If the user does not dial the multimedia extension, MCT traces any call held on the telephone.

Message Waiting

Message Waiting indication is handled at the telephone. Because H.320 calls are converted to voice before going to coverage, all messages are voice only.

Night Service

Incoming Basic mode data calls follow established night-service processing for data calls.

Remote Access

Communication Manager does not prevent Basic multimedia complexes from attempting to use remote access. However, these Basic mode endpoints will most likely not be able to dial the necessary codes.

Station Hunting

Basic mode data calls to endpoints that have an extension administered in the **Hunt-to-station** field hunt based on established hunting criteria. The call is converted to voice before station hunting.

Tenant Partitioning

Permission to make multimedia calls or add parties of any type to a conference is subject to standard tenant-partitioning restrictions.

Terminating Extension Groups

Basic mode data calls to a TEG are converted to voice and can terminate only at a voice endpoint. Effectively, Communication Manager treats the multimedia-complex extension as a voice-only endpoint.

Telephone Display

Display information for calls to or from a Basic multimedia complex contains the 1-number.

Enhanced Mode MM Complex

The Enhanced multimedia complex provides a much greater unified and integrated interface for control of voice and multimedia calls. The multifunction voice station is used to control all calls, whether voice or multimedia. The H.320 desktop video system is used to present the video stream, data stream and (optionally) audio stream to the user. The H.320 desktop video system is not used for call control. The Enhanced multimedia complex allows the multifunction voice station to handle voice or multimedia calls in an almost identical manner. Each call appearance on the voice station can represent a voice or multimedia call, allowing multiple voice or multimedia calls to be present simultaneously on the station. The user can manage the separate call appearances without regard to the voice or multimedia nature of the specific call. The standard HOLD/TRANSFER/CONFERENCE/DROP actions can be applied to any call, without regard to the voice or multimedia nature of the call.

1-number access

1-number access permits originating users to make voice or multimedia calls to an Enhanced multimedia complex by dialing the same number for either type of call. The number can be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya Communication Manager alerts the station of an incoming voice call. If the incoming call is 56K or 64K data call, Communication Manager recognizes it as a multimedia call, inserts resources to terminate the H.320 protocol, and alerts the voice station with a multimedia call.

Calls originating on the same Avaya S8XXX Server as the Enhanced mode complex destination can always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate non-PRI digital facilities for data, multimedia calls must be made to the data extension of the Enhanced mode complex.

AVD (alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

Originating Multimedia calls

The basic call sequence from an Enhanced mode complex is to originate a multimedia call and alert the destination. When the destination answers the call, the originating station's H.320 desktop video system will be alerted (that is, called by Communication Manager to establish the service link). If the H.320 DVC is not configured for auto-answer, the user must answer the H.320 calls via the DVC GUI. If the H.320 DVC is configured for auto-answer, no action is needed via the DVC GUI.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

If the far-end is providing a video signal, the 2-way video will be observed. If the destination is not providing a video signal (call was answered by a simple voice telephone), then loopback video will be provided at the Enhanced mode complex originator. The audio signal will exist at the handset of the voice telephone. The audio signal can be moved to the H.320 DVC system via activation of a **mm-pcaudio** button on the voice telephone.

Hourglass tone

The originating party might hear different things when the incoming multimedia call is answered depending on the nature of the answering party. If the call is being answered directly by an H.320 DVC system or if the answering party is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the originating party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio path now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the originating party that they should wait for the H.320 call to establish audio.

Originating voice calls

Voice calls are originated from the voice station of an Enhanced mode complex in the normal manner as for any voice station.

Originating multimedia calls

Multimedia calls from an Enhanced multimedia complex are originated from the VOICE STATION, NOT the H.320 desktop video system. All multimedia originations require the user to indicate the multimedia nature of the call prior to providing any address digits. There are several different ways to originate a multimedia call from the voice station.

1. **mm-call (Multimedia Call) button**. If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user can select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user can originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green LED) will light steadily, indicating a multimedia call.

   The user now dials the destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the **mm-multinbr** button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

   Note:

   The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a multimedia call.

2. **Multimedia Call feature access code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code can be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

   Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. The section below on the **mm-multinbr** button/FAC provides information on originating a 2B call where the destination has a different address for each B-channel.

Note:

The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a multimedia call.

3. **mm-multinbr (Multimedia Multi-number) button**. The **mm-multinbr** button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the **mm-multinbr** button. The user can select the **mm-multinbr** button and then go off-hook. If the user has a speakerphone on the station, the user can originate the call simply by selecting the **mm-multinbr** button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the **mm-multinbr** button, the **mm-multinbr** and **mm-call** (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the second address has been collected, the **mm-multinbr** status lamp will go off.

   Originating a multimedia call with the **mm-multinbr** button will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   Note:

   The **mm-multinbr** feature button is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

4. **Multimedia Multi-number Call feature access code**. For stations that do not have an administered **mm-multinbr** button, the Multimedia Multi-number call feature access code can be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the **mm-call** (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Note:

The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

5. **Multimedia parameter selection feature access code**. This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

   The MM parameter selection FAC can be used in conjunction with the **mm-multinbr** button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user can indicate a dual-address call at this point with the **mm-multinbr** button or FAC. The user now dials one or two sets of destination address digits. The destination address can be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Note:

   The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but can be used by any station to originate a dual address multimedia call.

6. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLs
   1. Single address with TAC
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** TAC, ***Dial*** destination digits
   2. Dual address with TAC
      - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
      - ***Dial*** TAC, ***Dial*** 1st dest. digits, ***Hear*** dialtone
      - ***Dial*** TAC, ***Dial*** 2nd dest. digits
   3. Single address with AAR/ARS
      - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
      - ***Dial*** AAR/ARS, ***Dial*** destination digits

4. Dual address with AAR/ARS
   - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
   - ***Dial*** AAR/ARS, ***Dial*** 1st dest. digits, ***Hear*** dialtone
   - ***Dial*** AAR/ARS, ***Dial*** 2nd dest. digits
5. Single address with AAR/ARS and authorization code
   - ***Dial*** mm-call button or FAC, ***Hear*** dialtone
   - ***Dial*** AAR/ARS FAC, ***Dial*** destination digits, ***Hear*** stutter dialtone
   - ***Dial*** authorization code
6. Dual address with AAR/ARS and authorization code
   - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
   - ***Dial*** AAR/ARS, ***Dial*** 1st dest. digits, ***Hear*** dialtone
   - ***Dial*** AAR/ARS, ***Dial*** 2nd dest. digits, ***Hear*** stutter dialtone
   - ***Dial*** authorization code
7. Single address with TAC or AAR/ARS and CDR account code
   - ***Dial*** **mm-call** button or FAC, ***Hear*** dialtone
   - ***Dial*** CDR FAC, ***Hear*** dialtone
   - ***Dial*** CDR account code, ***Hear*** dialtone
   - ***Dial*** TAC or AAR/ARS, ***Dial*** destination digits
8. Dual address with TAC or AAR/ARS and CDR account code
   - ***Dial*** **mm-multinbr** button or FAC, ***Hear*** dialtone
   - ***Dial*** CDR FAC, ***Hear*** dialtone
   - ***Dial*** CDR account code, ***Hear*** dialtone
   - ***Dial*** TAC or AAR/ARS, ***Dial*** 1st dest. digits
   - ***Dial*** TAC or AAR/ARS, ***Dial*** 2nd dest. digits

Answering multimedia calls

The user actions required to answer voice or multimedia calls at an Enhanced multimedia complex are identical if the H.320 DVC system is configured for auto-answer. If the H.320 DVC system is not configured for auto-answer an additional step is required. See Answering multimedia calls below.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

Answering voice calls

Incoming voice calls will alert at the voice station of the Enhanced multimedia complex in the normal manner. Standard alerting and call appearance flashing will occur. They are answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station.

Answering multimedia calls

Incoming multimedia calls will alert at the voice station of the Enhanced multimedia complex in the same manner as voice calls with one addition. If the alerting station has an administered mm-call button and the alerting call appearance is the selected call appearance (for instance, the red LED is lit, on the alerting call appearance), then the mm-call button status lamp will go on indicating that the call on the selected call appearance is a multimedia call.

The incoming multimedia call is answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station. If the H.320 DVC system for the answering party is configured for auto-answer, no other action is needed to complete the multimedia call. If the H.320 DVC system for the answering party is not configured for auto-answer, the H.320 DVC system will alert and must also be answered by the user.

Note:

Avaya recommends, but does not require, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.

If the originating party is providing a video signal, then a complete 2-way multimedia call will exist. If the originating party is not providing a video signal, the answering party will receive loopback video. The audio signal will exist at the handset of the voice station. The audio signal can be moved to the H.320 DVC system via activation of a **mm-pcaudio** button on the voice station.

Hourglass Tone

The answering party might hear different things when the incoming multimedia call is answered depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote server, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short "incoming call tone" indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as "hourglass tone." Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer

The answering party can administer their station in such a way as to avoid hearing hourglass tone. If the **Station** screen has set the **Early Answer** field to **y**, then the system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Enhanced mode complex destination. The station can then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

Multiple call appearance operation

With an Enhanced mode complex all calls to or from the complex are controlled via the voice station. Each voice or multimedia call has its own call appearance which can be selected without regard for the nature of the call using the specific call appearance. This allows a multifunction station to control multiple voice or multimedia calls in exactly the same way they would control multiple voice calls.

As an example, a user can originate a simple voice call on the first call appearance. A multimedia call can then arrive on the second call appearance. The user activates **HOLD** on the first call appearance and selects the second call appearance to answer the multimedia call. The user can then activate **HOLD** on the second call appearance and reselect the first call appearance or select a third call appearance and originate another call.

Creating a multi-party video conference

An Enhanced multimedia complex can create a spontaneous video conference in the same way that a spontaneous voice conference is created. Given an active call, the user activates the **CONFERENCE** button. This puts the current call on **HOLD** and activates a new call appearance. The user makes a multimedia call according to the instructions for originating a multimedia call and then selects **CONFERENCE** to combine or merge the two call appearances. This results in a 3-way conference.

If all three parties are video equipped, then a 3-way video conference results. Conference members see the current speaker on video. The current speaker sees the last speaker on video. If one of the parties is not video equipped, then a 3-way audio conference exists and the two video equipped parties have 2-way video. The **CONFERENCE** action can be repeated until 6 parties have been conferenced together. The 6 parties can be any mix of voice or video, local or remote parties.

The following steps create a multi-party voice/video conference:

1. Enhanced mode complex station A originates a multimedia call to, or receives a multimedia call from, party B. Station A and party B have 2-way voice and video.
2. Station A, activates **CONFERENCE**.
3. Station A originates a multimedia call (i.e. uses the mm-call button/FAC/etc.) and dials the party to be added, Enhanced multimedia complex C.

4. Party C, answers the call from station A.
5. Station A selects **CONFERENCE** to complete the 3-way conference. Parties A,B and C will be in a 3-way voice/video conference.

   Note:

   If party C is another Enhanced mode complex on the same Communication Manager server as station A, station A does not need to indicate a multimedia call prior to dialing the new party in step 3. While A consults with C, the call will be audio only. When A completes the conference in step 5, party C's video will be activated.

A multi-party video conference uses voice-activated switching to determine which parties are seen. The current speaker is seen by all other parties. The current speaker sees the previous speaker.

Additional voice or video parties can be added by repeating these steps.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration can be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Extended Services Module (ESM) server, which is an adjunct to the Avaya DEFINITY Server. Up to six parties can participate in a single data conference, and up to 24 parties can use ESM facilities for data collaboration at any given time.

The following steps add data sharing to a video conference:

1. Set up a multimedia conference.
2. Once a multimedia call is active, any member can initiate data collaboration by pressing the **mm-datacnf** button. Or, to use the feature access code to initiate a data conference, press the **Transfer** button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. Avaya Communication Manager will select an MLP data rate acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM server resources available for all parties currently in the call, activation of T.120 data sharing will be denied. The **mm-datacnf** status lamp will flash denial or the mm-datacnf FAC will produce reorder.
3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI might prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.

4. The user must now use the PC’s GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users can control the shared application, see the vendor provided documentation for your specific H.320 DVC system.
6. To end the data collaboration session and retain the voice/video conference, the station that selected the **mm-datacnf** button or FAC can press the **mm-datacnf** button or press **Transfer** and dial the mm-datacnf deactivation FAC.

   Note:

   Currently, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. Avaya recommends that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled.

If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM server resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing data rate, the new endpoint will only receive voice and video.

Activating HOLD while on a T.120 data collaboration conference.

If an Enhanced multimedia complex is active on a multimedia call and the call has activated T.120 data collaboration, the user should be receiving voice/video and data. If the station places this existing call on hold, audio and video will be disconnected for the current call. The data collaboration portion of the call will remain intact and unaffected. While this T.120 data conference is on hold, the user will only be allowed to receive audio on all other call appearances. Thus a user is limited to one call appearance that has T.120 data collaboration active.

Single server or switch data collaboration.

When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server or Avaya S8XXX Server, there is no restriction on the type of user. The parties can be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Multi-switch data collaboration

When all parties involved in data collaboration conference are *not* located on the same physical Avaya S8XXX Server, the parties located on the Avaya server hosting the data conference (i.e. the server that activated mm-datacnf) can be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Note:

All parties on remote servers must not be Enhanced multimedia complexes. They must be Basic multimedia complexes or stand-alone H.320 DVC systems.

Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Voice station audio vs. H.320 DVC system audio

When an Enhanced mode complex originates or receives a voice or multimedia call, the call is originated with the station handset or answered with the station handset. The audio path will be through the handset. If the user’s H.320 DVC system has speakers and a microphone, the user might wish to use the H.320 DVC system for audio in much the same manner as a built-in or separate telephone speakerphone. The user can move the station’s audio to the H.320 DVC system by selecting an **mm-pcaudio** feature button on the voice station. There is no feature access code for this function.

The **mm-pcaudio** feature button works very much like a speakerphone on/off button. If the station is off-hook and selects mm-pcaudio, audio is directed to the PC DVC system. The switch-hook can be placed on-hook. If the handset is taken off-hook, the audio moves back to the handset. If the **mm-pcaudio** button is selected while audio is already on the DVC system and the handset is on-hook, this acts as a speakerphone off action and disconnects the current call.

The **mm-pcaudio** feature button can be used for voice as well as multimedia calls. If the **mm-pcaudio** feature button is selected while on a voice only call, the DVC system is alerted and brought into the call. No video will be transmitted or displayed. Audio will be directed through the PC DVC system.

Switching between Basic and Enhanced modes

There might be occasions when an Enhanced mode complex needs to switch to Basic mode operation temporarily. One example is when a user wishes to make a direct point to point multimedia call originated directly from the H.320 DVC. Basic mode operation allows this functionality at the expense of losing multimedia call handling capabilities (i.e. hold/xfer/conf). To switch from Enhanced mode to Basic mode, the station can either select a mm-basic feature button or dial the mm-basic feature access code. Both of these actions are valid only if the Enhanced mode station has no multimedia calls active.

When in Basic mode, the status lamp for the mm-basic button, if present, will be on solid. The mm-basic feature button acts as a toggle. If the status lamp is on, when the button is selected, the lamp will go off and the station will return to Enhanced mode. The mm-enhanced feature access code will set the state of the station back to Enhanced. Switching to Enhanced mode is only valid if the associated H.320 DVC system is idle.

Note:

Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the **Station** screen will show **basic**. When in Enhanced mode this field on the **Station** screen will show **enhanced**. The current station Multimedia mode will be saved to translation when a `save translation` command is executed.

Forwarding of voice and multimedia calls

The Enhanced multimedia mode complex voice station can use the existing standard call forwarding mechanisms to activate forwarding for voice calls. If the forwarding destination is on the same server, then this will also forward multimedia calls as multimedia calls to the destination. If the forwarding destination is off-switch, multimedia calls will forward off-switch as voice-only calls. This is appropriate when the user will be at a location that is not able to receive multimedia calls.

To forward multimedia calls off-switch as multimedia calls, the user must activate multimedia call forwarding. This can be done with an **mm-cfwd** button or feature access code. The user can also activate standard voice call forwarding and select the mm-call button prior to entering the forwarding address.

Coverage

Multimedia calls to an Enhanced mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice Station of the Enhanced multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the **Station** screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the **Station** screen **Early Answer** field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Multimedia calls and off-net call coverage

If the principal station's coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Multimedia calls and coverage to voice mail

Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the **Early Answer** field on the **Hunt Group** screen set to **y**. This field will have no effect on voice calls to the voice mail system.

Hunt Groups using Enhanced Mode Complexes

When creating hunt groups with Enhanced multimedia mode complexes, only the station extension should ever be entered as a hunt group member. Any hunt group or ACD skill can include the voice station of an Enhanced multimedia complex as a member. The data extension of an Enhanced mode complex should never be entered as any hunt group member. A hunt group or skill might have a mix of members that are stand-alone stations and Enhanced mode complex stations. In order to guarantee that all members of the hunt group or skill can receive voice or multimedia calls, all members should have the **H.320** field on the **Station** screen set to **y**. Simple voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The **MM Early Answer** field on the **Hunt Group** screen tells the system to answer an incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately.

Other considerations

CMS measurements can indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Call association (routing)

Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya Communication Manager attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member.

In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group must have ANI information. The ANI information can be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same server as the hunt group are easily associated. If multimedia calls into a hunt group have insufficient ANI information (i.e. all calls from server X to sever Y include the LDN for server X), then as the volume of calls increases the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, Communication Manager will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Multimedia vectors

Very often, calls are routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls, you must set the **Multimedia** field on the **Call Vector** screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

Note:

Vectors which have the **Multimedia** field set must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the **Multimedia** field set to **y** should never be set up to route to a hunt group or number which is a data extension.

Interactions

Interactions are listed here only if the operation is different from standard.

Administered Connections

An Enhanced multimedia complex voice station can serve as the origination point or destination of an administered connection. If the Multimedia call feature access code is included in the administration of the administered connection, this will result in a video AC.

An Enhanced multimedia complex H.320 DVC system cannot serve as the origination point of an administered connection.

X-porting

You cannot use X in the **Port** field when administering a data module or the data endpoint in a multimedia complex. However, you can use this to administer the telephone.

Bridged Appearances

Enhanced multimedia complex voice station users can bridge onto a call if the user has a bridged appearance. If the bridged appearance is for a multimedia call, selecting the bridged appearance will result in a multimedia call.

Call Detail Recording

Each channel of a 2-channel multimedia call generates a separate CDR record that is tagged as data.

Call forwarding

Users cannot forward calls from a multimedia complex using multi-number dialing, either by **mm-multnmbr** button or feature access code.

Call Park

Any station can park a multimedia call, and unpark the call from another telephone. If a multimedia call is unparked by an Enhanced mode complex station, a multimedia call will result. Users cannot park or unpark calls using multimedia endpoints.

Call Pickup

Any member of a pickup group can answer a multimedia call after the call has begun alerting at a station call appearance. If the station picking up the call is an Enhanced mode complex station and the call is multimedia, a multimedia call will result. This is true for standard or directed call pickup.

Consult

After a multimedia call has been answered, consult can be used when transferring or conferencing the call.

COR/COS

The Class of Restriction and Class of Service for a multimedia call originated from an Enhanced multimedia complex are those of the voice station in the complex.

Data Call Setup

An Enhanced mode multimedia H.320 DVC system cannot originate calls from the DVC system. All calls, both voice or video are originated from the voice station.

Data Hotline

An Enhanced multimedia complex H.320 DVC endpoint cannot be used to originate a call for hotline dialing. In order to setup a video hotline function with an Enhanced mode complex, the hotline number administered for the voice station should include the Multimedia call feature access code.

Data Trunk Groups

Data trunk groups can be used to carry H.320 calls of a fixed (administered) bearer capability.

ISDN Trunk Groups

Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis that can be used to distinguish voice calls from multimedia calls.

Night Service

Incoming H.320 calls follow established night-service processing for data calls.

Remote Access

Communication Manager does not prevent Enhanced multimedia complexes from attempting to use remote access. However, these endpoints will most likely not be able to dial the necessary codes.

Station Hunting

Multimedia calls to Enhanced mode complex voice stations that have an extension administered in the hunt-to-station field hunt based on established hunting criteria. If the hunt-to-station is also an Enhanced mode complex station, a multimedia call will result when the call is answered.

Terminating Extension Groups

A multimedia call to a TEG can be answered by any member of the TEG. If the member answering the call is an Enhanced mode complex station, a multimedia call will result.

Telephone Display

Display information for calls to or from an Enhanced multimedia complex contains the display information associated with the voice station.

Troubleshooting

If one channel of a 2 B-channel call goes down, your choices are to continue with reduced transmission quality, or to hang up the call and start over. It is not possible to re-establish the second channel while the call is still active.

If you cannot share data with others, it might be that both parties do not have the same endpoint software. This is true for some data collaboration, but most whiteboard and file transfer software implementations are compatible.

Monitoring MMCH

This section briefly discusses some of the commands you can use to monitor multimedia complexes and conferences. The Maintenance manual for your Avaya server might discuss some of these commands and their output in more detail.

Action	Objects	Qualifier
display	station data module	xxxxx (extension) xxxxx (extension)
list	mmi measurements multimedia	 multimedia-interface voice-conditioner esm endpoints [‘print’ or ‘schedule’] h.320-stations [‘print’ or ‘schedule’]
status	attendant conference conference conference data module station trunk esm	xxxx (console number) all xxx (conference ID) xxx (conference ID) endpoint (endpoint ID) xxxxx (extension) xxxxx (extension) (group number or group number/ member number)

Status commands

The **status** commands for data module, station, trunk, and attendant provide the conference ID and endpoint ID for any of these involved in an active multimedia conference.

Figure 163: General Status screen

```
status station nnnn                                                 page 1 of x

                                 GENERAL STATUS

             Type: 7405D               Service State: in-service/on-hook
        Extension: 1002             Maintenance Busy? no
             Port: 01C0702             SAC Activated? no
      Call Parked? no               User Cntrl Restr: none
  Ring Cut Off Act? no             Group Cntrl Restr: none
Active Coverage Option: 1         CF Destination Ext:
                                     MM Conference ID:
                                       MM Endpoint ID:
   Message Waiting:
   Connected Ports:

            ACD STATUS                        HOSPITALITY STATUS
    Agent Logged In   Work Mode            AWU Call At:
                                              User DND: not activated
                                             Group DND: not activated
                                           Room Status: non-guest room
       On ACD Call? no
```

The following fields specific to multimedia appear on the station **General Status**, **Attendant**, **Data Module**, and **Trunk Group** screens.

- **MM Conference ID** — This field appears only if the station is active on a multimedia conference. It displays the ID for the conference. Enter this number with the **`status conference`** command to get more information about this conference.
- **MM Endpoint ID** — This field appears only if the station is active on a multimedia conference. It displays the endpoint ID for the station. Enter this number with the **`status conference endpoint`** command to learn more about this endpoint's involvement in the conference.

List commands

The **`list multimedia endpoints`** command shows you all the multimedia data modules that exist in your system, and their associated telephones, if any. The **`list multimedia H.320-stations`** command shows you all the stations that are administered for H.320 conversion. The **`list multimedia ip-stations`** command shows you the administered IP stations/modules and whether they are registered.

Figure 164: List Multimedia Endpoints screen

```
            MULTIMEDIA ENDPOINTS

Data Ext     MM Complex Voice Ext     H.320 Conversion?

100                  87654                    y
1321
15683                738                      n
```

Figure 165: List Multimedia H.320-Stations screen

```
                   MULTIMEDIA H.320-STATIONS

       Station Ext             MM Data Ext

          100                     87654
        1321
          15683                   738
```

Figure 166: List Multimedia IP-Stations screen

```
                   MULTIMEDIA IP STATIONS

             IP STATION             MEDIA COMPLEX
   Ext   Port   Registered?    Ext    Port    Registered?
    100              y       87654                 y
    1321
    15683            n       738                   n
```

Considerations

Each channel of a 2-channel BRI call takes one port on an MMI circuit pack. This alone limits the number of multimedia calls your system can handle. In addition, each conference takes one port on a voice-conditioner circuit pack. Also note that there is a limit to the total number of conversion calls the system can handle simultaneously. If you experience traffic problems after installing multimedia, you might want to reduce the number of stations that use H.320 conversion.

Chapter 10: Setting Up Telecommuting

Configuring Avaya Communication Manager for Telecommuting

Telecommuting emphasizes the ability to perform telephony activities while remote from Avaya Communication Manager. It is a combination of four features which permit you to remotely perform changes to your station's Coverage and Call Forwarding.

Note:

If you are operating in a Distributed Communications System (DCS) environment, you need to assign a different telecommuting-access extension to each Avaya S8XXX Server and tell your users which extension they should use. A user can set up call coverage from any of the DCS nodes, but needs to dial the telecommuting-access extension of the node on which their station is defined before using the feature access code.

You can also set up telecommuting with an IP (internet protocol) telephone. See Adding an IP Softphone on page 104 for more information.

- Coverage of Calls Redirected Off Net (CCRON) allows you to redirect calls off your network onto the public network and bring back unanswered calls for further coverage.

Note:

If a call covers or forwards off-net and an answering machine answers the call, or it is directed to a cellular telephone and a cellular announcement is heard, the server views this call as an answered call. Communication Manager does not bring the call back to the server for further routing.

- The Extended User Administration of Redirected Calls feature allows you to change the direction of calls to your station. This activates the capability to have two coverage-path options. These two path options can be specified on the **Station** screen; however, unless the **Can Change Coverage** field is set to **y** on the **Class of Restriction** screen, the second path option cannot be populated.
- The Personal Station Access feature gives you an extension number, a Merge feature access code, and a personalized security code, and tells you which office telephone you can use. This allows you to take your telephone, as long as the telephones are the same type, anywhere on the same server running Communication Manager.
- The Answer Supervision feature provides supervision of a call directed out of the server either by coverage or forwarding and determines whether Communication Manager should bring the call control back to its server.

Before you start

You can also set up telecommuting with an IP (internet protocol) telephone or IP Softphone. For example, see Adding an IP Softphone on page 104 for more information.

For DCP/ISDN telecommuting, ensure you have the following equipment:

- Call Classifier — Detector
- 1264-TMx software
- Avaya Communication Manager extender — switching module or standalone rack mount (Digital Communications Protocol (DCP) or Integrated Services Digital Network (ISDN))

  For more information about this equipment, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Verify the following fields on the **System Parameters Customer-Options (Optional Features)** screen are set to **y**.

- **Cvg Of Calls Redirected Off-Net**
- **Extended Cvg/Fwd Admin**
- **Personal Station Access**
- **Terminal Translation Initialization (TTI)**

If neither **Avaya Communication Manager extender** nor the **System Parameters Customer-Options (Optional Features)** fields are configured, contact your Avaya technical support representative.

Verify the telecommuting access extension is a direct inward dialing (DID) or a central office (CO) trunk destination for off-premises features to work.

Configure **TTI** for personal station access (PSA). For information about configuring TTI, see Setting up Personal Station Access on page 431.

Configure Security Violation Notification for Station Security Codes. For information about Security Violation Notification, see Setting up Security Violations Notification on page 466.

Instructions

In our example, we set up the telecommuting extension and enable coverage of calls redirected off-net.

To configure Avaya Communication Manager for telecommuting:

1. Type **`change telecommuting-access`**. Press **Enter**.

   The **Telecommuting Access** screen appears.

Figure 167: Telecommuting Access screen

```
add telecommuting-access                                          Page 1 of x
                         TELECOMMUTING ACCESS

                Telecommuting Access Extension: ________
```

2. In the **Telecommuting Access Extension** field, type **1234**. Press **Enter**.

   This is the extension you are configuring for telecommuting.

3. Type **`change system-parameters coverage`**. Press **Enter**.

   The **System Parameters Call Coverage/Call Forwarding** screen appears.

4. In the **Coverage Of Calls Redirected Off-Net Enabled** field, type **y**. Press **Enter**.

Related topics

See Telecommuting Access on page 1637 for information about and field descriptions on the **Telecommuting Access** screen.

Setting up Personal Station Access

Personal Station Access (PSA) allows you to associate the preferences and permissions assigned to your own extension with any other compatible telephone. When you request a PSA associate, the system automatically dissociates another extension from the telephone.

Preferences and permissions include the definition of terminal buttons, abbreviated dial lists, and class of service (COS) and class of restriction (COR) permissions assigned to your station. Extensions without a COS, such as Expert Agent Selection (EAS) agents or hunt groups, cannot use PSA.

PSA requires you to enter a security code and can be used on-site or off-site. Invalid attempts to associate a telephone generate referral calls and are recorded by Security Violation Notification, if that feature is enabled. If you interrupt the PSA dialing sequence by pressing the release button or by hanging up, the system does not log the action as an invalid attempt.

The disassociate function within PSA allows you to restrict the features available to a telephone. When a telephone has been dissociated using PSA, it can be used only to call an attendant, or to accept a TTI or PSA request. You can enable a dissociated set to make other calls by assigning a special class of restriction.

When a call that goes to coverage from a PSA-disassociated extension, Avaya Communication Manager sends a message to the coverage point indicating that the call was not answered. If the coverage point is a display telephone, the display shows **da** for "don't answer." If the coverage point is a voice-messaging system, the messaging system receives an indication from Communication Manager that this call was not answered, and treats the call accordingly.

Note:

Once a telephone has been associated with an extension, anyone using the terminal has the capabilities of the associated station. Be sure to execute a dissociate request if the terminal can be accessed by unauthorized users. This is particularly important if you use PSA and DCP extenders to permit remote DCP access.

Before you start

Verify that the **Personal Station Access** field is set to **y** on the **Class of Service** screen.

Verify that the extension has a COS that allows PSA.

Instructions

In our example, we specify the TTI State, the Record PSA/TTI Transactions, the class of service, and the feature access codes set up for PSA.

To set up Personal Station Access:

1. Type **`change system-parameters features`**. Press **Enter**.

   The **Feature-Related System Parameters** screen appears.

2. Complete the following fields. Press **Enter**.

   a. Type **voice** in the **TTI State** field.

   b. (Optional) Type **y** in the **Record CTA/PSA/TTI Transactions in History Log** field.

      These fields display only when the **Terminal Translation Initialization (TTI) Enabled** field on this screen is set to **y**.

3. Type **`change cos`**. Press **Enter**.

   The **Class of Service** screen appears.

4. Type **y** in the **Personal Station Access (PSA) 1** field. Press **Enter**.

5. Type **`change feature-access-codes`**. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.

6. Complete the following fields. Press **Enter**.

   a. Type **#4** in the **Personal Station Access (PSA) Associate Code** field.

      This is the feature access code you will use to activate Personal Station Access at a telephone.

   b. Type **#3** in the **Dissociate Code** field.

      This is the feature access code you will use to deactivate Personal Station Access at a telephone.

More information

You can allow users to place emergency and other calls from telephones that have been dissociated. To enable this, you must first assign a class of restriction (COR) for PSA-dissociated telephones. You do this on the **Feature-Related System Parameters** screen. In addition, you must set the restrictions for this COR on the **Class of Restriction** screen.

If you want users to be able to place emergency calls from dissociated telephones, it is also a good idea to have the system send calling party number (CPN) or automatic number identification (ANI) information for these calls. To do this, you must set the **CPN, ANI for Dissociated Sets** field to **y** on the **Feature-Related System Parameters** screen.

Related topics

See Changing Telecommuting Settings on page 447 for information on how to associate or disassociate PSA.

See Setting Up Enterprise Mobility User on page 171 for information on how to set up the Enterprise Mobility User feature.

Creating a Station Security Code

A Station Security Code (SSC) provides security to a station user by preventing other users from accessing functions associated with the user's station. Each station user can change their own SSC if they know the station's current settings.

You must create a system-wide SSC change feature access code (FAC) before users can change their SSC. You must also provide users with their individual SSC. A user cannot change a blank SSC.

Instructions

In our example, we set the station security code for a user.

To create a station security code:

1. Type **`change feature-access-codes`**. Press **Enter**.

   The [Feature Access Code (FAC)](#) screen appears.

2. Type **#5** in the **Station Security Code Change Access Code** field. Press **Enter**.

   This sets the access codes for this features. The Command prompt appears.

3. Type **`change system-parameters security`**. Press **Enter**.

   The [Security-Related System Parameters](#) screen appears.

4. Type **4** in the **Minimum Station Security Code Length** field. Press **Enter**.

   This determines the minimum required length of the Station Security Codes you enter on the **Station** screen. Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

5. Type **`change station 1234`**. Press **Enter**.

   This is the station extension you configured for telecommuting. The [Station screen](#) appears.

Figure 168: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
                                              Time of Day Lock Table:
              Loss Group: 2             Personalized Ringing Pattern: 3
             Data Module? n                         Message Lamp Ext: 1014
            Speakerphone: 2-way                  Mute button enabled? y
        Display Language? English
                   Model:                           Expansion Module?

 Survivable GK Node Name:                          Media Complex Ext:
          Survivable COR:                               IP Softphone? y
   Survivable Trunk Dest?                         Remote Office Phone? y
                                                   IP Video Softphone?
                                                             IP Video?

                                                   Customizable Labels?
```

6. Type **4321** in the **Security Code** field. Press **Enter**.

Related topics

See Station on page 1491 for information about and field descriptions on the **Station** screen.

See "Station Security Codes" in *Feature Description and Implementation for Avaya Communication Manager* (555-245-205*)* for a description of the Station Security Codes feature.

Assigning an Extender Password

Avaya Communication Manager allows you assign an extender password to a user. You can assign one password for each Avaya Communication Manager port.

Before you start

Use the Remote Extender PC in the server room to perform this procedure.

Instructions

In this example, we will set a system-generated random password for a user named John Doe.

To assign an extender password:

1. Double-click the **Security** icon.

   The **Password Manager** screen appears.
2. Double-click **User Password for User 01**.
3. Select **Enable Password** to enable the password.
4. Click **random**.

   This means that the password is a system generated random number. The system displays a 10-digit number in the **Password** field. Take note of this number, your user will need it at home to access the server running Communication Manager.
5. Type **Doe, John** and click **OK**.

   This is the last name and first name of the user. The system returns you to the **Password Manager** screen.
6. Select **CommLink:Select Cards**.

   A screen containing a list of cards (for example, Card A, Card B, and so on) appears. Each card corresponds to a port on your Avaya S8XXX Server.
7. Select **Card A** and click **OK**.

   The system returns you to the **Password Manager** screen.
8. Select **CommLink:Upload Password**.

   The error message screen appears with the message **Administrator password not loaded"**.

9. Click **OK**.

   The **Administrator** screen appears.

10. Type **123456** and click **OK**.

    This is the administrator's password.

11. Select **CommLink:Upload Password**.

    The password is uploaded.

12. When upload is complete, click **OK**.

    The system returns you to the **Password Manager** screen.

13. Select **File:Save As**.

    The **Save As** screen appears.

14. Type **doe.fil** in the **File** field and click **OK**.

The system saves the User01 information.

Setting up Call Forwarding

Avaya Communication Manager allows you to change your call forwarding from any on-site or off-site location.

Instructions

In our example, we assign the feature access codes and class of service to set up call forwarding. This allows your users to forward their calls to another extension.

To set up call forwarding:

1. Type `change feature-access-codes`. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.

2. Set a 2-digit access code for the following fields. Press **Enter**.

   a. Type ***8** in the **Extended Call Fwd Activate Busy D/A** field.

   b. Type ***7** in the **Extended Call Fwd Activate All** field.

   c. Type ***6** in the **Extended Call Fwd Activate Deactivation** field.

      This sets the access codes for these features. The Command prompt appears.

3. Type `change cos`. Press **Enter**.

   The **Class of Service** screen appears.

4. Set the following fields to **y**.
   - **Extended Forwarding All**
   - **Extended Forwarding B/DA**

   This allows you to change the forwarding of all your calls from an off-site location.
5. Set the **Restrict Call Fwd-Off Net** field to **n**. Press **Enter**.

   This allows your users to forward calls off-site.

Interactions

- Bridged Appearance

  When the pound key (#) is pressed from a bridged appearance immediately following any of this feature's four feature access codes (FACs), the system assumes that the currently active bridged extension will be administered. The station security code of the currently active bridged extension must be entered after the initial # to successfully complete the command sequence.

  If the station has only bridged appearances, the station's extension must be dialed after the FAC to successfully complete the command sequence, since the station's extension is not associated with any appearances.
- Distributed Communications System

  Assign a different telecommuting access extension for each server running Communication Manager. You can use Extended User Administration of Redirected Calls from any of the DCS nodes, but you must dial the extension of the node on which your station is defined before dialing the FAC.
- Tenant Partitioning

  The telecommuting access extension is always automatically assigned to Tenant Partition 1, so it can be accessed by all tenants.

  The tenant number of the extension being administered must be accessible by the tenant number from which the Extended User Administration of Redirected Calls FAC is dialed or the request is denied. If the FAC is dialed on site, the tenant number of the station or attendant must have access to the tenant number of the extension administered. If the FAC is dialed off site, the tenant number of the incoming trunk must have access to the tenant number of the extension administered.

Related topics

See Changing Telecommuting Settings on page 447 for information on how to change call forwarding.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Call Forwarding feature.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Tenant Partitioning feature.

Assigning Coverage Options

Avaya Communication Manager allows you to assign two previously administered coverage paths and/or time of day coverage tables on the **Station** screen. This allow telecommuters to alternate between the two coverage paths and/or time of day coverage tables administered to better control how their telephone calls are handled.

For information about creating a coverage path, see Creating coverage paths on page 216. For information about creating a time of day coverage table, see Assigning a coverage path to users on page 218.

Instructions

In our example, we assign two coverage options so a user can choose from either option to control how their calls are handled.

To assign 2 coverage options:

1. Type **`change feature-access-codes`**. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.
2. Type **#9** in the **Change Coverage Access Code** field. Press **Enter**.
3. Type **`change cor 1`**. Press **Enter**.

   The **Class of Restriction** screen appears.
4. In the **Can Change Coverage** field, type **y**. Press **Enter** to save your work.

   The Command prompt appears.
5. Type **`change station 1234`**. Press **Enter**.

   This is the station extension you configured for telecommuting. The **Station** screen appears.

6. Complete the following fields:
   a. Type **2** in the **Coverage Path 1** field.
   b. Type **8** in the **Coverage Path 2** field.

Related topics

See Coverage Path on page 881 for information about and field descriptions on the **Coverage Path** screen.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Call Coverage feature.

See Changing Telecommuting Settings on page 447 for information on how to alternate your coverage path option.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for information about the Extended User Administration of Redirected Calls feature.

Installing Home Equipment

Avaya Communication Manager allows you to install equipment in your home so that you can utilize system facilities from off-site.

Before you start

You can also set up telecommuting with an IP (internet protocol) telephone or IP Softphone. For example, see Adding an IP Softphone on page 104 for more information.

For DCP telecommuting, you need the following equipment:

- Avaya Communication Manager extender remote module
- DCP sets (office and home must match)

Configure a feature access code for associating your home number to your office number. For information about configuring an associate feature access code, see Setting up Personal Station Access on page 431.

Instructions

Installing home equipment

To install your home equipment:

1. Plug the telephone cord into the slot labeled line on the back of the module and into the wall jack.
2. Plug the telephone cord into the slot labeled port on the back of the module and into the slot labeled line on the telephone.
3. Plug the power cord into slot labeled power on the back of the module and the wall socket.

   The telephone display **Go Online** appears.
4. Press **3 (Nxt)**.

   The telephone display **Set Phone Number** appears.
5. Press **2 (OK)** to set the telephone number.
6. Type **5551234**. Press **Drop.**

   This is the assigned analog telephone number. In some areas, you might need to include your area code (for example, 3035551234). The telephone display **Set Phone Number** appears.
7. Press **1(Prv)**.

   This returns you to the **Go Online** telephone display.
8. Press **2 (OK)**.

   The module dials the number. When the modules connect, the telephone display **Enter Password** appears.
9. Type **0123456789**. Press **Drop**.

Associating your office telephone number to the home station

To associate your telephone number:

1. On your home station, type **#4**.

   This is the associate feature access code.
2. Type **4321**. Press **#**.

   This is your extension number.
3. Type **1996**. Press **#**.

   This is your password.

Disassociating your home station

To disassociate your home station:

1. Press **Hold** four times.

Related topics

See Configuring Avaya Communication Manager for Telecommuting on page 429 for step-by-step instructions on how to configure your office equipment.

See Changing Telecommuting Settings on page 447 for step-by-step instructions on how to use your home station.

Setting up Remote Access

Remote Access permits a caller located outside the system to access the server running Avaya Communication Manager through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer’s responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

If you do not intend to use Remote Access now or in the future, you can permanently disable the feature. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

Before you start

Configure the **Incoming Destination** and **Night Service** fields on the **CO Trunk** screen. For information about configuring a CO trunk, see Adding a CO, FX, or WATS trunk group on page 480.

Verify that the **Authorization Codes** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**.

Verify that the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Instructions

Setting up remote access

In our example, we set up a remote access extension with maximum security. This assists you in blocking unauthorized people from gaining access to your network.

To set up remote access:

1. Type `change remote-access`. Press **Enter**.

   The **Remote Access screen** appears.

Figure 169: Remote Access screen

```
change remote-access                                                Page 1 of x

                                    REMOTE ACCESS
   Remote Access Extension: ________                  Barrier Code Length:____
Authorization Code Required? y                  Remote Access Dial Tone: n

     Barrier        COR  TN  COS  Expiration    No. of     Calls
      Code          COR  TN  COS     Date       Calls      Used
   1:________       1__  1_  1__  __/__/__      ______     ______
   2:________       1__  1_  1__  __/__/__      ______     ______
   3:________       1__  1_  1__  __/__/__      ______     ______
   4:________       1__  1_  1__  __/__/__      ______     ______
   5:________       1__  1_  1__  __/__/__      ______     ______
   6:________       1__  1_  1__  __/__/__      ______     ______
   7:________       1__  1_  1__  __/__/__      ______     ______
   8:________       1__  1_  1__  __/__/__      ______     ______
   9:________       1__  1_  1__  __/__/__      ______     ______
  10:_______        1__  1_  1__  __/__/__      ______     ______

Permanently Disable? __  Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

2. Type **1234** in the **Remote Access Extension** field.

   This is the extension specified in the **Incoming Destination** field on the **CO Trunk** screen.

3. Type **7** in the **Barrier Code Length** field.

   This is the number of digits your barrier code must be when entered.

4. Type **y** in the **Authorization Code Required** field.

   This means you must also enter an authorization code when you access the system's Remote Access facilities. For information about setting up access codes, see Setting up Authorization Codes on page 461.

5. Type **y** in the **Remote Access Dial Tone** field.

   This means you hear dial tone as a prompt to enter your authorization code.

6. Type **1234567** in the **Barrier Code** field.

   This is the 7-digit barrier code you must enter to access the system's Remote Access facilities.

7. Type **1** in the **COR** field.

   This is the class of restriction (COR) number associated with the barrier code that defines the call restriction features.

8. Type **1** in the **TN** field.

   This is the Tenant Partition (TN) number.

9. Type **1** in the **COS** field.

   This is the class of service (COS) number associated with the barrier code that defines access permissions for Call Processing features.

10. Type **01/01/04** in the **Expiration Date** field.

    This is the date the barrier code expires. A warning message is displayed on the system copyright screen seven days before the expiration date. The system administrator can modify the expiration date to extend the time interval, if necessary.

11. Type **y** in the **Disable Following A Security Violation** field.

    This disables the remote access feature following detection of a remote access security violation.

12. Press **Enter** to save your work.

Disabling remote access permanently

To disable remote access permanently:

1. Type `change remote-access`. Press **Enter**.

   The **Remote Access** screen appears.

2. Type **y** in the **Permanently Disable** field.

   If you permanently disable this feature, it requires Avaya Services intervention to reactivate the feature. There is a charge for reactivation of this feature.

3. Press **Enter** to save your work.

   CAUTION:

   Your attempt to disable the Remote Access feature will be lost if the server running Communication Manager is rebooted without saving translations. Therefore, execute a `save translation` command after permanently disabling the Remote Access feature.

Using Secure Shell for remote login

You can log in remotely to the following platforms using Secure Shell (SSH) as a secure protocol:

- G350 Media Gateway
- S8300, S8400, S8500, or S87XX Server Linux command line
- Communication Manager System Administration Terminal (SAT) interface on an Avaya S8XXX Server using port 5022.

The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system. For details on disabling Telnet, see Turning off Telnet for increased security.

Note:

The client device for remote login must also be enabled and configured for SSH. Refer to your client P.C. documentation for instructions on the proper commands for SSH.

More information

Remote Access provides you with access to the system and its features from the public network. This allows you to make business calls from home or use Recorded Telephone Dictation Access to dictate a letter. If authorized, you can also access system features from any on-site extension.

With Remote Access you can dial into the system using Direct Inward Dialing (DID), Central Office (CO), Foreign Exchange (FX), or 800 Service trunks. When a call comes in on a trunk group dedicated to Remote Access, the system routes the call to the Remote Access extension you have assigned. If DID is provided and the Remote Access extension is within the range of numbers that can be accessed by DID, Remote Access is accessed through DID.

Barrier codes provide your system security and define calling privileges through the administered COR. You can administer up to 10 barrier codes, each with a different COR and COS. Barrier codes can be from 4 to 7 digits, *but all codes must be the same length*. You can also require that users enter an authorization code to use this feature. Both barrier codes and authorization codes are described under Setting up Authorization Codes on page 461.

Related topics

See QSIG to DCS TSC Gateway on page 1427 for information about and field descriptions on the **Remote Access** screen.

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Remote Access feature.

Changing Telecommuting Settings

Avaya Communication Manager allows you to associate and disassociate PSA, change the coverage path for your station, change the extension to which you forward your calls, and change your personal station's security code.

Before you start

Configure PSA. For information about configuring PSA, see Setting up Personal Station Access on page 431.

Assign two coverage options for your system. For information on how to assign coverage options, see Assigning Coverage Options on page 439.

Configure call forwarding for your system. For information about configuring call forwarding, see Setting up Call Forwarding on page 437.

Configure security codes for a station. For information about configuring personal station security codes, see Assigning an Extender Password on page 436.

Instructions

Associating PSA

In this example, we associate PSA (preferences and permissions) assigned to your station with another compatible terminal.

To associate PSA:

1. Dial **#4**.

   This is the associate PSA feature access code. You hear dial tone.

2. Type **1234**. Press **#**.

   This is your extension.

3. Type **4321**. Press **#**.

   This is your Station Security Code. You hear a confirmation tone.

Disassociating PSA

In our example, we disassociate PSA from the station you are using.

To disassociate PSA:

1. Dial **#3**.

   This is the disassociate PSA feature access code. You are no longer PSA associated to this station.

Changing a coverage option

In this example, we change the coverage option from path 1 to path 2 from a remote location.

To change a coverage option:

1. Dial **1234**.

   This is the extension you configured for telecommuting. You hear dial tone.

2. Dial **#9**. Press **#**.

   This is the feature access code you set for changing a coverage path. You hear dial tone.

3. Dial **4321**. Press **#**.

   This is the extension for which you want to change the coverage path.

4. Dial **87654321**. Press **#**.

   This is the extension security code.

5. Dial **2**.

   This is the new coverage path. You hear confirmation tone.

Changing call forwarding

In this example, we change call forwarding to extension 1235.

To change call forwarding:

1. Dial **1234**.

   This is the extension you configured for telecommuting.

2. Dial **#8. Press #**.

   This is the feature access code you set for activating extended call forward. You hear dial tone.

3. Dial **4321**. Press **#**.

   This is the extension from which you want to forward calls.

4. Dial **87654321**. Press **#**.

   This is the extension security code. You hear dial tone.

5. Dial **1235**.

   This is the extension to which you want to forward calls. You hear the confirmation tone.

Changing your personal station security codes

In this example, we change the security code for extension 1235 from 98765432 to 12345678.

To change your security code:

1. Dial **#5**.

   This is the feature access code you set for changing your security code. You hear dial tone.

2. Dial **1235**. Press **#**.

   This is the extension for which you want to change the security code.

3. Dial **98765432**. Press **#**.

   This is the current security code for the extension. You hear dial tone.

4. Dial **12345678**. Press **#**.

   This is the new security code. Security codes can be 3-8 digits long.

5. Dial **12345678**. Press **#**.

   This is to confirm your new security code. You hear the confirmation tone.

   Note:

   If you cannot change your security code, Manager 1 can clear the problem using the `Clear Audit Summary` command.

Interrupting the command sequence for personal station security codes

To interrupt the command sequence for personal station security codes:

1. To interrupt the command sequence before step 3, choose one of these options:

   - Hang up or press the disconnect or recall button before hearing intercept tone in step 3.

     The system does not log an invalid attempt. You must restart the process at step 1.

   - Type ***** before the second # in step 3.

     You must begin the change sequence at the point of entering your extension in step 2. (You should not enter the FAC again.)

   - Type ***** after the FAC has been entered and before the final #.

     You must restart the process at step1.

2. To interrupt the command sequence after step 3, type ***** in steps 4 or 5, you must begin the change sequence at the point of entering the new station security code (SSC) in step 4.

   If you hear intercept tone in any step, the command sequence has been invalidated for some reason and you must restart the process at step 1.

   If you hear intercept tone after step 3, the system logs an invalid attempt via the Security Violations Notification (SVN) feature. This is true even if you attempt to interrupt the change sequence with an asterisk.

Related topics

See *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a description of the Security Violations Notification (SVN) feature.

Chapter 11: Enhancing System Security

Basic Security

Keeping your system secure

The following is a partial list you can use to help secure your system. It is not intended as a comprehensive security checklist. See the *Avaya Toll Fraud and Security Handbook,* 555-025-600, for more information about these and other security-related features.

- Secure the system administration and maintenance ports and/or logins on Avaya Communication Manager using the Access Security Gateway. This optional password authentication interface program is provided to customers with maintenance contracts.
- Activate Security Violations Notification to report unsuccessful attempts to access the system. Security Violations Notification lets you automatically disable a valid login ID following a security violation involving that login ID and disable remote access following a security violation involving a barrier code or authorization code.
- Secure trunks using Automatic Route Selection (ARS), Class of Restriction (COR), Facility Restriction Levels (FRLs) and Alternate Facility Restriction Levels (AFRLs), Authorization Codes, Automatic Circuit Assurance (ACA), and Forced Entry of Account Codes (see "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information).
- You can log in remotely using Secure Shell (SSH) as a secure protocol. The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system.

Activate Enhanced Call Transfer for your voice messaging system, if available. This limits transfers to valid extensions, but you also need to restrict transfers to extensions that might offer dial tone to the caller, such as screen extensions.

Preventing Toll Fraud

Top 15 tips to help prevent toll fraud

Toll Fraud Tips:

1. Protect system administration access

   Make sure secure passwords exist for all logins that allow System Administration or Maintenance access to the system. Change the passwords frequently.

   Set logoff notification and forced password aging when administering logins. You must assign passwords for these logins at setup time.

   Establish well-controlled procedures for resetting passwords.

2. Prevent voice mail system transfer to dial tone

   Activate "secure transfer" features in voice mail systems.

   Place appropriate restrictions on voice mail access/egress ports.

   Limit the number of invalid attempts to access a voice mail to five or less.

3. Deny unauthorized users direct inward system access (screen)

   If you are not using the Remote Access features, deactivate or disable them.

   If you are using Remote Access, require the use of barrier codes and/or authorization codes set for maximum length. Change the codes frequently.

   It is your responsibility to keep your own records regarding who is allowed to use which authorization code.

4. Place protection on systems that prompt callers to input digits

   Prevent callers from dialing unintended digit combinations at prompts.

   Restrict auto attendants and call vectors from allowing access to dial tone.

5. Use system software to intelligently control call routing

   Create Automatic Route Selection or World Class Routing patterns to control how each call is to be handled.

   Use "Time of Day" routing capabilities to limit facilities available on nights and weekends.

   Deny all end-points the ability to directly access outgoing trunks.

6. Block access to international calling capability

   When international access is required, establish permission groups.

   Limit access to only the specific destinations required for business.

7. Protect access to information stored as voice

   Password restrict access to voice mail mailboxes.

   Use non-trivial passwords and change passwords regularly.

8. Provide physical security for telecommunications assets

   Restrict unauthorized access to equipment rooms and wire connection closets.

   Protect system documentation and reports data from being compromised.

9. Monitor traffic and system activity for abnormal patterns

   Activate features that "turn off" access in response to unauthorized access attempts.

   Use Traffic and Call Detail reports to monitor call activity levels.

10. Educate system users to recognize toll fraud activity and react appropriately

    From safely using calling cards to securing voice mailbox password, train your users on how to protect themselves from inadvertent compromises to the system's security.

11. Monitor access to the dial-up maintenance port. Change the access password regularly and issue it only to authorized personnel. Consider activating Access Security Gateway. See "Access Security Gateway" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information.

12. Create a system-management policy concerning employee turnover and include these actions:

    a. Delete any unused voice mailboxes in the voice mail system.

    b. Immediately delete any voice mailboxes belonging to a terminated employee.

    c. Immediately remove the authorization code if a terminated employee had screen calling privileges and a personal authorization code.

    d. Immediately change barrier codes and/or authorization codes shared by a terminated employee. Notify the remaining users of the change.

    e. Remove a terminated employee's login ID if they had access to the system administration interface. Change any associated passwords immediately.

13. Back up system files regularly to ensure a timely recovery. Schedule regular, off-site backups.

14. Callers misrepresenting themselves as the "telephone company," "AT&T," "RBOCS," or even known employees within your company might claim to be testing the lines and ask to be transferred to "900," "90," or ask the attendant to do "start 9 release." This transfer reaches an outside operator, allowing the unauthorized caller to place a long distance or international call. Instruct your users to never transfer these calls. Do not assume that if "trunk to trunk transfer" is blocked this cannot happen.

Hackers run random generator PC programs to detect dial tone. Then they revisit those lines to break barrier codes and/or authorization codes to make fraudulent calls or resell their services. They do this using your telephone lines to incur the cost of the call. Frequently these call/sell operations are conducted at public payphones located in subways, shopping malls, or airport locations. See QSIG to DCS TSC Gateway on page 1427 to prevent this happening to your company.

Physical Security

Physical security is your responsibility. Implement the following safeguards as an added layer of security:

1. Unplug and secure attendant console handsets when the attendant position is not in use.
2. Lock wiring closets and server rooms.
3. Keep a log book register of technicians and visitors.
4. Shred all Communication Manager information or directories you discard.
5. Always demand verification of a technician or visitor by asking for a valid I.D. badge.
6. Keep any reports that might reveal trunk access codes, screen barrier codes, authorization codes, or password information secure.
7. Keep the attendant console and supporting documentation in an office that is secured with a changeable combination lock. Provide the combination only to those individuals who need to enter the office.
8. Keep any documentation pertaining to Communication Manager operation secure.
9. Label all backup tapes or flash cards with correct dates to avoid using an outdated one when restoring data. Be sure that all backup media have the correct generic software load.

System Security Checklist

Here's some of the steps required for indemnification. Use these to analyze your system security.

1. Remove all default factory logins of **cust**, **rcust**, **browse**, **nms**, and **bcms** and assign unique logins with 7-character alphanumeric passwords and a 90-day password aging. Use the `list logins` command to find out what logins are there.
2. If you do not use Remote Access, be sure to disable it permanently.

   Tip:

   You can use the `display remote-access` command to check the status of your remote access.

   To disable Remote Access, on the **Remote Access** screen, in the **Permanently Disable** field, type **y**.

Note:

Avaya recommends that you permanently disable Remote Access using the **`change remote-access`** command. If you do permanently disable Remote Access, the code is removed from the software. Avaya charges a fee to restore the Remote Access feature.

3. If you use Remote Access, but only for internal calls, change announcements or remote service observing.
   a. Use a 7-digit barrier code.
   b. Assign a unique COR to the 7-digit barrier code.

      The unique COR must be administered where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, the **Calling Permissions** field is **n** on all unique Trunk Group COR.
   c. Assign **Security Violation Notification Remote** to **10** attempts in **2** minutes.
   d. Set the aging cycle to **90** days with **100** call limit per barrier code.
4. If you use Remote Access to process calls off-net or in any way access the public network:
   a. Use a 7-digit barrier code.
   b. Assign a unique COR to the barrier code.
   c. Restrict the COR assigned to each barrier code by FRL level to only the required calling areas to conduct business.
   d. Set the aging cycle to **90** days with **100** call limit per barrier code.
   e. Suppress dial tone where applicable.
   f. Administer Authorization Codes.
   g. Use a minimum of 11 digits (combination of barrier codes and authorization codes).
   h. Assign **Security Violation Notification Remote** to 10 attempts in 2 minutes.
5. If you use vectors:
   a. Assign all Vector Directory Numbers (VDN) a unique COR. See *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779, and *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for more information.

Note:

The COR associated with the VDN dictates the calling privileges of the VDN/vector. High susceptibility to toll fraud exists on vectors that have "collect digits" steps. When a vector collects digits, it processes those digits back to Communication Manager and if the COR of the VDN allows it to complete the call off-net, it will do so. For example, the announcement "If you know your party's 4-digit extension number, enter it now" results in 4 digits being collected in step 6. If you input "90##" or "900#", the 4 digits are analyzed and if "9" points towards ARS and "0" or "00" is assigned in the ARS Analysis Tables and the VDN COR allows it, the call routes out of the server to an outside local exchange or long distance operator. The operator then connects the call to the requested number.

b. If vectors associated with the VDN do not require routing the call off-net or via AAR, assign a unique COR where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, the **Calling Permissions** field is **n** on all unique Trunk Group COR.

c. If the vector has a "route-to" step that routes the call to a remote server via AAR, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an AAR call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate AAR route patterns on the AAR Partition Group using the **`change aar analysis partition x 2`** command.

Tip:

You can use the **`display aar analysis print`** command to print a copy of your Automatic Alternate Routing (AAR) setup before making any changes. You can use the printout to correct any mistakes.

d. If the vector has a "route-to" step that routes the call to off-net, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an ARS call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate complete dial string in the "route-to" step of the vector the unique ARS Partition Group using the **`change ars analysis partition x 2`** command.

6. On the **Feature Access Code (FAC)** screen, **Facility Test Calls Access Code**, the **Data Origination Access Code**, and the **Data Privacy Access Code** fields, change from the default or remove them.

Note:

These codes, when dialed, return system dial tone or direct access to outgoing trunking facilities. Transfers to these codes can take place via an unsecured vector with "collect digits" steps or an unsecured voice mail system.

7. Restrict Call Forwarding Off Net on every class of service.

See Class of Service on page 852 for more information on Class of Service.

Note:

You cannot administer loop-start trunks if Call Forwarding Off Net is required.

8. If loop start trunks are administered on Communication Manager and cannot be changed by the Local Exchange Company, block all class of service from forwarding calls off-net. In the **Class of Service** screen, **Restriction Call Fwd-Off Net** field, set to **y** for the 16 (0-15) COS numbers.

   See Class of Service on page 852 for more information.

   Note:

   If a station is call forwarded off-net and an incoming call to the extension establishes using a loop-start trunk, incorrect disconnect supervision can occur at the Local Exchange Central Office when the call terminates. This gives the caller recall or transfer dial tone to establish a fraudulent call.

9. Administer Call Detail Recording on all trunk groups to record both incoming and outgoing calls.

   See Collecting Information About Calls on page 635 for more information.

10. On the **Route Pattern** screen, be careful assigning route patterns with an **FRL** of **0**; these allow access to outgoing trunking facilities. Avaya recommends assigning routes with an FRL of 1 or higher.

    Note:

    An exception might be assigning a route pattern with an FRL of 0 to be used for 911 calls so even restricted users can dial this in emergencies.

    Tip:

    You can use the `list route-pattern print` command to print a copy of your FRLs and check their status.

11. On all **Trunk Group** screens, set the **Dial Access** field to **n**. If set to **y**, it allows users to dial Trunk Access Codes, thus bypassing all the ARS call screening functions.

    See the Trunk Group section of the Screen Reference for more information.

12. On the AAR and ARS Digit Analysis Table on page 723, set all dial strings not required to conduct business to **den** (deny).

13. If you require international calling, on the AAR and ARS Digit Conversion Table on page 730, use only the 011+ country codes/city codes or specific dial strings.

14. Assign all trunk groups or same trunk group types a unique Class of Restriction. If the trunk group does not require networking through Communication Manager, administer the Class of Restriction of the trunk group where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, and all unique Class of Restriction assigned to your outgoing trunk groups are **n**. See Class of Restriction on page 834 for more information.

    Tip:

    You can use the `list trunk-group print` command to have a printout of all your trunks groups. Then, you can use the `display trunk-group x` command (where *x* is the trunk group) to check the COR of each trunk group.

15. For your AUDIX, on the **System Appearance** screen, set:
    - the **Enhanced Call Transfer** field to **y**.
    - the **Transfer Type** field to **enhanced**. If set to **basic**, set the **Transfer Restriction** field to **subscribers**. See Feature-Related System Parameters on page 1001 for more information.

    Note:

    The COR of the voice mail ports dictates the calling restrictions of the voice mail. If the above settings are not administered correctly, the possibility exists to complete a transfer to trunk access codes or ARS/AAR feature codes for fraudulent purposes. Never assign mailboxes that begin with the digits or trunk access codes of ARS/AAR feature access codes. Require your users to use a mailbox password length greater than the amount of digits in the extension number.

16. Avaya recommends you administer the following on all voice mail ports:
    - Assign all voice mail ports a unique COR. See Class of Restriction on page 834 for more information.
    - If you are not using outcalling, fax attendant, or networking, administer the unique COR where the **FRL** is **0**, the **Calling Party Restriction** field is **outward**, and all unique trunk group COR on the Calling Permissions are **n**. See Class of Restriction on page 834 for more information.

    Note:

    Avaya recommends you administer as many layers of security as possible. You can implement Step 9 and Step 16 as a double layer of security. In the event that the voice mail system becomes unsecured or compromised for any reason, the layer of security on Avaya Communication Manager takes over, and vice versa.

17. Administer all fax machines, modems, and answering machines analog voice ports as follows:
    - Set the **Switchhook Flash** field to **n**.
    - Set the **Distinctive Audible Alert** field to **n**. See Station on page 1491 for more information.

18. Install a Call Accounting System to maintain call records. In the **CDR System Parameters** screen, **Record Outgoing Calls Only** field, set to **y**. See CDR System Parameters on page 819 for more information.

19. Call Accounting Systems produce reports of call records. It detects telephones that are being hacked by recording the extension number, date and time of the call, and what digits were dialed.

Administering User Profiles and Logins

Authentication, Authorization and Accounting (AAA) Services allows you to store and maintain administrator account (login) information on a central server. Login authentication and access authorization is administered on the central server.

For details on administering user profiles and logins, see "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, and *Maintenance Commands for Avaya Communication Manager*, 03-300431.

Using Access Security Gateway (ASG)

For more information on ASG, see "Access Security Gateway" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

For more information on SVN, see "Security Violations Notification" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Using Busy Verify

This section shows you how to use Busy Verify (also known as Busy Verification) to help find fraud problems.

When you suspect toll fraud, you can interrupt the call on a specified trunk group or extension number and monitor the call in progress. Callers will hear a long tone to indicate the call is being monitored.

SECURITY ALERT:
Listening to someone else's calls might be subject to federal, state, or local laws, rules, or regulations. It might require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

Before you start

On the **Trunk Group screen - page 1**, verify the **Dial Access** field is **y**. If it is not, contact your Avaya technical support representative.

Instructions

To use busy verify:

1. Type **`change station xxxx`**, where ***`xxxx`*** is the station to be assigned the busy verify button. Press **Enter**.

   For this example, enter extension **1014**. Press Next Page until you see the **Site Data** fields.

   The **Station screen (page 4)** appears.

Figure 170: Station screen

```
add station nnnn                                                   Page  4 of  X
                                    STATION

     SITE DATA
          Room: _______                                     Headset? n
          Jack: ___                                         Speaker? n
         Cable: ___                                        Mounting: d
         Floor: _______                                Cord Length: 0_
      Building: _______                                  Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                6:limit-call
 2: call-appr                7:team     Ext: 5381231          Rg:
 3: call-appr                8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000     9:cfwd-enh Ext: 5502
 5: release                 10:aux-work RC: 1 Group:

voice-mail Number:
```

2. In the **BUTTON ASSIGNMENTS** area, type **verify**.
3. Press **Enter** to save your changes.
4. To activate the feature, press the **Verify** button on the telephone and then enter the Trunk Access Code and member number to be monitored.

Setting up Authorization Codes

Authorization codes provide the means for extending control of system users' calling privileges. They extend calling-privilege control and provide an extra level of security for remote-access callers.

Note:

To maintain system security, Avaya recommends you use authorization codes.

See the *Avaya Toll Fraud and Security Handbook*, 555-025-600 for more information.

Before you start

On the **System Parameters Customer-Options (Optional Features)** screen, verify the **Authorization Codes** field is **y**. If not, contact your Avaya representative. This field turns on the feature and permits you to selectively specify levels of calling privileges that override in-place restrictions.

Instructions

To set up authorization codes:

1. Type `change system-parameters features`. Press **Enter**.

   The **Feature-Related System Parameters screen** appears. Click **Next** until you find the **Authorization Code Enabled** field.

Figure 171: Feature-Related System Parameters screen

```
change system-parameters features                            page 4 of x
                      FEATURE-RELATED SYSTEM PARAMETERS
                       Reserved Slots for Attendant Priority Queue: 5_
                                        Time Before Off-Hook Alert: 10__
                            Emergency Access Redirection Extension: ________
              Number of Emergency Calls Allowed in Attendant Queue: __
                                              Call Pickup Alerting? n
                       Temporary Bridged Appearance on Call Pickup? y
                                     Call Pickup on Intercom Calls? y
                                              Directed Call Pickup? n
                                        Extended Group Call Pickup: flexible
                 Deluxe Paging and Call Park Timeout to Originator? n
     Controlled Outward Restriction Intercept Treatment: tone
    Controlled Termination Restriction (Do Not Disturb): tone
              Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS           Authorization Code Enabled? y
                                         Authorization Code Length: 7
                            Authorization Code Cancellation Symbol? #
                                           Attendant Time Out Flag? n
                                        Display Authorization Code? _
                  Controlled Toll Restriction Replaces: station-station
          Controlled Toll Restriction Intercept Treatment: extension     3000
```

2. In the **Authorization Code Enabled** field, type **y**.

   This enables the Authorization Codes feature on a system-wide basis.

3. In the **Authorization Code Length** field, type **7**.

   This defines the length of the Authorization Codes your users need to enter. To maximize the security of your system, Avaya recommends you make each authorization code the maximum length allowed by the system.

4. In the **Authorization Code Cancellation Symbol** field, leave the default of **#**.

   This is the symbol a caller must dial to cancel the 10-second wait period during which your user can enter an authorization code.

5. In the **Attendant Time Out Flag** field, leave the default of **n**.

   This means a call is not to be routed to the attendant if a caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

6. In the **Display Authorization Code** field, type **n**.

   This prevents the authorization code from displaying on telephone sets thus maximizing your security.

7. Press **Enter** to save your changes.

8. Type **`change authorization-code`** ***nnnn***, where ***nnnn*** is the authorization code. Press **Enter**.

   The Authorization Code - COR Mapping screen appears.

Figure 172: Authorization Code - COR Mapping screen

```
change authorization-code                                          Page 1 of x
                       Authorization Code - COR Mapping
      Note:    XX codes administered.  Use "list" to display all codes.
     AC     COR   AC     COR   AC     COR   AC     COR   AC      COR    AC     COR

    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
```

9. In the **AC** field, enter the authorization code your users must dial.

   In this example, type **4285193**. The number of digits entered must agree with the number assigned in the **Feature-Related System Parameters** screen, **Authorization Code Length** field.

   Note:

   Remember, all authorization codes used in the system must be the same length.

10. In the **COR** field, enter the desired Class of Restriction number from 0 through 95.

    In our example, type **1**.

11. Type **`change trunk-group`** ***n***, where ***n*** is the assigned trunk group number. Press **Enter**.

    The Trunk Group screen - page 1 appears.

Figure 173: Trunk Group screen

```
add trunk-group next                                                 Page 1 of x
                                   TRUNK GROUP

Group Number: 8                    Group Type: co CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1 TN: 1 TAC:
   Direction: two-way        Outgoing Display? n
 Dial Access? n             Busy Threshold: 255   Night Service: 1234567890123
Queue Length: 0           Country: 1       Incoming Destination: 1234567890123
   Comm Type: voice                  Auth Code? n    Digit Absorption List:
    Prefix-1? y                    Trunk Flash? n          Toll Restricted? y

         Trunk Type:
```

12. In the **Auth Code** field, enter **y** to require callers to enter an authorization code in order to tandem a call through an AAR or ARS route pattern. The code will be required even if the facility restriction level of the incoming trunk group is normally sufficient to send the call out over the route pattern.
13. Press **Enter** to save your changes.

Related topics

See "Class of Restriction" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on setting up dialing out restrictions.

See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information on using trunk access codes.

See "Facility Restriction Levels and Traveling Class Marks" *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* and Route Pattern on page 1444 for more information on assigning Facility Restriction Levels.

See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* and Station on page 1491 for more information on using Call Detail Recording (CDR) on station telephones.

See Class of Restriction on page 834 and Station on page 1491 for more information on using Class of Restriction (COR) on station telephones.

See "Remote Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information on allowing authorized callers to access the system from remote locations.

See "Barrier Codes" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* or QSIG to DCS TSC Gateway on page 1427 for information on barrier codes.

See "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, and *Maintenance Commands for Avaya Communication Manager*, 03-300431 for details on administering user profiles and logins.

Setting up Security Violations Notification

This section shows you how to use Security Violations Notification (SVN) to set security-related parameters and to receive notification when established limits are exceeded. You can run reports related to invalid access attempts. You also can disable a login ID or remote access authorization that is associated with a security violation.

When a security violation has occurred, there are steps that you can take to be sure that this same attempt is not successful in the future. See the *Avaya Toll Fraud and Security Handbook*, 555-025-600, for more information.

Instructions

To set up Security Violations Notification:

1. Type **`change system-parameters security`**. Press **Enter**.

   The system displays the **Security-Related System Parameters screen**.

Figure 174: Security-Related System Parameters screen

```
change system-parameters security                                 Page 1 of x
                     SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Barrier Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
 Authorization Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____
```

2. In the **SVN Login Violation Notification Enabled** field, type **y**.

   This sets Security Violations Notification login violation notification.

   Note:

   If you are not using Security Violation Notification for logins, type **n** in the **SVN Login Violation Notification Enabled** field and go to Step 6.

3. In the **Originating Extension** field, type **3040**.

   This becomes the telephone extension for the purpose of originating and identifying SVN referral calls for login security violations.

4. In the **Referral Destination** field, type **attd** to send all calls to the attendant.

   This is the telephone extension that receives the referral call when a security violation occurs.

5. Press **Enter** to save your changes.

   Note:

   If you are not using Remote Access, go to Step 9.

6. (Optional) Type `change remote-access`. Press **Enter**.

   The Remote Access screen appears.

Figure 175: Remote Access screen

```
change remote-access

                                    REMOTE ACCESS
   Remote Access Extension: ________                 Barrier Code Length:____
Authorization Code Required? y            Remote Access Dial Tone: n

     Barrier      COR  TN  COS  Expiration    No. of     Calls
      Code        COR  TN  COS     Date        Calls      Used
   1:________     1__  1_  1__  __/__/__      ______     ______
   2:________     1__  1_  1__  __/__/__      ______     ______
   3:________     1__  1_  1__  __/__/__      ______     ______
   4:________     1__  1_  1__  __/__/__      ______     ______
   5:________     1__  1_  1__  __/__/__      ______     ______
   6:________     1__  1_  1__  __/__/__      ______     ______
   7:________     1__  1_  1__  __/__/__      ______     ______
   8:________     1__  1_  1__  __/__/__      ______     ______
   9:________     1__  1_  1__  __/__/__      ______     ______
  10:_______      1__  1_  1__  __/__/__      ______     ______

Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

7. (Optional) In the **Disable Following A Security Violation** field, type **y**.

   This disables Remote Access following detection of a remote access security violation.

8. (Optional) Press **Enter** to save your changes.

9. Type **`change station`** ***`xxxx`***, where ***`xxxx`*** is the station to be assigned the notification halt button. Press **Enter**.

   The Station screen (page 4) appears.

Figure 176: Station screen

```
add station nnnn                                                Page  4 of  X
                                    STATION

     SITE DATA
          Room: _______                                       Headset? n
          Jack: ___                                           Speaker? n
         Cable: ___                                          Mounting: d
         Floor: _______                                  Cord Length: 0_
      Building: _______                                    Set Color: _____


ABBREVIATED DIALING
     List1: ________           List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                 6:limit-call
 2: call-appr                 7:team     Ext: 5381231          Rg:
 3: call-appr                 8:cfwd-enh Ext:
 4: audix-rec   Ext: 4000     9:cfwd-enh Ext: 5502
 5: release                  10:aux-work RC: 1 Group:

voice-mail Number:
```

10. In the **BUTTON ASSIGNMENTS** section, type one of the following:

- **asvn-halt** — The Authorization Code Security Violation Notification call is activated when an authorization code security violation is detected. This applies only if you are using authorization codes.
- **lsvn-halt** — The Login Security Violation Notification call is activated a referral call when a login security violation is detected.
- **rsvn-halt** — The Remote Access Barrier Code Security Violation Notification call is activated as a call referral. This applies only if you are using Remote Access barrier codes.
- **ssvn-halt** — The Station Code Security Violation Notification call is activated when a station code security violation is detected. This applies only if you are using station codes.

Note:

Any of the above 4 security violations will cause the system to place a notification call to the designated telephone. The call continues to ring until answered. To stop notification of any further violations, press the button associated with the type of violation.

11. Press **Enter** to save your changes.

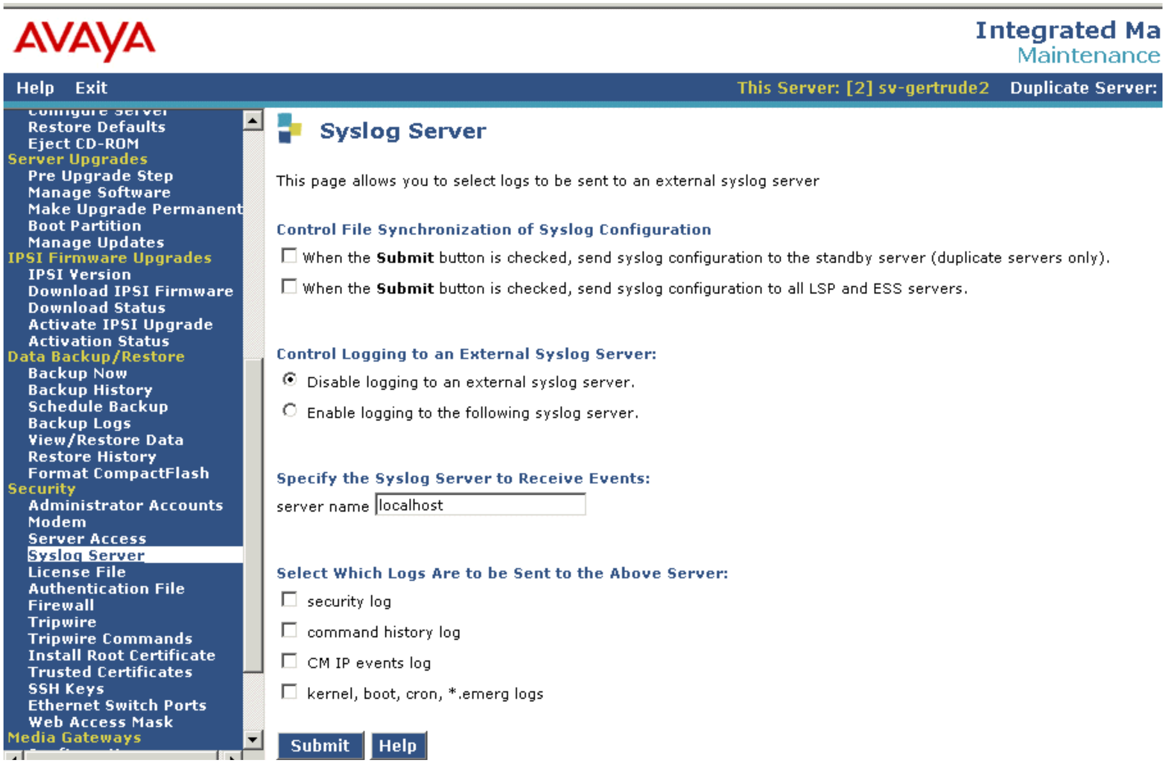
Enhanced security logging

Enhanced security logging increases the granularity of logging of user activity, and allows you to specify an external server or Linux syslog to which to send a copy of system logs. Enhanced security logging consolidates several existing Communication Manager log files, and routes copies of the files to an industry standard external log server or the internal Linux syslog.

SAT activities are logged according to a logging level set by the administrator using the SAT **Logging Levels** screen. For more information on this screen, see Logging Levels on page 1334.

On the Integrated Management Maintenance Web Pages, use the **Syslog Server** web screen to enable or disable the ability to send logs to an external server, and to specify the logs to be sent. Figure 177 shows the **Syslog Server** screen with the options that you can administer for this feature.

Figure 177: Syslog Server web page



Using Station Lock

With the Station Lock feature, users can lock the telephone to prevent others from placing outgoing calls from the telephone.

A user with an analog telephone uses a feature access code (FAC) to lock the telephone. A user with a digital telephone can use a feature access code (FAC) or a feature button to lock the telephone. Station Lock:

- Blocks unauthorized outgoing calls
- Allows outgoing emergency calls
- Allows incoming calls

The feature button lights when the user presses the button to activate Station Lock. Then, when a user attempts to place an outgoing call, the system generates a special dial tone to indicate that the Station Lock feature is active.

If a digital telephone has a feature button for Station Lock, but uses a FAC to activate the feature, the LED lights. The system does not generate the special tone.

If a digital telephone does not have a feature button for Station Lock, and uses a FAC to activate the feature, the system generates the special tone.

Avaya recommends that a user of a digital telephone use a Station Lock button, instead of a FAC, to activate Station Lock.

Any user who knows the system-wide FAC for Station Lock, and the Station Security Code (SSC) of a specific telephone, can lock or unlock the telephone.

A user can also lock or unlock a telephone from a remote location.

The attendant console can lock or unlock other telephones. The attendant console cannot be locked.

Note:

Avaya recommends digital telephones use a Station Lock button rather than a feature access code.

Setting Station Lock

We will set Station Lock to allow authorized users to access the system through a particular station.

Before you start

- Be sure the **Station Lock COR** field on the **Class of Restriction** screen has the COR the user is using to define the calling restrictions.

Instructions

We will set Station Lock on a digital telephone (extension 7262):

1. Type `change station 7262`. Press **Enter**.
2. In the **Security Code** field, enter a security code of up to 8 digits.

   In the **COR** field, leave the default at **1**.
3. In the **BUTTON ASSIGNMENTS** section, type **sta-lock**.
4. Press **Enter** to save your changes.
5. Type `change cor 1`. Press **Enter**.
6. In the **Calling Party Restriction** field, type **none**.

   This means that no calling party restrictions exist on extension 7262.
7. In the **Station Lock COR** field, type **2**.
8. Press **Enter** to save your changes.

9. Type **`change cor 2`**. Press **Enter**.
10. In the **Calling Party Restriction** field, verify it is **outward**.
11. Press **Enter** to save your changes.

    Now when extension 7262 activates Station Lock, calling restrictions are determined by the Station Lock COR, COR 2. Based on the administration of COR 2, extension 7262 is not allowed to call outside the private network. When Station Lock is not active on extension 7262, calling restrictions are determined by the COR administered on the **Station** screen, COR 1. In this example, when extension 7262 is unlocked, calls outside the private network are allowed.

Set Station Lock on an analog, x-mobile, or digital telephone without a Station Lock button (extension 7262 and use a feature access code of 08):

1. Type **`change station 7262`**. Press **Enter**.
2. In the **Security Code** field, enter a security code of up to 8 digits.

   In the **COR** field, leave the default at 1. This means that anyone can call outside on extension 7262.
3. Press **Enter** to save your changes.
4. Type **`change system-parameters features`**. Press **Enter**.

   The **Feature-Related System-Parameters** screen appears.
5. In the **Special Dial Tone** field, type **y** for an audible tone indicating the station is locked.
6. Press **Enter** to save your changes.
7. Type **`change feature-access-codes`**. Press **Enter**.

   The **Feature Access Code (FAC)** screen appears.
8. Move the cursor to the **Station Lock Activation** field.
9. In the **Activation** field, type ***08**.
10. In the **Deactivation** field, type **#08**.
11. Press **Enter** to save your changes.

    Now when a user activates Station Lock, no one can call outside from extension 7262.

Station Lock by time of day

Beginning with Communication Manager 4.0 or later, you can you can also lock stations using a Time of Day (TOD) schedule.

To engage the TOD station lock/unlock you do not have to dial the station lock/unlock FAC, or use **stn-lock** button push.

When the TOD feature activates the automatic station lock, the station uses the Class of Restriction (COR) assigned to the station lock feature for call processing. The COR used is the same as it is for manual station locks.

The TOD lock/unlock feature does not update displays automatically, because the system would have to scan through all stations to find the ones to update.

The TOD Station Lock feature works as follows:

- If the station is equipped with a display, the display will show “Time of Day Station Locked”, if the station invokes a transaction which is denied by the Station Lock COR.
  Whenever the station is within a TOD Lock interval, the user will hear a special dial tone instead of the normal dial tone, if the special dial tone is administered.
- For analog stations or without a display, the user hears a special dial tone. The special dial tone has to be administered and the user hears it when the station is off hook.

After a station is locked by TOD, it can be unlocked from any other station if the Feature Access Code (FAC) or button is used. You have to also know the Station Security Code, and that the **Manual-unlock allowed?** field on the **Time of Day Station Lock Table** screen is set to **y**.

Once a station has been unlocked during a TOD lock interval, the station remains unlocked until next station lock interval becomes effective.

If the station was locked by TOD and by Manual Lock, an unlock procedure will unlock the Manual Lock as well as the TOD Lock (“Manual-unlock allowed?” field on the **Time of Day Station Lock Table** screen is set to **y**).

The TOD feature does not unlock a manually locked station.

Note:
The attendant console cannot be locked by TOD or manual station lock.

Screens for administering Station Lock

Screen name	Purpose	Fields
COR	Administer a Class of Service (COR) that allows the user to activate Station Lock with a feature access code (FAC).	**Station Lock COR**
Feature Access Code (FAC)	Assign a FAC for Station Lock.	**Station Lock Activation**

Screen name	Purpose	Fields
Station	Assign the user a COR that allows the user to activate Station Lock with an FAC.	**COR** **Time of Day Lock Table**
	Assign a **sta-lock** feature button for a user.	Any available button field in the **BUTTON ASSIGNMENTS** area
	Assign a Station Security Code (SSC) for a user.	**Security Code**
Time of Day Station Lock Table	Administer station lock by time of day.	**Table Active?** **Manual Unlock Allowed?** **Time Intervals**

Dealing with Security Violations

When a security violation occurs, there are steps that you can take to be sure that this same attempt is not successful in the future.

Enabling remote access

You might have to enable Remote Access that has been disabled following a security violation, or disabled manually.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **`enable remote-access`**. Press **Enter**.

Disabling remote access

There might be occasions when you have to disable remote access for one of your users because of a security violation.

1. Log in to Avaya Communication Manager using a login ID with the correct permissions.
2. Type **`disable remote-access`**. Press **Enter**.

Chapter 12: Managing Trunks

Tips for working with trunk groups

You'll find detailed procedures for administering specific trunk groups elsewhere in this chapter. However, there's more to working with trunks than just administering trunk groups.

Following a process

Trunking technology is complex. Following a process can prevent mistakes and save you time. To set up new trunks and trunk groups, Avaya recommends following the process below (some steps might not apply to your situation):

1. Install the necessary circuit packs and perform any administration the circuit pack requires.
2. Connect the appropriate ports to your network service provider's trunks.
3. Administer a trunk group to control the operation of the trunks.
4. Assign the ports you're using to the trunk group.
5. For outgoing or 2-way trunks, administer Automatic Route Selection so Avaya Communication Manager knows which outgoing calls to route over this trunk group.
6. Test your new trunk group. Using the trunk access code, place a variety of calls.

This chapter provides instructions for steps 3 and 4 in this process.

Working with your network service provider

Depending on the type of trunk you want to add, the vendor might be your local telephone company, a long distance provider, or some other service provider. Key settings on Avaya Communication Manager must be identical to the same settings on the provider's equipment for your trunks to work. Clear, frequent communication with your provider is essential — especially since some providers might use different terms and acronyms than Avaya does!

Once you decide that you want to add a new trunk, contact your vendor. The vendor should confirm the type of signal you want and provide you with a circuit identification number for the new trunk. Be sure to record any vendor-specific ID numbers or specifications in case you ever have any problems with this trunk.

Keeping records

In addition to recording vendor-specific information such as ID numbers, you should record the following information about every trunk group you have.

The questions you need to answer	The kind of information you need to get
What type of trunk group is it?	You need to know what kind of trunks these are (central office (CO), foreign exchange (FX), etc.) and whether they use any special services (such as T1 digital service). You also need to know what kind of signaling the group uses. For example, you might have a CO trunk group with ground-start signaling running on a robbed-bit T1 service.
Which telephone numbers are associated with each trunk group?	For incoming or two-way trunk groups: 1. What number or numbers do outside callers use to call into your server over this group? 2. What is the destination extension to which this trunk group delivers calls? Does it terminate at an attendant or a voice-mail system? For outgoing trunk groups: ● What extensions can call out over this trunk group?
Is the service from your network service provider sending digits on incoming calls?	Direct Inward Dial and Direct Inward/ Outward Dial trunks send digits to Avaya Communication Manager. Tie trunks can send digits, depending on how they're administered. You need to know: ● How many digits is your service provider sending? ● Are you inserting any digits? What are they? ● Are you absorbing any digits? How many? ● What range of numbers has your service provider assigned you?

Helpful tips for setting common fields

The procedures in this section cover the specific fields you must administer when you create each type of trunk group. Here are some tips for working with common fields that are available for most trunk groups.

- **Dial Access** — Typing **y** in this field allows users to route calls through an outgoing or two-way trunk group by dialing its trunk access code.

  SECURITY ALERT:
  Calls dialed with a trunk access code over Wide Area Telecommunications Service (WATS) trunks are not validated against the ARS Digit Analysis Table, so users can dial anything they wish. For security, you might want to leave the field set to **n** unless you need dial access to test the trunk group.

- **Outgoing Display** — Typing **y** in this field allows display telephones to show the name and group number of the trunk group used for an outgoing call. This information might be useful to you when you're trying to diagnose trunking problems.
- **Queue Length** — Don't create a queue for two-way loop-start trunks, or you might have a problem with glare (the interference that happens when a two-way trunk is seized simultaneously at both ends).
- **Trunk Type** — Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end. Try to use loop-start signaling only for one-way trunks.

Related topics

See the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207, for information on the types of circuit packs available and their capacities.

See your server's Installation manual for circuit-pack installation instructions.

See Modifying Call Routing on page 319 for detailed information on Automatic Route Selection.

Adding a CO, FX, or WATS trunk group

Basic administration for Central Office (CO), Foreign Exchange (FX), and WATS trunk groups is identical, so we've combined instructions for all 3 in the following procedure. In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type

CAUTION:
Use the list above as a starting point and talk to your service provider. Depending on your particular application, you might need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Instructions

As an example, we will set up a two-way CO trunk group that carries voice and voice-grade data only. Incoming calls terminate to an attendant during business hours and to a night service destination the rest of the time

To add the new CO trunk-group:

1. Type **`add trunk-group next`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

Figure 178: Trunk Group screen

```
add trunk-group next                                              Page   1 of  x
                                TRUNK GROUP

Group Number: 1                  Group Type: co           CDR Reports: y
  Group Name: OUTSIDE CALL              COR: 1        TN: 1        TAC:
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0          Country: 1              Incoming Destination:
   Comm Type: voice                Auth Code? n   Digit Absorption List:
    Prefix-1? y                  Trunk Flash? n         Toll Restricted? y

         Trunk Type:
```

2. In the **Group Type** field, type **co**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can make and receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

6. In the **Direction** field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. In the **Night Service** field, type **1234**.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the **Incoming Destination** field, type **attd**.

   This field assigns an extension to which incoming calls are routed during business hours. By entering attd in this field, incoming calls go to the attendant and the system treats the calls as Listed Directory Number calls.

9. In the **Comm Type** field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data.

10. In the **Trunk Type** field, type **ground-start**.

    This field tells the system what kind of signaling to use on this trunk group. To prevent glare, Avaya recommends ground start signaling for most two-way CO, FX, and WATS trunk groups.

11. Press **Next Page** until you find the **Outgoing Dial Type** field.

12. In the **Outgoing Dial Type** field, type **tone**.

    This field tells Communication Manager how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both dual-tone multifrequency (DTMF) and rotary signals, so Avaya recommends that you always put tone in this field.

13. In the **Trunk Termination** field, type **rc**.

    Use **rc** in this field when the distance to the central office or the server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

14. Press **Enter** to save your changes.

Now you are ready to add trunks to this trunk group. See Adding trunks to a trunk group on page 497.

Adding a DID trunk group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. For Direct Inward Dialing (DID) trunk groups, settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type
- Expected Digits (only if the digits your provider sends *do not* match your dial plan)

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you might need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Tip:

In the **DID/Tie/ISDN Intercept Treatment** field on the **Feature-Related System Parameters** screen, enter **attd**. Incoming calls to invalid extensions will be routed to the attendant.

Instructions

To add the new Direct Inward Dialing (DID) trunk-group:

1. Type **`add trunk-group next`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

2. In the **Group Type** field, type **did**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Incoming calls**.

   You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This code identifies the trunk group on CDR reports.

6. In the **Trunk Type** field, type **wink-start**.

   This field tells the system what kind of signaling to use on this trunk group. In most situations, use wink start for DID trunks to minimize the chance of losing any of the incoming digit string.

7. In the **Incoming Dial Type** field, type **tone**.

   This field tells Communication Manager how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

8. In the **Trunk Termination** field, type **rc**.

   Use **rc** in this field when the distance to the central office or the server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

9. Press **Enter** to save your changes.

Now you're ready to add trunks to this trunk group. See Adding trunks to a trunk group on page 497.

Related topics

See Inserting and absorbing digits on page 501 for instructions on matching modifying incoming digit strings to match your dial plan.

Adding a PCOL trunk group

In most cases, when administering Personal Central Office Line (PCOL) trunk groups, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Trunk Type
- Trunk Direction

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you might need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Instructions

As an example, we will set up a new PCOL group and administer the group as a CO trunk for two-way voice traffic.

To add the new PCOL group:

1. Type **`add personal-co-line next`**. Press **Enter**.

   The **Personal CO Line Group screen** appears.

Figure 179: Personal CO Line Group screen

```
add personal-CO-line                                                Page 1 of x
                            PERSONAL CO LINE GROUP

 Group Number: __                    Group Type:_____________  CDR Reports: _
   Group Name: _________________                                       TAC: ____
Security Code: ____               Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________            Trunk Direction: ________
                 Trunk Port: _______          Disconnect Timing(msec): ____
                 Trunk Name: __________             Trunk Termination: ______
         Outgoing Dial Type: _________                Analog Loss Group: ___
                   Prefix-1? _                       Digital Loss Group: ___
Disconnect Supervision - In? _                          Call Still Held? _
 Answer Supervision Timeout: ___          Receive Answer Supervision? _
                 Trunk Gain: ____                               Country: __
          Charge Conversion: _____             DS1 Echo Cancellation: _
              Decimal Point: ______
            Currency Symbol: ___
                Charge Type: _______
```

2. In the **Group Type** field, type **co**.

   This field specifies the kind of trunk group you're creating. PCOL groups can be administered as CO, FX, or WATS trunks.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field. (You might want to put the telephone number here that's assigned to this trunk.)

4. In the **TAC** field, type **111**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

5. In the **Trunk Type** field, type **ground start**.

   This field tells the system what kind of signaling to use on this trunk group. To prevent glare, Avaya recommends ground start signaling for most two-way CO, FX, and WATS trunk groups.

6. In the **Trunk Port** field, type **01D1901**.

   This is the port to which the trunk is connected.

7. In the **Trunk Termination** field, type **rc**.

   Use **rc** in this field when the distance to the central office or the server at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

8. In the **Outgoing Dial Type** field, type **tone**.

   This field tells Communication Manager how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

9. Press **Enter** to save your changes.

You assign telephones to a PCOL group by administering a CO Line button on each telephone. Once assigned, the **Assigned Members** page of the **Personal CO Line Group** screen displays member telephones:

Figure 180: Personal CO Line Group screen

```
add personal-CO-line                                                Page 2 of x
                            PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

        Ext    Name                            Ext    Name

     1:                                     9:
     2:                                    10:
     3:                                    11:
     4:                                    12:
     5:                                    13:
     6:                                    14:
     7:                                    15:
     8:                                    16:
```

More information

Call Detail Recording

Call detail recording (CDR) can be activated for calls on a personal CO line, but the CDR record does not specifically identify the call as PCOL. Calls over personal CO lines can, however, be identified by the trunk access code used on the call. The call is recorded to the extension number assigned to the telephone where the call was originated or answered.

Restrictions

- Abbreviated Dialing can be used with a personal CO line, but the accessed lists are associated with the individual telephones.
- Auto Hold and Leave Word Calling do not work with calls on a personal CO line.
- Send All Calls cannot be activated for a personal CO line.
- INTUITY AUDIX cannot be in the coverage path of a PCOL group.
- Only telephones in the same PCOL group can bridge onto calls on the personal CO line. If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call.
- When a user puts a call on hold on a personal CO line, the status lamp associated with the PCOL button does not track the busy/idle status of the line.

Adding a Tie or Access trunk group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings (or the setting on the far-end server, if this is a private network trunk group):

- Direction
- Comm Type
- Trunk Type

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you might need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Tip:

In the **DID/Tie/ISDN Intercept Treatment** field on the **Feature-Related System Parameters** screen, enter **attd**. Incoming calls to invalid extensions will be routed to the attendant.

Instructions

As an example, we will add a two-way tie trunk group that supports voice and voice-grade data.

To add the new tie trunk-group:

1. Type **`add trunk-group next`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

2. In the **Group Type** field, type **tie**.

   This field specifies the kind of trunk group you're creating.

3. In the **Group Name** field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the **Outgoing Display?** field to **y**. You can type any name up to 27 characters long in this field.

4. In the **COR** field, type **85**.

   This field controls which users can make or receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the **TAC** field, type **105**.

   This field defines a unique code users can dial to access this trunk group.

6. In the **Direction** field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. In the **Night Service** field, type **1234**.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the **Comm Type** field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data. If you're administering a T1 connection in North America, type **rbavd** in this field.

9. In the **Trunk Type** field, type **wink/wink**.

   This field tells the system what kind of signaling to use on this trunk group. Because we're receiving and sending digits over this trunk group, we're using wink/wink signaling to minimize the chance of losing part of the digit string in either direction.

10. Type **tone** in both the **Outgoing Dial Type** and **Incoming Dial Type** fields.

    These fields tell Communication Manager how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

11. Press **Enter** to save your changes.

Now you're ready to add trunks to this trunk group. See Adding trunks to a trunk group on page 497.

Adding a DIOD trunk group

Administration for Direct Inward and Outward Dialing (DIOD) trunk groups varies from country to country. See your local Avaya representative for more information. Remember that the central office serving your switching system might be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

If you are using Incoming Caller ID (ICLID) on analog trunks connected to a DIOD Central Office trunk circuit pack, DO NOT put these trunks in an outgoing AAR or ARS route pattern. Since the loop-start trunks supported on the DIOD Central Office trunk circuit pack do not provide answer supervision, the potential for toll fraud exists.

Setting up digital trunks

Any of the common trunks, except for PCOL trunks, can be analog or digital. (PCOL trunks can only be analog.) Administering a digital trunk group is very similar to administering its analog counterpart, but digital trunks must connect to a DS1 circuit pack and this circuit pack must be administered separately. The example in this section shows you how to do this.

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information.

Your settings in the following fields *must* match your provider's settings:

- Bit Rate
- Line Coding (unless you're using a channel service unit to convert between your line coding method and your provider's)
- Framing Mode
- Signaling Mode
- Interface Companding

CAUTION:
Use the list above as a starting point and talk to your service provider. Depending on your particular application, you might need to coordinate additional administration with your service provider.

Before you start

Assign the DS1 circuit pack before you administer the members of the associated trunk groups.

CAUTION:

If enhanced DS1 administration is not enabled, you cannot make changes to the **DS1 Circuit Pack** screen before you remove related member translations of all trunks from the trunk group. See Enhanced DS1 administration on page 494.

Before you can administer a digital trunk group, you must have one or more circuit packs that support DS1 with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Instructions

The following example shows a DS1 circuit pack configured for T1 service. The circuit pack is supporting a two-way CO trunk group that carries only voice and voice-grade data.

To configure a new DS1 circuit pack:

1. Type `add ds1 07A19`. Press **Enter**.

   The **DS1 Circuit Pack screen** appears. You must enter a specific port address for the circuit pack.

Figure 181: DS1 Circuit Pack screen

```
add ds1 nnnn                                                     Page 1 of x
                                DS1 CIRCUIT PACK

            Location: _____                             Name: _______________
            Bit Rate: _____                      Line Coding: ____
   Line Compensation: _                         Framing Mode: ___
      Signaling Mode: ___________                  D-Channel: __
             Connect: _________                    Interface: ___________
        Interconnect: ___                      Peer Protocol: ________
                                            Country Protocol: ____
                                            Protocol Version: _
Interface Companding: _____                             CRC? _
           Idle Code: ________
                               DCP/Analog Bearer Capability: ________
                                            T303  Timer(sec):

   MMI Cabling Board: _____  MMI Interface: ESM

MAINTENANCE PARAMETERS

      Slip Detection? _                    Near-end CSU Type: ________
                                      Block Progress Indicator? n
```

2. In the **Name** field, type **two-way CO**.

   Use this name to record useful information such as the type of trunk group associated with this circuit pack or its destination.

3. In the **Bit Rate** field, type **1.544**.

   This is the standard for T1 lines.

4. In the **Line Coding** field, type **b8zs**.

   Avaya recommends you use **b8zs** whenever your service provider supports it. Since this trunk group only carries voice traffic, you could also use **ami-zcs** without a problem.

5. In the **Framing Mode** field, type **esf**.

   Avaya recommends you use **esf** whenever your service provider supports it.

6. In the **Signaling Mode** field, type **robbed-bit**.
7. In the **Interface Companding** field, type **mulaw**.

   This is the standard for T1 lines in North America.

8. Press **Enter** to save your changes.

More information

T1 recommended settings

The table below shows recommended settings for standard T1 connections to your local exchange carrier.

Field	Value	Notes
Line Coding	**b8zs**	Use **ami-zcs** if **b8zs** is not available.
Signaling Mode	**robbed-bit**	Robbed-bit signaling gives you 56K bandwidth per channel. If you need a 64K clear channel for applications like asynchronous data transmission or remote administration access, use common channel signaling.
Framing	**esf**	Use **d4** if **esf** is not available.

If you use b8zs line coding and esf framing, it will be easier to upgrade your T1 facility to ISDN should you want to. You can upgrade without reconfiguring external channel service units, and your service provider won't have to reconfigure your network connection.

E1 recommended settings

DS1 administration for E1 service varies from country to country. See your local Avaya technical support representative for more information.

Note:

Remember that the central office serving your switching system might be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

Enhanced DS1 administration

Normally, you can't change the **DS1 Circuit Pack** screen unless you remove all related trunks from their trunk group. However, if the **DS1 MSP** field on the **System-Parameters Customer-Options (Optional Features)** screen is **y**, and you are assigned the associated login permissions, you can change some of the fields on the **DS1 Circuit Pack** screen without removing the related trunks from their trunk group.

If you busy out the DS1 circuit pack, you can change the following fields: **CRC**, **Connect**, **Country Protocol**, **Framing Mode**, **Interface**, **Interconnect**, **Line Coding**, and **Protocol Version**. After changing these fields, you might also have to change and resubmit associated screens.

Matching field settings on different screens

For enhanced DS1 administration, some field values on the **DS1 Circuit Pack** screen must be consistent with those on other screens as shown in the table below. If you change field values on the **DS1 Circuit Pack** screen, you must change the related fields on the other screens and resubmit them.

DS1 Circuit Pack field	Affected screens
Line Coding	Route Pattern Access Endpoint Signaling Group Tone Generation
Connect	Signaling Group
Protocol Version	Signaling Group
Interface	Signaling Group
Interconnect	Tone Generation
Country Protocol	Signaling Group Tone Generation

Specific combinations of settings for some of these fields are shown below.

ITC, Bit Rate, and Line Coding

The **ITC (Information Transfer Capability)** field appears on the **Route Pattern** screen, **Trunk Group** screen, and **Access Endpoint** screen. The **Line Coding** and the **Bit Rate** fields appear on the **DS1 Circuit Pack** screen. The settings for these fields on all the screens must be coordinated as shown in the following tables.

ITC field	Bit Rate	Line Coding field
restricted	1.544 Mbps	**ami-zcs**
	2.048 Mbps	**ami-basic**
unrestricted	1.544 Mbps	**b8zs**
	2.048 Mbps	**hdb3**

Interconnect and corresponding Group Type entries.

The **Interconnect** field appears on the **DS1 Circuit Pack** screen. The **Group Type** field appears on the **Trunk Group** screen. Set these fields as shown in the following table.

Interconnect field	Group Type field
co	**co**, **did**, **diod**, **fx**, or **wats**
pbx	**access**, **aplt**, **isdn-pri**, **tandem**, or **tie**

Related topics

See DS1 Circuit Pack on page 945 for information on administering DS1 service.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205, for detailed information on DS1 service.

Adding trunks to a trunk group

Use this procedure to add new trunks or to change the assignment of existing trunks. To change the assignment of existing trunks, remove them from their current trunk group and add them to the new group.

Before you start

You must add a trunk group before you can assign and administer individual trunks. To add a new trunk group, see the instructions in this chapter for the type of group you want to add.

Instructions

As an example, we will assign 5 trunks to a new tie trunk group, trunk group 5. We'll use ports on several circuit packs for members of this group.

To assign trunks to trunk group 5:

1. Type **`change trunk-group 5`**. Press **Enter**.
2. Click **Next Page** to move to the **Group Member Assignments screen**.

   Some of the fields on this screen won't appear for every trunk group.

Figure 182: Trunk Group screen

```
add trunk-group next                                          Page x of y
                                 TRUNK GROUP
                                        Administered Members(min/max): xxx/yyy
                                           Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port      Code  Sfx  Name          Night          Mode      Type    Ans Delay
    1: _______              ___________   _____          ________  _______ ____
    2: _______              ___________   _____          ________  _______ ____
    3: _______              ___________   _____          ________  _______ ____
    4: _______              ___________   _____          ________  _______ ____
    5: _______              ___________   _____          ________  _______ ____
    6: _______              ___________   _____          ________  _______ ____
    7: _______              ___________   _____          ________  _______ ____
    8: _______              ___________   _____          ________  _______ ____
    9: _______              ___________   _____          ________  _______ ____
   10: _______              ___________   _____          ________  _______ ____
   11: _______              ___________   _____          ________  _______ ____
   12: _______              ___________   _____          ________  _______ ____
   13: _______              ___________   _____          ________  _______ ____
   14: _______              ___________   _____          ________  _______ ____
   15: _______              ___________   _____          ________  _______ ____
```

3. In the **Port** field in row 1, type **1B1501**.

   This field assigns the first member of the trunk group to a port on a circuit pack.

4. In the **Name** field in row 1, type **5211**.

   This is the extension assigned to this trunk. In general, type the circuit ID or telephone number for each trunk in this field. The information is helpful for tracking your system or troubleshooting problems. Update these fields whenever the information changes.

5. In the **Mode** field, type **e&m**.

   CAUTION:

   An entry in this field is only required for some circuit packs. Dip switch settings on the circuit pack control the signalling mode used on the trunk group, so the entry in the Mode field must correspond to the actual setting on the circuit pack.

6. In the **Type** field, type **t1-comp**.

   An entry in this field is only required for some circuit packs.

7. Repeat steps 3 to 6, as appropriate, for the remaining trunks.

   Notice that you can assign trunks in the same trunk group to ports on different circuit packs.

8. Press **Enter** to save your changes.

Removing trunk groups

There's more to removing a trunk group than just executing the **`remove trunk-group`** command. If you're using Automatic Route Selection (ARS), you must remove an outgoing or two-way trunk group from any route patterns that use it. If you've administered **Trunk-Group Night Service** buttons for the trunk group on any telephones, those buttons must be removed or assigned to another trunk group.

Instructions

As an example, we will remove trunk group 5. This two-way group is used in ARS route pattern 2. In addition, a **Trunk-Group Night Service** button on extension 8410 points to this group.

To remove trunk group 5:

1. In the **Route Pattern** screen for route pattern 2, clear the entries for trunk group 5.

   If you're replacing trunk group 5 with another trunk group, just type the information for the new trunk group over the old entries. Remember to press **Enter** to save your changes.

2. In the **Station** screen for extension 8410, clear the entry in the **BUTTON ASSIGNMENTS** field for the **Trunk-Group Night Service** button.

   Remember to press **Enter** to save your changes.

3. In the **Group Member Assignments screen** for trunk group 5, remove all member trunks from the group.

   See Adding trunks to a trunk group on page 497 for instructions.

4. Type **`remove trunk-group 5`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears.

5. Press **Enter** to remove the trunk group.

Resetting trunks

To "reset" a trunk, use the **`busyout`** command followed by the **`release`** command, both executed in a SAT window. You can run these commands on a board, a port, a trunk group, or an individual trunk. The availability of these commands depends on your login permissions.

Note:

These commands can tear calls down, so use them with great caution. Contact your Avaya technical representative for details.

To reset a trunk group:

1. Type **`busyout trunk`** ***n***, where ***n*** is the number of the trunk group.
2. Type **`release trunk`** ***n***.

   The trunk group is reset. (Example: **`busyout trunk 43`** followed by **`release trunk 43`**.)

To reset a trunk member:

1. Type **`busyout trunk`** ***n/x***, where ***n*** is the number of the trunk, and ***x*** is the trunk group member.
2. Type **`release trunk`** ***n/x***.

   The trunk group member is reset. (Example: **`busyout trunk 43/1`** followed by **`release trunk 43/1`**. Another example operation for an ISDN trunk is **`test trunk 43`**.)

Inserting and absorbing digits

Use this procedure to modify the incoming digit string on DID and tie trunks by inserting (adding) or absorbing (deleting) digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Instructions

As an example, let us say you have a DID trunk group. It's group number is 5. Your service provider can only send 4 digits, but your dial plan defines 5-digit extensions beginning with 6:

1. Type **`change trunk-group 5`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears.
2. In the **Digit Treatment** field, type **insertion**.

   This field tells Communication Manager to add digits to the incoming digit string. These digits are always added at the beginning of the string.
3. In the **Digits** field, type **6**.

   For insertion, this field defines the specific digits to insert. Communication Manager will add a "6" to the front of the digit strings delivered with incoming calls. For example, if the central office delivers the string "4444," Avaya Communication Manager will change it to "64444," an extension that fits your dial plan.
4. In the **Expected Digits** field, type **4**.

   This field tells Communication Manager how many digits the central office sends.

   Note:

   The **Expected Digits** field does not appear on the screen for tie trunk groups.
5. Press **Enter** to save your changes.

If your service provider sends 7 digits but you only need 5, you need to absorb the first 2 digits in the digit string. To absorb digits:

1. Type **`change trunk-group 5`**. Press **Enter**.

   The **Trunk Group screen - page 1** appears.
2. In the **Digit Treatment** field, type **absorption**.

   This field tells Communication Manager to remove digits from the incoming digit string. These digits are always removed from the beginning of the string.

3. In the **Digits** field, type **2**.

   For absorption, this field defines *how many* digits will be absorbed. Communication Manager will remove the first 2 digits from the digit strings delivered with incoming calls. For example, if the central office delivers the string "556-4444," Avaya Communication Manager will change it to "64444," an extension that fits your dial plan.

4. In the **Expected Digits** field, type **7**.

   This field tells Communication Manager how many digits the central office sends.

   Note:

   The **Expected Digits** field does not appear on the screen for tie trunk groups.

5. Press **Enter** to save your changes.

Related topics

See Adding a DID trunk group on page 483 for instructions on administering a DID trunk group.

See Adding a Tie or Access trunk group on page 489 for instructions on administering a tie trunk group.

Administering trunks for Listed Directory Numbers

Listed directory numbers (LDN) are the telephone numbers given for an organization in public telephone directories. You can administer Avaya Communication Manager so that calls to different listed directory numbers go to the same attendant group. How you administer your system for LDN calls depends on whether the calls are coming in over DID and tie trunks or over CO and FX trunks.

Instructions

As an example, let us say that one attendant group answers calls for 3 different businesses, each with its own listed directory number:

- Company A — 855-2020
- Company B — 855-1000
- Company C — 855-1111

DID trunks and some tie trunks transmit part or all of the dialed digit string to Communication Manager. If you want these calls to different numbers to go to one attendant group, you must identify those numbers for Communication Manager on the **Listed Directory Numbers** screen.

We will take the 3 businesses listed above as an example. We will assume your server receives 4 digits from the central office on a DID trunk group and that you're not using Tenant Partitioning. To make these calls to different listed directory numbers terminate to your attendant group:

1. Type **`change listed-directory-numbers`**. Press **Enter**.

   The **Listed Directory Numbers screen** appears.

Figure 183: Listed Directory Numbers screen

```
change listed-directory-number                                Page   1 of   x
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext      Name                         TN
      1:
      2:                                       1
      3:                                       1
      4:                                       1
      5:                                       1
      6:                                       1
      7:                                       1
      8:                                       1
```

2. In the **Ext 1** field, type **2020**.

   This is the LDN for Company A.

3. In the **Name** field, type **Company A**.

   This name will appear on the console display so the attendant knows which business the call is for and how to answer it.

4. Repeat steps 2 and 3 for the other two businesses.

   You can enter up to 20 different listed directory numbers on this screen.

5. Press **Enter** to save your changes.

To make LDN calls over a CO or FX trunk group terminate to an attendant group, you must type attd in the **Incoming Destination** field on the **Trunk Group** screen for that group.

When you use the **Listed Directory Number** screen to assign some extensions to the attendant group, or when you enter **attd** in the **Incoming Destination** field on the **Trunk Group** screen for CO or FX trunks, Avaya Communication Manager treats these calls as LDN calls.

Related topics

See Listed Directory Numbers on page 1319 for detailed information about this feature.

Administering Answer Detection

Use this procedure to administer an outgoing or two-way trunk group for network answer supervision or answer supervision by timeout. If your network supplies answer supervision to a trunk group, you can administer Avaya Communication Manager to recognize and respond to that signal. If your network does not supply answer supervision, you can set a timer for all calls on that group. When the timer expires, Communication Manager assumes the call has been answered and call detail recording starts (if you are using CDR).

For information about answer detection by call classification, contact your Avaya technical support representative or see "Answer Detection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205 for an introduction.

Before you start

Determine whether the trunk group receives answer supervision from your service provider or private network. For example, most loop-start CO, FX, and WATS trunks do not provide answer supervision.

Instructions

As an example, we will administer trunk group 5 for both types of answer detection.

To administer trunk group 5 for answer supervision from the network:

1. On the **Trunk Group** screen for group 5, type **y** in the **Receive Answer Supervision** field.
2. Press **Enter** to save your change.

Now we will administer answer supervision by timeout. We'll set the timer to 15 seconds. To administer trunk group 5 for answer supervision by timeout:

1. On the **Trunk Group** screen for group 5, type **15** in the **Answer Supervision Timeout** field.
2. Press **Enter** to save your change.

Related topics

See "Answer Detection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for detailed information about this feature.

Administering ISDN trunk groups

Integrated Services Digital Network (ISDN) trunk groups support the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group can contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they can also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

When an ISDN trunk connects two servers or switches, set the trunk options identically at both ends of the connection, with the exception of the **Trunk Hunt** fields. When ISDN-PRI interfaces are used, it is acceptable for both ends to have the **Trunk Hunt** fields administered as **cyclical**, but if one end is administered as **ascend**, the other end must be administered as **descend**. This helps avoid the possibility of glare conditions. When ISDN-BRI is used, the **Trunk Hunt** field has to be **cyclical**.

Hardware requirements

ISDN-BRI interfaces are supported by the TN2185 Trunk-side BRI circuit pack (for implementing the user side of the BRI interface) and by the TN556B or TN556C ISDN-BRI Line circuit pack or the TN2198 ISDN BRI (U-LT) Line circuit pack (for the network side of the BRI interface). The TN2464 circuit supports T1 and E1 digital facilities.

ISDN-PRI interfaces are supported by the TN767 circuit pack (for assignment of a T1 signaling link and up to 24 ISDN-PRI trunk group members), or the TN464C or later circuit pack (for assignment of a T1 or E1 signaling link and up to 24 or 31 ISDN-PRI trunk group members, respectively). The TN2464 and TN2207 circuit pack can also be used with ISDN-PRI.

- The D-channel for ISDN-PRI interfaces switches through either the TN765 Processor Interface (PI) circuit pack or the TN778 Packet Control (PACCON) circuit pack. The D-channel for ISDN-BRI interfaces only switches through the TN778 Packet Control (PACCON) circuit pack.

  Note:

  You cannot use the TN765 circuit pack with ISDN-BRI interfaces.

- A TN780 or TN2182 Tone Clock circuit pack provides synchronization for the DS1 circuit pack.

  Note:

  The TN767 cannot be used to carry the D-channel if either the TN778 (PACCON) or TN1655 (PKTINT) circuit packs are used to switch the D-channel. However, in these circumstances, the TN767 can be used for NFAS interfaces carrying only B-channels.

Screens used to administer ISDN trunk groups

Screen	Field
Feature-Related System Parameters	**Send Non-ISDN Trunk Group Name as Connected Name?** **Display Connected Name/Number for ISDN DCS Calls?**
Incoming Call Handling Treatment	**All**
Numbering - Public/Unknown Format	**All**
System Parameters Customer-Options (Optional Features)	**Version** **ISDN-BRI Trunks** **ISDN-PRI** **QSIG Optional Features**
Synchronization Plan	**All**
Trunk Group (ISDN)	**All**
ISDN-BRI Circuit Pack screen (if using ISDN-BRI interfaces) or **DS1 Circuit Pack screen** (if using ISDN-PRI interfaces)	**All** **All**
	1 of 2

Screen	Field
ISDN Numbering - Private	**All**
Route Pattern	**All**
Hunt Groups	**ISDN Caller Display**
Signaling Group (if using ISDN-PRI interfaces)	**All**
Terminating Extension Group	**ISDN Caller Display**
	2 of 2

Table Notes

System Parameters Customer-Options (Optional Features) — The **ISDN-BRI Trunks** or **ISDN-PRI** fields must be set to **y**. For a TN778 and if using ISDN-PRI interfaces, the **PRI Over PACCON** field must be set to **y**. These features are provided via license file. To enable these features, contact your Avaya representative.

- **QSIG Optional Features** fields can be enabled to allow appropriate administration for Supplementary Service Protocol.
- **Feature-Related System-Parameters** — Set the **Send Non-ISDN Trunk Group Name as Connected Name** and **Display Connected Name/Number for ISDN DCS Calls** fields.
- **ISDN-BRI Trunk Circuit Pack** — This screen is required if using ISDN-BRI trunk interfaces. Assign all fields as required.
- **DS1 Circuit Pack** — This screen is required if using ISDN-PRI interfaces.
  - DS1 (T1) Circuit Pack

    Assign all fields as required. For **Facility Associated Signaling**, up to 23 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. The 24th port is used as a signaling channel. For **Non-Facility Associated Signaling**, all 24 ports can be used on certain DS1 circuit packs. The D-channel signaling function for these packs must be provided by a designated DS1 pack on its 24th channel.
  - E1 Circuit Pack

    Assign all fields as required. For **Facility Associated Signaling**, up to 30 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. Port number 16 is used as a signaling channel.
- **Maintenance-Related System-Parameters** — Use this screen only for a TN778. Set the **Packet Bus Maint** field to **y**.
- **ISDN Trunk Group** — Enter information in all the fields except the trunk group members. When using ISDN-PRI interfaces, enter the members after you establish the signaling links.

- **Signaling Group** — This screen is required if ISDN-PRI interfaces are used. Complete all fields. This screen identifies groups of ISDN-PRI DS1 interface B-channels for which a given D-channel (or D-channel pair) will carry the associated signaling information (supports the Facility and Non-Facility Associated Signaling feature). Each DS1 board that is required to have a D-channel must be in a different signaling group by itself (unless D-channel backup is needed, in which case a second DS1 is administered as a backup D-channel). You are not required to select a channel for a trunk group, but if you do, you must have already defined the trunk group as type ISDN.

  Note:

  The following three screens, **Processor Interface Data Module**, **Communication Interface Links**, and **Communication Processor Channel Assignment** are used only to support the ISDN-PRI interfaces using PI TN765.

- **Processor Interface Data Module** — Use this screen only for a TN765. Assign up to 8 interface links using 8 **Processor Interface Data Module** screens for multi-carrier cabinet systems, and up to 4 links for single-carrier cabinet systems. One **Processor Interface Data Module** screen must be completed for each interface link to be assigned.
- **Communication Interface Links** — Use this screen only for a TN765. Assign link numbers 01 to 08 for a multi-carrier cabinet system or links 01 to 04 for a single-carrier cabinet system as required. When first administering this screen for ISDN in Avaya Communication Manager, do not administer the **Enable** field.
- **Communication Processor Channel Assignment** — Use this screen only for a TN765. Enter assigned link numbers and assign associated channel numbers to each link. Complete all fields of the screen as required. When first administering this screen for ISDN in Avaya Communication Manager, you need to:
  - First, administer the **Interface Links** screen, except the **Enable** field.
  - Second, administer the **ISDN** fields on the **Processor Channel** screen.
  - Last, go back to the **Interface Links** screen and administer the **Enable** field.
- **ISDN Numbering - Public/Unknown** — Complete all fields. This screen supports the ISDN Call Identification Display.
- **ISDN Numbering - Private** — Complete all fields. This screen supports the ISDN Call Identification Display.
- **Routing Pattern** — Complete all fields including the **Supplemental ISDN Routing Information** fields as required.
- **Hunt Group** — Complete the **ISDN Caller Display** field by entering either **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).
- **Terminating Extension Group** — Complete the **ISDN Caller Display** field by entering either **grp-name** or **mbr-name** to specify whether the group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).

- **Synchronization Plan** — Assigns primary and secondary external synchronization sources for the ISDN-BRI Trunk or DS1 circuit pack. Complete all screen fields as required.

  Note:

  ISDN-BRI and ISDN-PRI interfaces cannot be mixed in the same trunk group. Therefore, consider the following:

  - The earliest trunk member (the lowest numbered one) administered is considered correct.
  - If an offending member is subsequently found (meaning the first member was BRI and a later member was PRI, or vice versa), the cursor positions on the offending member, and the following error message appears: **You cannot mix BRI and PRI ports in the same trunk group.**

Administering displays for QSIG trunks

For proper transmission of QSIG name data for display, administer the following settings:

- **Trunk Group** screen
  - **Group Type**: **ISDN**
  - **Character Set for QSIG Names**: **iso8859-1**
  - **Outgoing Display**: **y**
  - **Send Calling Number**: **y**
  - **Send Name**: **y**
- **Signaling Group** screen
  - **Supplementary Service Protocol**: **b**
- **System-Parameters Country-Options** screen
  - **Display Character Set**: **Roman**

Chapter 13: Managing Announcements

An announcement is a recorded message a caller can hear while the call is in a queue, or if a call receives intercept treatment for some reason. An announcement is often used in conjunction with music.

The source for announcements can be either integrated or external.

- Integrated announcements reside on a circuit pack in the carrier, such as the TN2501AP circuit pack, or embedded in a media gateway processor board (called a “v VAL source” throughout this chapter).
- External announcements are stored on a separate piece of equipment (called an “adjunct”), and played back from the adjunct equipment.

This chapter uses the term “announcement source” to mean either integrated or external sources for announcements.

Getting Started with the VAL or G700 Virtual VAL

Before you can use the capabilities of the VAL or G700 v VAL announcement circuit pack, it must be properly installed and configured. These instructions are contained in other documents in the Avaya Communication Manager documentation library.

- For a complete description of Announcement information and procedures, see the “Announcements” feature in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.
- For a complete description of the related Locally Sourced Announcement feature, see the “Locally Sourced Announcements and Music” feature in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.
- For more information about these and other tasks related to using the VAL, see the documents listed in the following table.

.

Task	Information source
Installing the VAL circuit pack Administering IP Connections Adding IP Routes Testing the IP Connections	*Made Easy Tool for DEFINITY Server Configurations* *Installation, Upgrades and Additions for the Avaya CMC1 Media Gateway*
	1 of 3

Task	Information source
Installing v VAL for a G700 Media Gateway using the **Media-Gateway** screen and the **`enable announcement`** command Administering IP Connections Adding IP Routes Testing the IP Connections **Note:** G700 Media Gateway embedded VAL announcements (v VAL) must have the gateway(s) that will provide announcements enabled in order for announcement extensions assigned to that gateway to be played.	Each G700 Media Gateway that will be used to provide announcements through the embedded VAL circuitry on the Gateway processor circuit pack must be assigned on the **Media-Gateway** screen and enabled using the **`enable announcements`** command before announcements can be recorded using the telephone or played from that gateway. **Note:** For more information about the **Media-Gateway** screen, and for a description of commands, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300191. Announcements can be administered to a gateway and files can be FTPed to that gateway even though it is not enabled. However, the G700 Media Gateway first must be assigned on the **Media-Gateway** screen so as to be used for gateway announcements. Each G700 Media Gateway when enabled is counted as a VAL circuit pack towards the system limit of either 1 VAL circuit pack (if the **VAL Maximum Capacity** field is **n**) or 10 circuit packs (for the Avaya S8XXX Servers) if the **VAL Maximum Capacity** field is **y**. First the G700 Media Gateway must have the **V9** field assigned to **gateway-announcements** on the **Media-Gateway** screen before the G700 embedded VAL (v VAL) can be enabled. Then the G700 Media Gateway embedded VAL is enabled using the **`enable announcement-board gggV9`** command (where **ggg** is the gateway number assigned on the **Media-Gateway** screen).
	The G700 Media Gateway embedded VAL also can be disabled using the **`disable announcement-board ggV9`** command. This removes that gateway from the VAL circuit pack count but announcements already assigned and recorded/FTPed on that circuit pack remain but will not play.
Administering Announcements (recording, copying, deleting, etc.)	*Feature Description and Implementation for Avaya Communication Manager*
	2 of 3

Task	Information source
Viewing announcement usage measurements (**`list measurements announcement`** command)	*Reports for Avaya Communication Manager* and *Feature Description and Implementation for Avaya Communication Manager*
Troubleshooting announcements	*Feature Description and Implementation for Avaya Communication Manager*.
Troubleshooting VAL hardware	*Maintenance Procedures for Avaya Communication Manager* for your model(s).
	3 of 3

Chapter 14: Managing Group Communications

Setting up Voice Paging Over Loudspeakers

Use this procedure to allow users to make voice pages over an external loudspeaker system connected to Avaya Communication Manager. If you're using an external paging system instead of an auxiliary trunk circuit pack, don't use this procedure. External systems typically connect to a trunk or station port and are not administered through the **Loudspeaker Paging** screen.

Before you start

Your server running Communication Manager must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. For information on specific circuit packs, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Instructions

As an example, we will set up voice paging for an office with 5 zones. We'll allow users to page all 5 zones at once, and we'll assign a class of restriction of 1 to all zones.

1. Type **`change paging loudspeaker`**. Press **Enter**.

   The **Loudspeaker Paging screen** appears.

Figure 184: Loudspeaker Paging screen

```
change paging loudspeaker                                          Page   1 of   x
                             LOUDSPEAKER PAGING

                                   CDR? _
            Voice Paging Timeout (sec): ___
           Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                 Voice Paging     Code Calling
 Zone  Port      TAC  COR  TN     TAC  COR  TN     Location:
   1: _______    ____ __   _      ____ __   _      ___________________________
   2: _______    ____ __   _      ____ __   _      ___________________________
   3: _______    ____ __   _      ____ __   _      ___________________________
   4: _______    ____ __   _      ____ __   _      ___________________________
   5: _______    ____ __   _      ____ __   _      ___________________________
   6: _______    ____ __   _      ____ __   _      ___________________________
   7: _______    ____ __   _      ____ __   _      ___________________________
   8: _______    ____ __   _      ____ __   _      ___________________________
   9: _______    ____ __   _      ____ __   _      ___________________________
 ALL: _______    ____ __   _      ____ __   _
```

2. In the **Voice Paging Timeout** field, type **30**.

   This field sets the maximum number of seconds a page can last. In our example, the paging party will be disconnected after 30 seconds.

3. In the **Port** field for **Zone 1**, type **01C0501**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the **Voice Paging — TAC** field type **301**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the **Voice Paging — COR** field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the **Zone 1** row, type **Reception area** in the **Location** field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 to 5.

8. In the **ALL** row, type **310** in the **Voice Paging — TAC** field and **1** in the **Voice Paging — COR** field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.

9. Press **Enter** to save your changes.

You can integrate loudspeaker voice paging and call parking. This is called "deluxe paging." You enable deluxe paging by entering **y** in the **Deluxe Paging and Call Park Timeout to Originator** field on the **Feature-Related System Parameters** screen. To allow paged users the full benefit of deluxe paging, you should also enter a code in the **Answer Back Access Code** field on the **Feature Access Code (FAC)** screen if you haven't already: paged users will dial this code + an extension to retrieve calls parked by deluxe paging.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.
Calls to an extension are heard over the loudspeakers.	The extension might have been forwarded to a trunk access code used for paging.	Deactivate call forwarding or change the extension calls are forwarded to.

More information

Users page by dialing the trunk access code assigned to a zone and speaking into their handset. For your users' convenience, you might also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.
- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the Direct Trunk Group Select buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

When deluxe paging is enabled, if a user with an active call dials the trunk access code for a paging zone the active call is automatically parked.

- Users dial the trunk access code + "#" to page and park an active call on their own extensions.
- Users with console permission can park a call on any extension by dialing the trunk access code + the extension.
- Attendants or users with console permissions can park calls to common shared extensions.
- Parked calls can be retrieved from any telephone. Paged users simply dial the answer back feature access code + the extension where the call is parked.

Related topics

See Paging Over Speakerphones on page 523 for another way to let users page.

See "Loudspeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on voice paging over loudspeakers.

Setting up Chime Paging Over Loudspeakers

Use this procedure to allow users to make chime pages over an external loudspeaker system connected to your Avaya S8XXX Server. Users page by dialing a trunk access code and the extension of the person they want to page. The system plays a unique series of chimes assigned to that extension. This feature is also known as Code Calling Access.

Before you start

Your server running Communication Manager must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. For information on specific circuit packs, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Instructions

To set up chime paging, you fill out the necessary fields on the **Loudspeaker Paging** screen and then assign chime codes to individual extensions on the **Code Calling IDs** screen.

As an example, we will set up chime paging for a clothing store with 3 zones. We'll allow users to page all zones at once, and we will assign a class of restriction of 1 to all zones.

1. Type **`change paging loudspeaker`**. Press **Enter**.

   The **Loudspeaker Paging screen** appears.

Figure 185: Loudspeaker Paging screen

```
change paging loudspeaker                                         Page   1 of   x
                              LOUDSPEAKER PAGING

                                   CDR? _
            Voice Paging Timeout (sec): ___
           Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                 Voice Paging     Code Calling
 Zone  Port      TAC  COR  TN     TAC  COR  TN     Location:
   1: _______    ____ __   _      ____ __   _      ___________________________
   2: _______    ____ __   _      ____ __   _      ___________________________
   3: _______    ____ __   _      ____ __   _      ___________________________
   4: _______    ____ __   _      ____ __   _      ___________________________
   5: _______    ____ __   _      ____ __   _      ___________________________
   6: _______    ____ __   _      ____ __   _      ___________________________
   7: _______    ____ __   _      ____ __   _      ___________________________
   8: _______    ____ __   _      ____ __   _      ___________________________
   9: _______    ____ __   _      ____ __   _      ___________________________
 ALL: _______    ____ __   _      ____ __   _
```

2. In the **Code Calling Playing Cycles** field, type **2**.

   This field sets the number of times a chime code plays when someone places a page.

3. In the **Port** field for **Zone 1**, type **01A0301**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the **Code Calling — TAC** field type **80**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the **Code Calling — COR** field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the **Zone 1** row, type **Men's Department** in the **Location** field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 and 3.

8. In the **ALL** row, type **89** in the **Code Calling — TAC** field and **1** in the **Code Calling — COR** field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.

9. Press **Enter** to save your changes.

To assign chime codes to individual extensions:

1. Type **`change paging code-calling-ids`**. Press **Enter**.

   The **Code Calling IDs screen** appears.

Figure 186: Code Calling IDs screen

```
change paging code-calling-ids                                    Page 1 of x

                                     CODE CALLING IDs
    ID ASSIGNMENTS
           Id   Ext        Id   Ext        Id   Ext        Id   Ext        Id   Ext
           111: ____       141: ____       221: ____       251: ____       331: ____
           112: ____       142: ____       222: ____       252: ____       332: ____
           113: ____       143: ____       223: ____       253: ____       333: ____
           114: ____       144: ____       224: ____       254: ____       334: ____
           115: ____       145: ____       225: ____       255: ____       335: ____
           121: ____       151: ____       231: ____       311: ____       341: ____
           122: ____       152: ____       232: ____       312: ____       342: ____
           123: ____       153: ____       233: ____       313: ____       343: ____
           124: ____       154: ____       234: ____       314: ____       344: ____
           125: ____       155: ____       235: ____       315: ____       345: ____
           131: ____       211: ____       241: ____       321: ____       351: ____
           132: ____       212: ____       242: ____       322: ____       352: ____
           133: ____       213: ____       243: ____       323: ____       353: ____
           134: ____       214: ____       244: ____       324: ____       354: ____
           135: ____       215: ____       245: ____       325: ____       355: ____
```

2. Type the first extension, **2130**, in the **Ext** field for **Id 111**.

   Each code Id defines a unique series of chimes.

3. Assign chime codes to the remaining extensions by typing an extension number on the line following each code Id.

   You can assign chime codes to as many as 125 extensions.

4. Press **Enter** to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.

More information

Users page by dialing the trunk access code assigned to a zone. For your users' convenience, you might also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.

  Note:

  Don't use special characters in abbreviated dialing lists used with chime paging.

- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the **Direct Trunk Group Select** buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

Related Topics

See Paging Over Speakerphones below for another way to let users page.

See "Loudpeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on chime paging over loudspeakers.

Paging Over Speakerphones

Use this procedure to allow users to make an announcement over a group of digital speakerphones. By dialing a single extension that identifies a group, users can page over all the speakerphones in that group. Speakerphone paging is one-way communication: group members hear the person placing the page but cannot respond directly.

Before you start

You must have DCP set speakerphones or IP set speakerphones.

Instructions

To set up speakerphone paging, you create a paging group and assign telephones to it. In the following example, we'll create paging group 1 and add 4 members.

1. Type **`add group-page 1`**. Press **Enter**.

   The **Group Paging Using Speakerphone screen** appears.

Figure 187: Group Paging Using Speakerphone screen

```
add group-page next                                             Page 1 of x
                        GROUP PAGING USING SPEAKERPHONE
   Group Number: 1                          Group Extension: 3210
     Group Name: Sales staff                            COR: 5
GROUP MEMBER ASSIGNMENTS                                 TN: 1
   Ext      Name                           Ext      Name
 1: 2009    B. Smith                    17:
 2: 2010    R. Munoz                    18:
 3: 2011    Y. Lu                       19:
 4: 2012    A. Sullivan                 20:
 5:                                     21:
 6:                                     22:
 7:                                     23:
 8:                                     24:
 9:                                     25:
10:                                     26:
11:                                     27:
12:                                     28:
13:                                     29:
14:                                     30:
15:                                     31:
16:                                     32:
```

2. In the **Group Extension** field, type **3210**.

   This field assigns the extension users dial to page the members of this group.

3. In the **Group Name** field, type **Sales staff**.

   This name appears on callers' telephone displays when they page the group.

4. In the **COR** field, type **5**.

   Any user who wants to page this group must have permission to call COR 5.

5. In the **Ext** field in row 1, type **2009**.

6. Enter the remaining extensions that are members of this group.

   Communication Manager fills in the **Name** fields with the names from the **Station** screen when you save your changes.

7. Press **Enter** to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users get a busy signal when they try to page.	All telephones in the group are busy or off-hook.	Wait a few minutes and try again.
	All telephones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.
Some group members report that they don't hear a page.	Some telephones in the group are busy or off-hook.	Wait a few minutes and try again.
	Some telephones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.

More information

- You can create up to 32 paging groups on Avaya Communication Manager.
- Each group can have up to 32 extensions in it.
- One telephone can be a member of several paging groups.

Related topics

See "Group Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on paging over speakerphones.

Paging Users Who are on Active Calls

Use this procedure to allow one user to interrupt another user's call and make a private announcement. This is called whisper paging. The paging user dials a feature access code or presses a feature button, then dials the extension they want to call. All 3 users can hear the tone that signals the page, but only the person on the paged extension can hear the pager's voice: other parties on the call cannot hear it, and the person making the page cannot hear anyone on the call.

Before you start

Before you administer whisper paging:

- Your Communication Manager server must have a circuit pack that supports whisper paging. For information on specific models, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.
- Users must have 6400-, 7400-, 8400-, or 9400-series DCP (digital) telephones.

Instructions

You give users the ability to use whisper paging by administering feature buttons or feature access codes.

You can give users feature buttons that make, answer, or block whisper pages. Using the **Station** screen, you can administer these buttons in any combination as appropriate:

- Whisper Page Activation — allows this user to place a whisper page
- Answerback — allows this user to answer a whisper page

  Pressing the answerback button automatically puts any active call on hold and connects the paged user to the paging user.
- Whisper Page Off— allows this user to block whisper pages

  If possible, assign this function to a button with a lamp so the user can tell when blocking is active. You cannot administer this button to a soft key.

To allow users to make a whisper page by dialing a feature access code, you simply need to enter a code in the **Whisper Page Activation Access Code** field on the **Feature Access Code (FAC)** screen.

Related topics

See "Whisper Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on whisper paging.

Using Telephones as Intercoms

Use this procedure to make communications quicker and easier for users who frequently call each other. With the intercom feature, you can allow one user to call another user in a predefined group just by pressing a couple of buttons. You can even administer a button that always calls a predefined extension when pressed.

Instructions

Administering the intercom feature is a 2-step process. First, you create an intercom group and assign extensions to it. Then, to allow group members to make intercom calls to each other, you administer feature buttons on their telephones for automatic intercom, dial intercom, or both. This section also provides instructions for allowing one user to pick up another user's intercom calls.

In this example, we'll create intercom group 1 and add extensions 2010 to 2014.

1. Type **`add intercom-group 1`**. Press **Enter**.

   The Intercom Group screen appears.

Figure 188: Intercom Group screen

```
change intercom-group n                                           Page 1 of x
                                 INTERCOM GROUP
                                               Group Number: n
                                        Length of Dial Code: _

GROUP MEMBER ASSIGNMENTS
              Ext      DC   Name
          1: ________ __
          2: ________ __
          3: ________ __
          4: ________ __
          5: ________ __
          6: ________ __
          7: ________ __
          8: ________ __
          9: ________ __
         10: ________ __
         11: ________ __
         12: ________ __
         13: ________ __
         14: ________ __
         15: ________ __
         16: ________ __
```

2. Type **1** in the **Length of Dial Code** field.

   Dial codes can be 1 or 2 digits long.
3. On row 1, type **2010** in the **Ext** field.
4. On row 1, type **1** in the **DC** field.

   This is the code a user will dial to make an intercom call to extension 2010. The length of this code must exactly match the entry in the **Length of Dial Code** field.
5. Repeat steps 3 and 4 for the remaining extensions.

   Dial codes don't have to be in order. Communication Manager fills in the **Name** field with the name from the **Station** screen when you save changes.
6. Press **Enter** to save your changes.

To allow users to make intercom calls, you must administer feature buttons on the telephones in the intercom group. You can administer buttons for dial intercom, automatic intercom, or both on multi-appearance telephones. You can't administer either intercom feature on single-line telephones, but you can assign single-line telephones to intercom groups so those users can receive intercom calls.

As an example, we will set up automatic intercom between extensions 2010 (dial code = 1) and 2011 (dial code = 2) in intercom group 1.

To set up automatic intercom between extensions 2010 and 2011:

1. Type **`change station 2010`**. Press **Enter**.

   The Station screen (page 4) appears.

Figure 189: Station screen

```
add station nnnn                                                 Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                         Headset? n
          Jack: ___                                             Speaker? n
         Cable: ___                                            Mounting: d
         Floor: _______                                    Cord Length: 0_
      Building: _______                                      Set Color: _____


ABBREVIATED DIALING
     List1: ________            List2: _________            List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                  6:limit-call
 2: call-appr                  7:team     Ext: 5381231          Rg:
 3: call-appr                  8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000       9:cfwd-enh Ext: 5502
 5: release                   10:aux-work RC: 1 Group:

voice-mail Number:
```

2. Move to the page with the **BUTTON ASSIGNMENTS** fields.
3. In **BUTTON ASSIGNMENTS** field **4**, type **auto-icom**. Press **Tab**.

   The **Grp** and **DC** fields appear.
4. In the **Grp** field, type **1**.

   This is the number of the intercom group. Since an extension can belong to more than one intercom group, you must assign a group number to intercom buttons.
5. In the **DC** field, type **2**.

   This is the dial code for extension 2011, the destination extension.
6. Press **Enter** to save your changes.
7. Repeat steps 1 to 6 for extension 2011.

   Assign a dial code of **1** to 2011's automatic intercom button.

To give a member of a group the ability to make intercom calls to all the other members, administer a **Dial Intercom** button on the member's telephone. Type the number of the intercom group in the **Grp** field beside the **Dial Intercom** button.

You can also give one user instant, one-way access to another. For example, to give user A instant, one-way access to user B, administer an **Automatic Intercom** button on A's telephone only. You don't have to administer any intercom button on B's telephone. If B has a **Dial Intercom** button, he can make an intercom call to A the same way as he would to any other group member.

When users are in the same call pickup group, or if Directed Call Pickup is enabled on your server running Communication Manager, one user can answer an intercom call to another user. To allow users to pick up intercom calls to other users, you must enter **y** in the **Call Pickup on Intercom Calls** field on the **Feature-Related System Parameters screen**.

Related topics

See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for information on another way for users to call each other without dialing complete extension numbers.

See "Intercom" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on intercom functions.

Setting up Automatic Answer Intercom Calls

Automatic Answer Intercom Calls (Auto Answer ICOM) allows a user to answer an intercom call within the intercom group without pressing the intercom button. Auto Answer ICOM works with digital, BRI, and hybrid telephones with built-in speaker, headphones, or adjunct speakerphone.

SECURITY ALERT:
Press the **Do Not Disturb** button or the **Send All Calls** button on your telephone when you don't want someone in your intercom group to listen in on a call. Auto Answer ICOM does not work when the **Do Not Disturb** button or the **Send All Calls** button is pressed on the telephone.

Administering Auto Answer ICOM

This section contains an example, with step-by-step instructions, on how to set up Auto Answer ICOM.

In this example, you set up Auto Answer ICOM on station 12345. To do so, complete the following steps:

1. Type **`change station 12345`**.

   The **Station** screen for extension 12345 appears. Click **Next Page** until you see the **Feature Options** page on the Station screen (page 2).

Figure 190: Station screen

```
change station nnnn                                                Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
             CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                       Call Waiting Indication:
 Per Button Ring Control? n                 Attd. Call Waiting Indication:
   Bridged Call Alerting? n                     Idle Appearance Preference? n
        Switchhook Flash? n                   Bridged Idle Line Preference? y
    Ignore Rotary Digits? n                       Restrict Last Appearance? y
  Active Station Ringing: single        Conf/Trans On Primary Appearance? n
                                                        EMU Login Allowed?
        H.320 Conversion? n         Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed      Busy Auto Callback without Flash? y
         Multimedia Mode: basic
    MWI Served User Type: ______                Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                      Select Last Used Appearance? n
                               _                 Coverage After Forwarding? _
     Recall Rotary Digit? n                          Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n       IP Audio Hairpinning? n
Precedence Call Waiting? y
```

2. Move to the **Auto Answer** field and enter **icom**.
3. Press **Enter** to save your changes.

Observing Calls

Use this procedure to allow designated users, normally supervisors, to listen to other users' calls. This capability is often used to monitor service quality in call centers and other environments where employees serve customers over the telephone. On Avaya Communication Manager, this is called "service observing" and the user observing calls is the "observer."

This section describes service observing in environments without Automatic Call Distribution (ACD) or call vectoring. To use service observing in those environments, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Before you start

On the **System Parameter Customer-Options** screen, verify that the:

- **Service Observing (Basic)** field is **y**.

If you want to enable remote service observing by allowing remote users to dial a feature access code, verify the:

- **Service Observing (Remote/By FAC)** field is **y**.

If the appropriate field is not enabled, contact your Avaya representative.

Instructions

SECURITY ALERT:
Listening to someone else's calls might be subject to federal, state, or local laws, rules, or regulations. It might require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

In this example, we'll set up service observing for a manager. The manager's class of restriction is 5. We'll assign a feature button to the manager's telephone and allow her to monitor calls on local extensions that have a class of restriction of 10. Everyone on an observed call will hear a repetitive warning tone.

To set up service observing:

1. Set the observer's class of restriction to permit service observing:
   a. In the **Class of Restriction** screen for COR 5, type **y** in the **Can Be A Service Observer?** field.
   b. Move to the page of the **Class of Restriction** screen that shows service observing permissions.
   c. Type **y** in the field for class of restriction 10.
2. In the **Class of Restriction** screen for COR 10, type **y** in the **Can Be Service Observed?** field.

   Anyone with class of restriction 5 now has permission to observe extensions with class of restriction 10. To further restrict who can observe calls or be observed, you might want to create special classes of restriction for both groups and use these classes only for the appropriate extensions.
3. In the **Station** screen, assign a **Service Observing** button to the observer's telephone.

   A service observing button permits users to switch between listen-only and listen-and-talk modes simply by pressing the button.
4. To activate the warning tone, type **y** in the **Service Observing — Warning Tone** field on the **Feature-Related System Parameters** screen.

   A unique 2-second, 440-Hz warning tone plays before an observer connects to the call. While the call is observed, a shorter version of this tone repeats every 12 seconds.

In order for users to activate service observing by feature access codes, use the **Feature Access Code (FAC)** screen to administer codes in one or both of the following fields:

- **Service Observing Listen Only Access Code**
- **Service Observing Listen/Talk Access Code**

When using feature access codes, observers must choose a mode at the start of the session. They cannot switch to the other mode without ending the session and beginning another.

Note:

Feature access codes are required for remote observing.

Related topics

See "Service Observing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for detailed information on service observing.

Chapter 15: Managing Data Calls

Types of Data Connections

You can use Avaya Communication Manager to allow the following types of data elements/ devices to communicate to the world:

- Data Terminals
- Personal computers
- Host Computers (for example, CentreVu CMS or INTUITY AUDIX)
- Digital Phones (Digital Communications Protocol (DCP) and Integrated Services Digital Network-Basic Rate Interface (ISDN-BRI))
- Audio/Video Equipment
- Printers
- Local area networks (LAN)

You enable these connections using a large variety of data communications equipment, such as:

- Modems
- Data Modules
- Asynchronous Data Units (ADU)
- Modem Pools
- Data/modem pooling circuit packs

Once you have connected these data devices to Communication Manager, you can use networking and routing capabilities to allow them to communicate with other devices over your private network or the public network.

This section describes the system features available to enable data communications.

Data Call Setup

Data Call Setup provides multiple methods to set up a data call:

- Data-terminal (keyboard) dialing
- Telephone dialing
- Hayes AT command dialing
- Administered connections
- Hotline dialing

Administering Data Call Setup

For data-terminal dialing:

1. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module
   - 7500 Data Module
2. On the **Modem Pool Group** screen, administer the **Circuit Pack Assignments** field.

   See Modem Pool Group on page 1347 for more information.

For telephone dialing:

1. Choose one of the following:
   - On the **Feature Access Code (FAC)** screen, administer the **Data Origination Access Code** field. See Feature Access Code (FAC) on page 980 for more information.
   - On the **Station** screen, assign one button as data-ext (Ext:). See Station on page 1491 for more information.
2. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module
3. On the **Modem Pool Group** screen, administer the **Circuit Pack Assignments** field. See Modem Pool Group on page 1347 for more information.

Depending on the hardware used, assign ports to the following:

- Data modules
- 7400D-series or CALLMASTER digital telephones
- 7500D-series telephones with asynchronous data module (ADM)
- Analog modems (port is assigned using 2500 telephone screen)

Characters used in Data Call Setup

Basic-digit dialing is provided through an ADM or 7500B data module. The user can enter digits from 0 to 9, *, and # from a 7500 or 8500 series telephone keypad or an EIA-terminal interface. In addition, the user can dial the following special characters.

Table 6: Special characters

Character	Use
SPACE, -, (, and)	improves legibility. Communication Manager ignores these characters during dialing.
+ character (wait)	interrupts or suspends dialing until the user receives dial tone
, (pause)	inserts a 1.5-second pause
% (mark)	indicates digits for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is touch-tone.
UNDERLINE or **BACKSPACE**	corrects previously typed characters on the same line
@	deletes the entire line and starts over with a new DIAL: prompt

Each line of dialing information can contain up to 42 characters (the + and % characters count as two each).

Examples of dialing are:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8, 555-2368
- DIAL: 9+555-2368+%9999+123 (remote access)

The following call-progress messages and their meanings are provided for DCP and ISDN-BRI modules.

Table 7: Call-progress messages

Message	Application	Meaning
DIAL:	DCP	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
CMD	BRI	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
RINGING	DCP, BRI	Equivalent to ringing tone. Called terminal is ringing.
BUSY	DCP, BRI	Equivalent to busy tone. Called number is busy or out of service.
ANSWERED	DCP, BRI	Call is answered.
ANSWERED - NOT DATA	DCP	Call is answered and a modem answer tone is not detected.
TRY AGAIN	DCP, BRI	Equivalent to reorder tone. System facilities are currently not available.
DENIED	DCP, BRI	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED	DCP, BRI	Calling user has abandoned the call.
NO TONE	DCP, BRI	Tone is not detected.
CHECK OPTIONS	DCP, BRI	Data-module options are incompatible.
XX IN QUEUE	DCP, BRI	Current position in queue.
PROCESSING	DCP, BRI	Out of queue. Facility is available.
TIMEOUT	DCP, BRI	Time is exceeded. Call terminates.
FORWARDED	DCP, BRI	Equivalent to redirection-notification signal. Called terminal activates Call Forwarding and receives a call, and call is forwarded.
INCOMING CALL	DCP, BRI	Equivalent to ringing.
		1 of 2

Table 7: Call-progress messages (continued)

Message	Application	Meaning
INVALID ADDRESS	DCP	Entered name is not in alphanumeric-dialing table.
WRONG ADDRESS	BRI	Entered name is not in alphanumeric-dialing table.
PLEASE ANS-	DCP, BRI	Originating telephone user transferred call to data module using One-Button Transfer to Data.
TRANSFER	DCP	Data Call Return-to-Voice is occurring.
CONFIRMED	DCP, BRI	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
OTHER END	DCP, BRI	Endpoint has terminated call.
DISCONNECTE D	DCP, BRI	Call is disconnected.
WAIT	DCP, BRI	Normal processing continues.
WAIT, XX IN QUEUE	DCP	Call is in a local hunt-group queue.
		2 of 2

DCP data modules

Data-terminal dialing

DCP data-terminal dialing allows a user to set up and disconnect data calls directly from a data terminal as follows.

1. At the **DIAL**: prompt, the user types the data number. The message **RINGING** displays.
2. If the call is queued, the message **WAIT, XX IN QUEUE** displays. The queue position XX updates as the call moves up in queue.
3. To originate and disconnect a call, the user presses **BREAK**. If the terminal does not generate a 2-second continuous break signal, the user can press originate/disconnect on the data module.
4. The user can enter digits at the **DIAL**: prompt.

Telephone dialing

DCP telephone dialing allows telephone users to originate and control data calls from a telephone.

Users can set up a call using any unrestricted telephone and then transfer the call to a data endpoint.

The primary way to make data calls is with multiappearance telephone data-extension buttons. Assign any administrable feature button as a data-extension button. The data-extension button provides one-touch access to a data module. The number of assigned data-extension buttons per telephone is not limited.

The following options, either alone or combined, permit flexibility in making data calls from a telephone.

- One-Button Transfer to Data

  A user can transfer a call to the associated data module by pressing the data-extension button after the endpoint answers.

- Return-to-Voice

  A user can change the connection from data to voice. The user presses the data-extension button associated with the busy data module. If the user hangs up, the call disconnects. Return of a data call to the telephone implies that the same data call is continued in the voice mode, or transferred to point.

  The Return-to-Voice feature is denied for analog adjuncts.

- Data Call Preindication

  A user, before dialing a data endpoint, can reserve the associated data module by pressing the data-extension button. This ensures that a conversion resource, if needed, and the data module are reserved for the call. Avaya recommends the use of Data Call Preindication before 1-button transfer to data for data calls that use toll-network facilities. Data Call Preindication is in effect until the associated data-extension button is pressed again for a 1-button transfer; there is no time-out.

ISDN-BRI data modules

Data-terminal dialing

A user can set up and disconnect data calls directly from a data terminal without using a telephone as follows:

1. The user presses **Enter** a few times. If the CMD: prompt does not appear, the user presses **Break A + T** at the same time, and then presses **Enter**.
2. At the **CMD**: prompt, the user types and presses **Enter**.
3. To disconnect, the user enters +++. At the CMD: prompt, the user types **end** and presses **Enter**.

Telephone dialing

To make a data call, an ISDN-BRI telephone user presses the data button on the terminal, enters the number on the dial pad, and then presses the data button again.

The following data functions are not available on ISDN-BRI telephones:

- One-Button Transfer to Data
- Return-to-Voice
- Data Call Preindication
- Voice-Call Transfer to Data and Data-Call Transfer to Voice

The system handles all presently defined BRI bearer data-call requests. Some capabilities that are not supported by Avaya terminals are provided by non-Avaya terminals. If Communication Manager does not support a capability, a proper cause value returns to the terminal.

BRI terminals receive a cause or reason code that identifies why a call is being cleared. The BRI data module converts certain cause values to text messages for display.

In a passive-bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When you change the configuration of a BRI from point-to-point to multipoint, the original endpoint does not need to reinitialize. Only endpoints that support service profile identifier (SPID) initialization can be administered in a multipoint configuration.

Analog modems

When a telephone user places a data call with a modem, the user dials the data-origination access code assigned in the system before dialing the endpoint.

Considerations

- A BRI telephone cannot call a data terminal, and a data terminal cannot call a BRI telephone.

Interactions

- Abbreviated Dialing

  Only 22 of the 24 (maximum) digits in an abbreviated-dialing number are available for keyboard dialing. The remaining two digits must contain the wait indicator for tone detection.

- Call Coverage

  A hunt group made up of data endpoints cannot be assigned a coverage path.

- Call Detail Recording

  CDR records the use of modem pools on trunk calls.

- Call Forwarding All Calls

  Calls received by a data module can be forwarded. Activate Call Forwarding All Calls with data-terminal (keyboard) dialing. If the forwarded-to endpoint is an analog endpoint and the caller is a digital endpoint, modem pooling is activated automatically.

- Pooled Modems with Hunt Groups

  UCD can provide a group of data modules or analog modems for answering calls to connected facilities (for example, computer ports).

- World-Class Tone Detection

  Multiple-line data-terminal dialing is supported if the administered level of tone detection is precise. You can administer tone-detection options. The message that Data Call Setup sends to users varies according to the option.

  If the option is not set to precise, and a data call is set up over an analog trunk, messages describing the status of the called endpoint (for example, RINGING, BUSY, TRY AGAIN) change according to which tone-detection option is selected.

Alphanumeric Dialing

Alphanumeric Dialing enhances data-terminal dialing by allowing users to place data calls by entering an alphanumeric name rather than a long string of numbers.

For example, a user could type 9+1-800-telefon instead of 9+1-800-835-3366 to make a call. Users need to remember only the alpha-name of the far-end terminating point.

Alphanumeric Dialing allows you to change a mapped string (digit-dialing address) without having to inform all users of a changed dial address. Users dial the alpha name.

When a user enters an alphanumeric name, the system converts the name to a sequence of digits according to an alphanumeric-dialing table. If the entered name is not found in the table, the system denies the call attempt and the user receives either an `Invalid Address` message (DCP) or a `Wrong Address` message (ISDN-BRI).

Because data terminals access Communication Manager via DCP or ISDN-BRI data modules, dialing procedures vary:

- For DCP, at the `DIAL`: prompt users type the alphanumeric name. Press **Enter**.
- For ISDN-BRI, at the **CMD:** prompt users type **d**, a space, and the alphanumeric name. Press **Enter**.

More than one alphanumeric name can see the same digit string.

Administering Alphanumeric Dialing

To set up alphanumeric dialing:

1. On the **Alphanumeric Dialing Table** screen, administer the **Alpha-name** and **Mapped String** fields. See Alphanumeric Dialing Table on page 767 for more information.

Considerations

Note:

Alphanumeric dialing does not apply to endpoints with Hayes modems.

Data Hotline

Data Hotline provides for automatic-nondial placement of a data call preassigned to an endpoint when the originating server goes off-hook. Use for security purposes.

Administering Data Hotline

To administer a data hotline:

1. You can use an abbreviated dialing list for your default ID. See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.
2. On the **Station** screen, administer the following fields.
   - Abbreviated Dialing List
   - Special Dialing Option
   - Hot Line Destination
3. On the **Data Module** screen, administer the **Abbreviated Dialing List1** field.

The system automatically places Data Hotline calls to preassigned extensions or off-premises numbers. Calling terminals are connected to the system by a data module. Users should store the destination number in the abbreviated dialing list for future reference.

Interactions

- Call Forwarding — All Calls

  A Data Hotline caller cannot activate both Call Forwarding and Data Hotline. Dialing the Call Forwarding feature access code (FAC) causes activation of the Data Hotline instead.

Data Privacy

Data Privacy protects analog data calls from being disturbed by any of the system's overriding or ringing features.

Administering Data Privacy

To administer data privacy:

1. Choose either of the following:
   - On the **Feature Access Code (FAC)** screen, administer the **Data Privacy Access Code** field.
   - On the **Class of Service** screen, administer the **Data Privacy** field.
2. On the **Station** screen, administer the **Class of Service** field.

To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

- Data Privacy applies to both voice and data calls. You can activate Data Privacy on Remote Access calls, but not on other incoming trunk calls. Data Privacy is canceled if a user transfers a call, is added to a conference call, is bridged onto a call, or disconnects from a call. You can activate Data Privacy on calls originated from attendant consoles.
- For virtual extensions, assign the Data Privacy Class of Service to the mapped-to physical extension.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is active, Call Waiting is denied.
- Bridged Call Appearance — Single-Line Telephone

  If you activate Data Privacy or assign Data Restriction to a station involved in a bridged call and the primary terminal or bridging user attempts to bridge onto the call, this action overrides Data Privacy and Data Restriction.
- Busy Verification

  Busy Verification cannot be active when Data Privacy is active.

- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction active cannot originate an intercom call. The user receives an intercept tone.
- Music-on-Hold Access

  If a user places a call with Data Privacy on hold, the user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.
- Priority Calls

  If a user activates Data Privacy, Priority Calls are denied on analog telephones. However, Priority Calls appear on the next available line appearance on multiappearance telephones.

Default Dialing

Default Dialing provides data-terminal users who dial a specific number the majority of the time a very simple method of dialing that number. Normal data terminal dialing and alphanumeric dialing are unaffected.

Default Dialing enhances data terminal (keyboard) dialing by allowing a data-terminal user to place a data call to a preadministered destination by either pressing **Enter** at the DIAL: prompt (for data terminals using DCP data modules) or typing **d** and pressing **Enter** at the CMD: prompt (for data terminals using ISDN-BRI data modules). The data-terminal user with a DCP data module can place calls to other destinations by entering the complete address after the DIAL: prompt (normal data terminal dialing or alphanumeric dialing). The data-terminal user with an ISDN-BRI data module can place calls to other destinations by typing **d**, a space, the complete address. Press **Enter** after the CMD: prompt.

Note:

DU-type hunt groups connecting the system to a terminal server on a host computer have hunt-group extensions set to **no** keyboard dialing.

For the AT command interface supported by the 7400A/7400B/8400B data module, to dial the default destination, enter the ATD command (rather than press return).

Administering Default Dialing

To administer default dialing:

1. You can use an abbreviated dialing list for your default ID. See "Abbreviated Dialing" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.
2. On the **Data Module** screen, administer the following fields:
   - **Special Dialing Option** as default.
   - **Abbreviated Dialing List**, enter the list to use.
   - **AD Dial Code**.

Data Restriction

Data Restriction protects analog-data calls from being disturbed by any of the system's overriding or ringing features or system-generated tones.

Data Restriction applies to both voice and data calls.

Once you administer Data Restriction for an analog or multiappearance telephone or trunk group, the feature is active on all calls to or from the terminal or trunk group.

Note:
Do not assign Data Restriction to attendant consoles.

Administering Data Restriction

To administer data restriction:

1. On the **Station** screen, set the **Data Restriction** field to **y**.
2. Choose one of the following trunk groups and set the **Data Restriction** field to **y**. See Trunk Group on page 1669 for more information.
   - Access
   - Advanced Private-Line Termination (APLT)
   - Circuit Pack (CP)
   - Customer-Premises Equipment (CPE)
   - Direct Inward Dialing (DID)

- Foreign Exchange (FX)
- Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI)
- Release-Link Trunk (RLT)
- Tandem
- Tie
- Wide Area Telecommunications Service (WATS)

Interactions - Data Restriction

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is active, Call Waiting is denied.
- Busy Verification

  Busy Verification cannot be active when Data Restriction is active.
- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. The user receives an Intercept tone.
- Music-on-Hold Access

  If a user places a call with Data Restriction on hold, The user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.
- Priority Calls

  Priority Calls are allowed if the analog station is idle. Call Waiting (including Priority Call Waiting) is denied if the station is busy. However, Priority Calls appear on the next available line appearance on multiappearance telephones.
- Service Observing

  A data-restricted call cannot be service observed.

Data-Only Off-Premises Extensions

Data-Only Off-Premises Extensions allows users to make data calls involving data communications equipment (DCE) or digital terminal equipment (DTE) located remotely from the system site.

A Data-Only Off-Premises Extension uses an on-premises modular trunk data module (MTDM). The system communicates with remote data equipment through the private-line facility linking the on-premises MTDM and the remote data equipment.

Users can place data calls to this type of data endpoint using Telephone Dialing or Data Terminal (Keyboard) Dialing. Since there is no telephone at the remote site, originate data calls from the remote data terminal using Keyboard Dialing only.

Administering Data-Only Off-Premises Extensions

To administer data-only off-premises extensions:

1. On the **Processor/Trunk Data Module** screen, administer all fields.

   See Data Module on page 895 for more information.

Considerations

- The system does not support communications between two TDMs. Modem Pooling is similar to a TDM, it cannot be used on calls to or from a Data-Only Off-Premises Extension.

Interactions

- Telephone Dialing

  An on-premises multiappearance telephone might have a Data Extension button associated with the TDM used for a Data-Only Off-Premises Extension. The telephone user and the remote user share control of the data module. Actions of the user at the telephone might affect the remote user.

  - 1-Button Transfer to Data

    The telephone user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button lamp on the telephone lights and the Call in Progress lamp on the data module lights during a data call.

- Data Call Preindication

  The multiappearance telephone user presses the idle associated Data Extension button to reserve a data module. The data module is busy to all other users. When the user reserves a data module, the lamp associated with the Data Extension button winks and lights at any other associated telephones. A remote user receives the BUSY message when attempting to originate a call.

- Return-to-Voice

  To establish a data call, the telephone user presses the associated busy Data Extension button to transfer the call to the telephone. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

Data Modules — General

A data module is a connection device between a basic-rate interface (BRI) or DCP interface of the Avaya S8XXX Server and DTE or DCE.

The following types of data modules can be used with the system:

- Announcement data module
- Data line data module
- Processor/trunk data module (P/TDM)
- 7500 data module
- World Class BRI data module
- Ethernet data module.
- Point-to-Point Protocol (PPP) data module.

For more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Note:

The 51X series Business Communications Terminals (BCT) are not administered on the **Data Module** screen. The 510 BCT (equivalent to a 7405D with a display and built-in DTDM), 515 BCT (equivalent to a 7403D integrated with 7405D display module function, data terminal and built-in DTDM), and the 7505D, 7506D, and 7507D have a DCP interface but have built-in data module functionality. Both are administered by means of the **Station** screen in Communication Manager.

Detailed description of data modules

TTI allows data modules without hardware translation to merge with an appropriate data module connected to an unadministered port. The unadministered port is given TTI default translation sufficient to allow a terminal connected to the data module (connected to the port) to request a TTI merge with the extension of a data module administered without hardware translation.

Note:

TTI is not useful for Announcement and X.25 hardware.

Administration Without Hardware supports PDM, TDM, Data-Line, Announcement, and X.25 data modules.

Note:

The 513 BCT has an EIA interface rather than a DCP interface (no built in data module, attachable telephone, or telephone features). The 513 BCT is not administered; only the data module to which the 513 BCT is connected is administered.

7400A/7400B+/8400B+ Data Module

Use the 7400A data module instead of an MTDM when you support combined Modem Pooling. The 7400A data module supports asynchronous operation at speeds up to 19200-bps, and provides a DCP interface to the server and an EIA 232C interface to the associated modem. The 7400A operates in stand-alone mode as a data module.

7400B+ and 8400B+ data modules support asynchronous-data communications and operate in stand-alone mode for data-only service or in linked mode, which provides simultaneous voice and data service. The 7400B+ and 8400B+ provide voice and data communications to 7400D series telephones and 602A1 CALLMASTER telephones that have a connection to a data terminal or personal computer. The data modules integrate data and voice into the DCP protocol required to interface with the server via a port on a digital-line circuit pack. Use the 7400B+ or 8400B+ instead of an MPDM when you need asynchronous operation at speeds up to 19.2-kbps to provide a DCP interface to the server for data terminals and printers. The 7400B+ and 8400B+ do not support synchronous operation and keyboard dialing. Dialing is provided using the standard Hayes command set.

7400D

This data module supports synchronous operation with AUDIX, CMS, and DCS. It provides synchronous data transmissions at speeds of 19.2-Kbps full duplex.

7400C High Speed Link

The 7400C high-speed link (HSL) is a data-service unit that allows access to DCP data services. It provides synchronous data transmission at speeds of 56- and 64-Kbps and provides a link to high-speed data networks. Used for Group 4 fax applications that include electronic mail and messaging, and electronic storage of printed documents and graphics. Use the 7400C for video teleconferencing and LAN interconnect applications.

7500 Data Modules

The 7500 Data Module connects DTE or DCE to the ISDN network. The 7500 Data Module supports EIA 232C and V.35 interfaces and RS-366 automatic-calling unit interface (for the EIA 232C interface only).

The 7500 has no voice functions. Configure in the following ways:

- Asynchronous DCE

  300, 1200, 2400, 4800, 9600, 19200-bps

- Synchronous DCE

  1200, 2400, 4800, 9600, 19200, 56000, 64000-bps

- Asynchronous DTE (used for modem pooling)

  up to 19200-bps

The 7500 Data Module is stand-alone or in a multiple-mount housing.

Asynchronous Data Module

Note:

The **`alias station`** command cannot be used to alias data modules.

Use the Asynchronous Data Module (ADM) with asynchronous DTEs as a data stand for the 7500 and 8500 Series of ISDN-BRI telephones, thus providing connection to the ISDN network. The ADM provides integrated voice and data on the same telephone and supports data rates of 300, 1200, 2400, 4800, 9600, and 19200-bps. This module also supports the Hayes command set, providing compatibility with PC communications packages.

Administered Connection

An Administered Connection (AC) is a connection between two access or data endpoints. Avaya Communication Manager automatically establishes and maintains the connection based on the attributes you administer. ACs provides the following capabilities.

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per AC
- Administrable alarm strategy per AC
- Establishment/retry/auto restoration order based on administered priority

Detailed description

Establish an AC between the following:

- Two endpoints on the same Avaya DEFINITY server or Avaya S8XXX Server
- Two endpoints in the same private network, but on different servers
- One endpoint on the controlling server and another endpoint off the private network

In all configurations, administer the AC on the server having the originating endpoint. For an AC in a private network, if the two endpoints are on two different servers, normally the connection routes via Automatic Alternate Routing (AAR) through tie trunks (ISDN, DS1, or analog tie trunks) and intermediate servers. If required, route the connection via Automatic Route Selection (ARS) and Generalized Route Selection (GRS) through the public network. The call routes over associated ISDN trunks. When the far-end answers, a connection occurs between the far-end and the near-end extension in the `Originator` field on the **Administered Connection** screen.

Because the system makes an administered connection automatically, you do not use the following:

- Data Call Setup

  Do not assign a default dialing destination to a data module when it is used in an AC.
- Data Hotline

  Do not assign a hotline destination to a data module that is used in an AC.
- Terminal Dialing

  Turn off terminal dialing for data modules involved in an AC. This prevents display of call-processing messages (INCOMING CALL,...) on the terminal.

Access endpoints

Access endpoints are non-signaling trunk ports. They neither generate signaling to the far-end of the trunk nor respond to signaling from the far-end. Designate an access endpoint as the originating endpoint or destination endpoint in an AC.

Typical AC applications

The following are typical AC applications:

- A local data endpoint connection to a local or remote-access endpoint. Examples: an MPDM ACCUNET digital service connecting to SDDN via an ISDN trunk-group DS1 port; an MPDM ACCUNET digital service connecting to an ACCUNET Switched 56 Service via a DS1 port.
- A local-access endpoint connecting to a local or remote-access endpoint. Examples: a DSO cross-connect and a 4-wire leased-line modem to a 4-wire modem connection via an analog tie trunk.
- A local data endpoint connecting to a local or remote data endpoint such as a connection between two 3270 data modules.

Note:

The following guidelines do not include AAR and ARS, or GRS administration information for routing AC calls over trunk groups. See the respective feature elsewhere in this book for that information.

Establishing Administered Connections

The originating server or Avaya S8XXX Server attempts to establish an AC only if one of the following conditions exist:

- AC is active.
- AC is due to be active (either a permanent AC or time-of-day requirements are satisfied if it is a scheduled AC).
- Originating endpoint is in in-service or idle state.

If the originating endpoint is not in service or is idle, no activity takes place for the AC until the endpoint transitions to the desired state. The originating server uses the destination address to route the call to the desired endpoint. When the server establishes two or more ACs at the same time, Communication Manager arranges the connections in order of priority.

AC attempts can fail for the following reasons:

- Resources are unavailable to route to the destination.
- A required conversion resource is not available.

- Access is denied by class of restriction (COR), facilities restriction level (FRL), or bearer capability class (BCC). Or, an attempt is made to route voice-band-data over SDDN trunks in the public switch network.
- Destination address is incorrect.
- Destination endpoint is busy.
- Other network or signaling failures occur.

In the event of a failure, an error is entered into the error log, which generates an alarm, if it is warranted by your alarming strategy. You can display AC failures via the **`status-administered connection`** command.

As long as an AC is due to be active, the originating server continues to establish an AC unless the attempt fails because of an administrative error (for example, a wrong number) or service-blocking condition (for example, outgoing calls barred).

- The frequency with which failed attempts are retried is determined by the administered retry interval (1 to 60 minutes) for each AC.
- Retries are made after the retry interval has elapsed regardless of the restorable attribute of the AC.
- ACs are retried in priority order.
- When you change the time of day on Communication Manager, an attempt is made to establish all ACs in the waiting-for-retry state.

Dropping Administered Connections

An AC remains active until one of the following occurs:

- The AC is changed, disabled, or removed.
- The time-of-day requirements of a scheduled AC are no longer satisfied.
- One of the endpoints drops the connection. This could be because of user action (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, busying out of the endpoint, or handshake failure. If the endpoints are incompatible, the connection is successful until handshake failure occurs.

  Note:

  An AC between access endpoints remains connected even if the attached access equipment fails to handshake.
- An interruption (for example, facility failure) occurs between the endpoints.

If an AC drops because it was disabled/removed or is no longer due to be active, no action is taken. If an AC drops because of changed AC attributes, an immediate attempt is made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If handshake failure resulted in the dropping of the connection, in the case of an AC involving at least one data endpoint, no action is taken for that AC until the **`change administered-connection`** command is executed.

Administered Connections failure: Auto Restoration and Fast Retry

When an active AC drops prematurely, you must invoke either auto restoration or fast retry to determine whether auto restoration is attempted for an active AC.

If you option AC for auto restoration and the connection was routed over SDDN trunks, auto restoration is attempted. During restoration, connections are maintained between the server and both endpoints. In addition to allowing the active session to be maintained, AC also provides a high level of security by prohibiting other connections from intervening in active sessions. Auto restoration generally completes before the 60-second endpoint holdover interval. If auto restoration is successful, the call might be maintained (no guarantee). The restoration is transparent to the user with the exception of a temporary disruption of service while restoration is in progress. A successful restoration is reflected by the *restored* state on the status **Administered Connection** screen. Although the restoration was successful, the data session might not have been preserved.

If auto restoration is not active or if the AC is not routed over SDDN trunks, Communication Manager immediately attempts to reestablish the connection (fast retry). Communication Manager also attempts a retry if the originating endpoint initiated the drop. With fast retry, connections are not maintained on both ends. Fast Retry is not attempted for an AC that was last established via fast retry, unless the AC is active for at least two minutes.

If auto restoration or fast retry fails to restore or reestablish the connection, the call drops and the AC goes into retry mode. Retry attempts continue, at the administered retry interval, as long as the AC is due to be active.

Administering Administered Connections

To administer administered connections:

1. Choose one of the following data modules and administer all fields:
   - Data Line Data Module (use with Data Line circuit pack)
   - Processor/Trunk Data Module (use with one of the following:)

MPDMs, 700D, 7400B, 7400D, or 8400B
MTDMs, 700B, 700C, 700E, or 7400A

- Processor Interface Data Module (for more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504)
- X.25 Data Module (for more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504)
- 7500 Data Module (use with ISDN Line 12-BRI-S-NT or ISDN Line 12-BRI-U-NT circuit pack)
- World Class Core BRI Data Module (use with wcbri)

2. On the **DS1 Circuit Pack** screen, administer all fields. Use with switch node carriers. See DS1 Circuit Pack on page 945 for more information.
3. On the **Access Endpoint** screen, administer all fields. See Access Endpoint on page 745 for more information.
4. Choose one of the following trunk groups and administer all fields. See Trunk Group on page 1669 for more information.
   - ISDN-BRI
   - ISDN-PRI
   - Tie
5. On the **Class of Restriction** screen, administer all fields. See Class of Restriction on page 834 for more information.
6. On the **Class of Service** screen, administer all fields. See Class of Service on page 852 for more information.
7. On the **Dial Plan Record** screen, administer the **Local Node Number** field with a number from 1-63 that matches the DCS switch node number and the CDR node number. See QSIG to DCS TSC Gateway on page 1427 for more information.
8. On the **Administered Connection** screen, administer all fields. See Administered Connection on page 750 for more information.
9. On the **Station** screen, assign one button as ac-alarm. See Station on page 1491 for more information.
10. On the **Attendant Console** screen, assign one button as ac-alarm. See Attendant Console on page 776 for more information.

Interactions

- Abbreviated Dialing

  Use Abbreviated Dialing entries in the `Destination` field. Entries must comply with restrictions.
- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because they are used only for data.
- Call Detail Recording

  For an AC that uses a trunk when CDR is active, the origination extension is the originator of the call.
- Class of Restriction

  Reserve a COR for AC endpoints and SDDN trunks. This restricts endpoints that are not involved in AC from connecting to SDDN trunks or endpoints involved in AC.

- Class of Service/Call Forwarding

  Assign to an AC endpoint a COS that blocks Call Forwarding activation at the endpoint.
- Digital Multiplexed Interface (DMI)

  Use DMI endpoints as the destination in an AC. DMI endpoints do not have associated extensions, so do not use them as the originator in an AC.
- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.
- Modem Pooling

  If you require a modem in an AC, one is inserted automatically. If no modem is available, the connection is dropped.
- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto restoration for an AC that is initially routed over an NFAS facility can fail if the only backup route is over the facility on which the backup D-channel is administered. The backup D-channel might not come into service in time to handle the restoration attempt.
- **`Set Time`** Command

  When you change the system time via the **`set time`** command, all scheduled ACs are examined. If the time change causes an active AC to be outside its scheduled period, the AC is dropped. If the time change causes an inactive AC to be within its scheduled period, Communication Manager attempts to establish the AC.

  If any AC (scheduled or continuous) is in retry mode and the system time changes, Communication Manager attempts to establish the AC.
- System Measurements

  Access endpoints are not measured. All other trunks in an AC are measured as usual.

Modem Pooling

Modem Pooling allows switched connections between digital-data endpoints (data modules) and analog-data endpoints via pods of acoustic-coupled modems. The analog-data endpoint is either a trunk or a line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires conversion via a modem, because the DCP format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa.

Modem Pooling feature provides pools of integrated-conversion modems and combined-conversion modems.

Integrated-conversion modem pools have functionality integrated on the Pooled Modem circuit pack, providing two modems. Each one emulates a TDM cabled to a 212 modem. Integrated are modem pools not available in countries that use A-law companding.

Combined-conversion modem pools are TDMs cabled to any TDM-compatible modem. Combined-conversion modem pools can be used with all systems.

The system can detect the needs for a modem. Data calls from an analog-data endpoint require that the user indicate the need for a modem, because the system considers such calls to be voice calls. Users indicate this need by dialing the data-origination access code field on the **Feature Access Code (FAC)** screen before dialing the digital-data endpoint.

The system provides a Hold Time parameter to specify the maximum time any modem can be held but not used (while a data call is in queue).

Administering Modem Pooling

For Integrated modem poolings:

1. On the **Modem Pool Group** screen, administer all fields. See Modem Pool Group on page 1347 for more information.
2. On the **Feature Access Code (FAC)** screen, administer the **Data Origination Access Code** field. See Feature Access Code (FAC) on page 980 for more information.
3. On the **Data Module** screen, administer all fields. See Data Module on page 895 for more information.

For Combined modem poolings:

1. On the **Modem Pool Group** screen, administer all fields. See Modem Pool Group on page 1347 for more information.
2. On the **Feature Access Code (FAC)** screen, administer the **Data Origination Access Code** field. See Feature Access Code (FAC) on page 980 for more information.

Considerations

- On data calls between a data module and an analog-data endpoint, Return-to-Voice releases the modem and returns it to the pool. The telephone user connects to the analog-data endpoint.
- For traffic purposes, the system accumulates data on modem-pooling calls separate from voice calls. Measurements on the pools also accumulate.
- Modem Pooling is not restricted. Queuing for modems is not provided, although calls queued on a hunt group retain reserved modems.
- Avoid mixing modems from different vendors within a combined pool because such modems might differ in transmission characteristics.
- Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect TDM bus-blocking characteristics.
- Tandem switches or servers do not insert a pooled modem. The originating and terminating servers or switches insert a pooled modem.

Interactions

- Call Detail Recording

  Data call CDR records the use of modem pools on trunk calls.
- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.
- Data Privacy and Data Restriction

  The insertion of a modem pool does not turn off Data Privacy or Data Restriction.
- Data-Only Off-Premises Extensions

  Calls to or from a Data-Only Off-Premises Extension cannot use Modem Pooling, when this type of digital-data endpoint uses a TDM.
- DMI Trunks

  If you place a data call from a local analog-data endpoint to a DMI trunk, you must dial the data-origination access code to obtain a modem. Data calls on DMI trunks to local analog-data endpoints automatically obtain modems.
- DS1 Tie Trunk Service

  Connect modems used for Modem Pooling to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature-access code for data origination.

PC Interface

The personal computer (PC) Interface consists of the PC/PBX platforms and PC/ISDN Platform product family. These products are used with Avaya Communication Manager to provide users of IBM-compatible PCs fully-integrated voice and data workstation capabilities.

Two groups of different configurations are available for PC Interface: group 1 uses DCP and group 2 uses the ISDN-BRI (Basic Rate Interface) protocol.

The group 1 configurations consist of DCP configurations that use a DCP expansion card in the PC to link to the server or Avaya S8XXX Server. Group 1 (shown in DCP PC interface configuration (Group 1) on page 561) uses the following connections:

- The PC Interface card plugs into an expansion slot on the PC. The card has 2 standard 8-pin modular jacks (line and telephone).
- The digital telephone plugs into the telephone jack on the PC Interface card.
- The line jack on the card provides a digital port connection to Avaya DEFINITY servers.
- The distance between the PC Interface card and the PBX should be no more than 1524m for 24-gauge wire or 1219m for 26-gauge wire.

Figure 191: DCP PC interface configuration (Group 1)

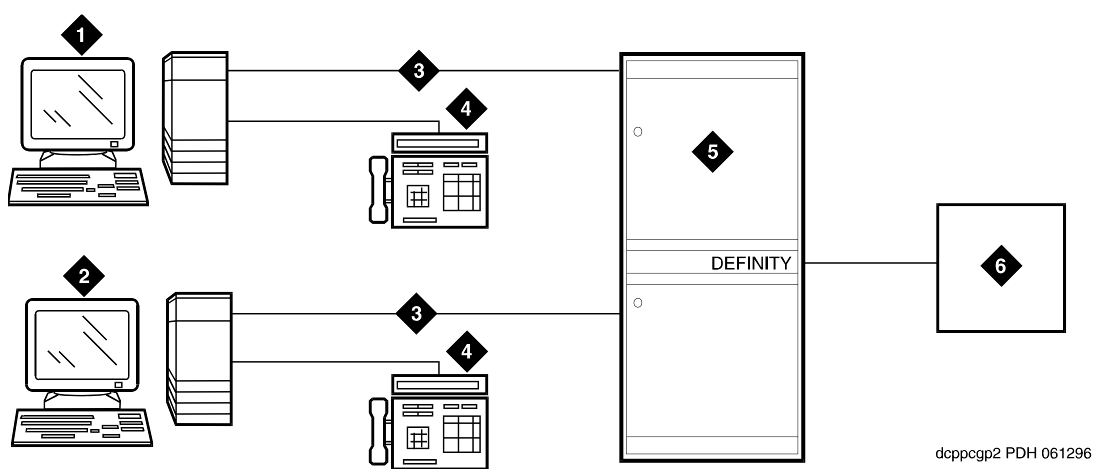


Figure notes:

1. **IBM-compatible PC with DCP Interface card**
2. **IBM-compatible PC with DCP Interface card**
3. **DCP**
4. **DCP telephone**
5. **Avaya (Digital Line, Digital Line (16-DCP-2-Wire), or Digital Line (24-DCP-2-wire) circuit pack)**
6. **Host**

The group 2 configurations link to the server using a PC/ISDN Interface card installed in the PC. This group can include a stand-alone PC terminal, or up to 4 telephones, handsets, or headsets. Group 2 (shown in Figure 192) uses PC/ISDN Interface cards (up to four cards) which plug into expansion slots on the PC. These cards each provide 2 standard 8-pin modular-jack connections for both line connections (to the server or Avaya S8XXX Server) and telephone connections. A standard 4-pin modular jack is also available for use with a handset or headset.

Figure 192: ISDN—BRI PC interface configuration (Group 2)

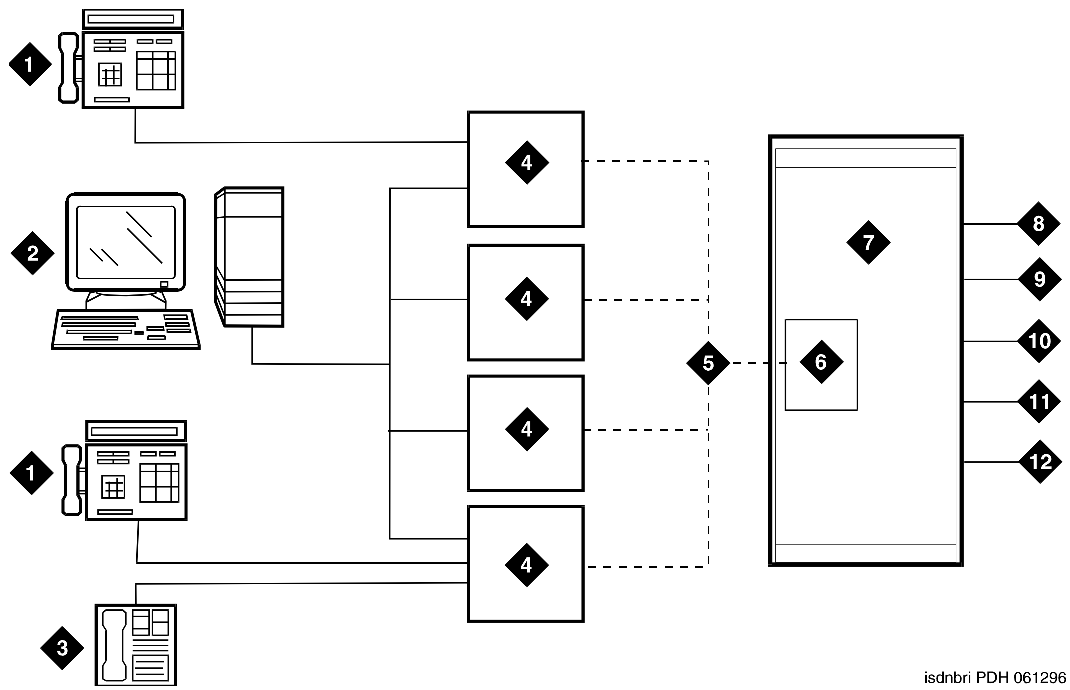


Figure notes:

1. **ISDN telephone**
2. **PC with application**
3. **Handset or Headset**
4. **BRI Interface card**
5. **2B + D**
6. **ISDN Line (12-BRI-S-NT) circuit pack)**
7. **Avaya S8XXX Server**
8. **PRI trunks**
9. **BRI stations**
10. **Interworking**
11. **DMI**
12. **Switch features**

PC Interface users have multiple appearances (depending on the software application used) for their assigned extension. Designate one or more of these appearances for use with data calls. With the ISDN-BRI version, you can use up to 4 separate PC/ISDN Interface cards on the same PC. Assign each card a separate extension, and assign each extension one or more appearances. The availability of specific features depends on the COS of the extension and the COS for Communication Manager. Modem Pooling is provided to ensure general availability of off-net data-calling services.

Security

There are two areas where unauthorized use might occur with this feature: unauthorized local use and remote access.

SECURITY ALERT:
Unauthorized local use involves unauthorized users who attempt to make calls from a PC. The PC software has a security setting so users can place the PC in Security Mode when it is unattended. You also can assign Automatic Security so that the administration program on the PC is always active and runs in Security Mode. This mode is password-protected.

SECURITY ALERT:
Remote access involves remote access to the PC over a data extension. Remote users can delete or copy PC files with this feature. You can password-protect this feature. See the *Avaya Toll Fraud and Security Handbook*, 555-025-600, for additional steps to secure your system and to find out about obtaining information regularly about security developments.

Administering a PC interface

To administer a PC interface:

1. On the **Station** screen, set the **Type** field to **pc**.

Considerations

- Use the Function Key Module of the 7405D with PC Interface.
- BRI terminals normally are initializing terminals and require you to assign an SPID. The PC/ISDN Platform (Group 2), in a stand-alone configuration, is a non-initializing BRI terminal and does not require you to assign a SPID.
  - Set a locally-defined terminal type with General Terminal Administration
  - Define the terminal type as a non-initializing terminal that does not support Management Information Messages (MIM).
  - Assign the PC/ISDN Platform with an associated (initializing) ISDN-BRI telephone (such as an ISDN 7505) using a SPID.
  - Assign the station (using a locally-defined terminal type) to take full advantage of the capabilities of the PC Interface. This terminal type is also non-initializing with no support of MIMs.

- Do not use telephones with data modules with the PC Interface. (You can still use 3270 Data Modules if you also use 3270 emulation). If you attach a DCP data module or ISDN data module to a telephone that is connected to a PC Interface card, the data module is bypassed (not used). All the interface functions are performed by the interface card even if a data module is present.
- The 7404D telephone with messaging cartridge cannot be used with PC Interface. However, the 7404D with PC cartridge can be used, but only with Group 1 configurations.

Wideband Switching

Wideband Switching provides the ability to dedicate 2 or more ISDN-PRI B-channels or DS0 endpoints for applications that require large bandwidth. It provides high-speed end-to-end communication between endpoints where dedicated facilities are not economic or appropriate. ISDN-BRI trunks do not support wideband switching.

Wideband Switching supports:

- High-speed video conferencing
- WAN disaster recovery
- Scheduled batch processing (for example, nightly file transfers)
- LAN interconnections and imaging
- Other applications involving high-speed data transmission, video transmission, or high bandwidth

Detailed description

ISDN-PRI divides a T1 or E1 trunk into 24 (32 for E1) channels, where one channel is used for signaling, and all others for standard narrowband communication. Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya Communication Manager serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 Converter circuit packs are used for wideband switching at DS1 remote EPN locations. They are compatible with both a 24-channel T1 and 32-channel E1 facility (transmission equipment). They support circuit-switched wideband connections (NxDS0) and a 192 Kbps packet channel.

The following table provides information on Wideband Switching channel types.

Channel Type	Number of Channels (DSOs)	Data Rate
H0 (T1 or E1)	6 (grouped 4 (T1) or 5 (E1) quadrants of 6 B-channels each)	384 Kbps
H11 (T1 or E1)	24 (on T1 - all 24 B-channels, with the D-channel not used; on E1 - B-channels 1 to 15, and 17 to 25, and B-channels 26 to 31 unused)	1536 Kbps
H12 (E1 only)	30 (B-channels 1 to 15 and 17 to 31)	1920 Kbps
NxDS0 (T1)	2-24	128 to 1536 Kbps
NxDS0 (E1)	2-31	128 to 1984 Kbps

Channel allocation

For standard narrowband communication, ISDN-PRI divides a T1 or E1 trunk as follows:

- T1 trunks are divided into 23 information channels are 1 signaling channel
- E1 trunks are divided into 30 information channels, 1 signaling channel, and 1 framing channel

Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya Communication Manager serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 converters are used for wideband switching at remote locations.

Performed using one of the three allocation algorithms: fixed, flexible, or floating.

- Fixed allocation — Provides contiguous-channel aggregation. The starting channel is constrained to a predetermined starting point. (Used only for H0, H11, and H12 calls.)
- Flexible allocation — Allows a wideband call to occupy non-contiguous positions within a single T1 or E1 facility (NxDS0).
- Floating allocation — Enforces contiguous-channel aggregation. The starting channel is not constrained to a predetermined starting point (NxDS0).

Typical uses

A typical video application uses an ISDN-PRI interface to DS0 1 through 6 of the line-side facility. See Figure 193: Wideband Switching Video Application.

Figure 193: Wideband Switching Video Application

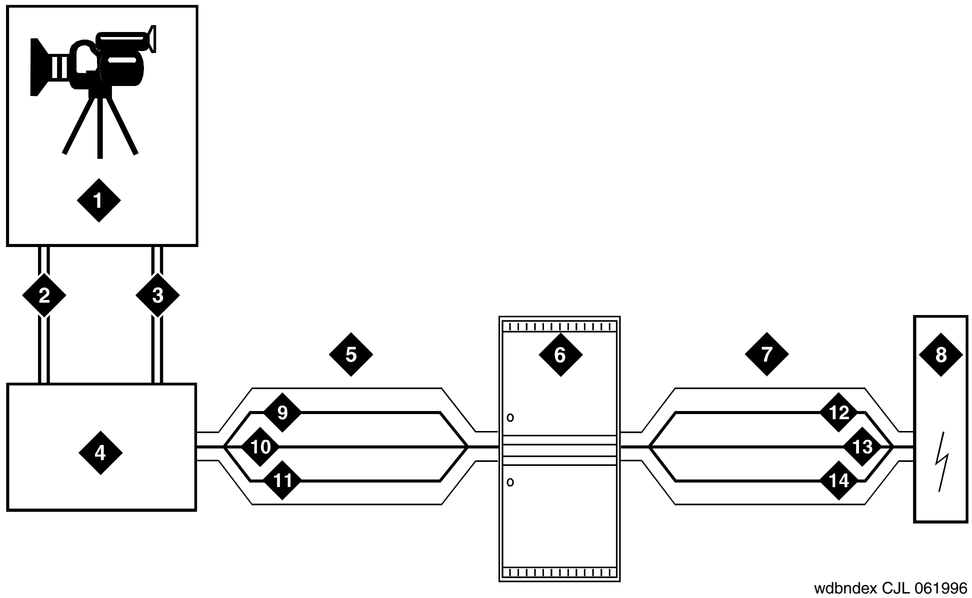


Figure notes:

1. **Video application**
2. **Port 1**
3. **Port 2**
4. **ISDN terminal adaptor**
5. **Line-side ISDN-PRI**
6. **Avaya S8XXX Server**
7. **ISDN trunk**
8. **Network**
9. **DS0 24 D-channel**
10. **DS0 23 unused**
11. **DS0 1 to 6 wideband**
12. **DS0 24 D-channel**
13. **DS0 7 to 23 narrow bands**
14. **DS0 1 to 6 wideband**

Endpoint applications

An endpoint application is the origination or destination of a wideband call. Endpoint application can be any number of data applications based on the customer’s particular needs.

ISDN-PRI terminal adapters

For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter. ISDN-PRI terminal adapters translate standard ISDN signaling into a form that can be used by the endpoint application and vice versa. The terminal adapter also must adhere to the PRI-endpoint boundaries as administered on Avaya Communication Manager when handling both incoming (to the endpoint) applications and outgoing calls.

Line-side (T1 or E1) ISDN-PRI facility

A line-side ISDN-PRI (T1 or E1) facility is comprised of a group of DS0s (24 for a T1 facility and 32 for an E1 facility). In this context, these DS0s are also called channels. T1 facilities have 23 B-channels and a single D-channel. E1 facilities have 30 B-channels, 1 D-channel, and a framing channel. Data flows bi-directionally across the facility between the server and the ISDN-PRI terminal adapter.

PRI-endpoints

A PRI-endpoint (PE) is a combination of DS0 B-channels on a line-side ISDN-PRI facility that has been assigned an extension.

A PRI-endpoint can support calls of lower bandwidth. In other words, a PE having a width 6 (six DS0s) can handle a call of one channel (64 Kbps) up to an including 6 channels. For example, an endpoint application connected to a PE defined as using B-channels 1 through 6 of an ISDN-PRI facility could originate a call using B-channels 1, 3, and 5 successfully. If the PE has been administered to use flexible channel allocation, the algorithm for offering a call to the PE starts from the first DS0 administered to the PE. Since only one active call is permitted on a PE, contiguous B-channels always are selected unless one or more B-channels are not in service.

One facility can support multiple separate and distinct PRI-endpoints (several extensions) within a single facility. Non-overlapping contiguous sets of DS0s (B-channels) are associated with each PE.

Universal digital signaling level 1 circuit pack

The UDS1 circuit pack is the interface for line-side and network facilities carrying wideband calls.

Non-signaling configuration

Wideband also can support configurations using non-signaling (non-ISDN-PRI) line-side T1 or E1 connections. The endpoints are the same as those defined for configurations with signaling.

Data service unit/channel service unit

This unit simply passes the call to the endpoint application. Unlike terminal adapters, the DSU/CSU does not have signaling capability.

Note:

No DSU/CSU is needed if the endpoint application has a fractional T1 interface.

Line-side (T1 or E1) facility

This facility, like the ISDN-PRI facility, is composed of a group of DS0s (24 for a T1 facility and 32 for an E1 facility; both T1 and E1 use 2 channels for signaling purposes). Line-side facilities are controlled solely from the server or Avaya S8XXX Server. Through the `access-endpoint` command, a specific DS0 or group of DS0s is assigned an extension. This individual DS0 or group, along with the extension, is known as a Wideband Access Endpoint (WAE).

Wideband access endpoint

WAEs have no signaling interface to the server or Avaya S8XXX Server. These endpoints simply transmit and receive wideband data when the connection is active.

Note:

Communication Manager can determine if the connection is active, but this does not necessarily mean that data is actually coming across the connection.

A WAE is treated as a single endpoint and can support only one call. If all DS0s comprising a wideband access endpoint are in service, then the wideband access endpoint is considered in service. Otherwise, the wideband access endpoint is considered out of service. If an in-service wideband access endpoint has no active calls on its DS0s, it is considered idle. Otherwise, the wideband access endpoint is considered busy.

Multiple WAEs are separate and distinct within the facility and endpoint applications must be administered to send and receive the correct data rate over the correct DS0s. An incoming call at the incorrect data rate is blocked.

Guidelines and examples

This section examines wideband and its components in relation to the following specific customer usage scenarios:

- Data backup connection
- Scheduled batch processing
- Primary data connectivity
- Networking

Data backup connection

Using wideband for data transmission backup provides customers with alternate transmission paths for critical data in the event of primary transmission path failure.

Scheduled batch processing

Scheduled batch processing applications are used for periodic database updates (for example, retail inventory) or distributions (for example, airline fare schedules). These updates are primarily done after business hours and are often referred to as nightly file transfers. Wideband meets the high bandwidth requirements at low cost for scheduled batch processing. In addition, wideband allows the dedicated-access bandwidth for busy-hour switching traffic to be used for these applications after business hours; no additional bandwidth costs are incurred.

The non-ISDN backup data connection is also appropriate for scheduled batch processing applications. Administered Connections are used to schedule daily or weekly sessions originating from this application.

Primary data connectivity

Permanent data connections (those always active during business hours), such as interconnections between LANs, are well suited for Avaya Communication Manager when ISDN-PRI endpoints are used. The ISDN end-to-end monitoring and the endpoint's ability to react to failures provide for critical data availability needs. With ISDN, endpoints can detect network failures and initiate backup connections through the server running Communication Manager; ISDN endpoints can also establish additional calls when extra bandwidth is needed.

Any failures not automatically restored by Avaya Communication Manager are signaled to the endpoint application, which can initiate backup data connections over the same PRI endpoint. Communication Manager routes the backup data connections over alternate facilities if necessary.

Networking

All of the wideband networking is over ISDN-PRI facilities (and the emulation of them by ATM-CES) but might connect to a variety of networks, other domestic interexchange carriers' services, private line, RBOC services, and services in other countries.

ISDN-PRI trunk groups and channel allocation

Only ISDN-PRI trunks (and the emulation of them by ATM-CES) support wideband calls to the network. Wideband's bandwidth requirements have necessitated modification of the algorithms by which trunks look for idle channels. The following section describes the search methods and their relationship to the available wideband data services.

Facility lists

The system always sends a wideband call over a single trunk group and a single DS1 facility (or other ISDN-PRI-capable facility). Since a trunk group can contain channels (trunk members) from several different DS1 facilities, the system maintains a facility list for each trunk group.

A facility list orders the trunk members based on signaling group. If the system is using non-facility associated signaling groups with multiple DS1 facilities, the system sorts trunk members in that signaling group according to the interface identifier assigned to the corresponding DS1 facility.

When searching for available channels for a wideband call placed over a given trunk group, the system starts with the channels in the lowest-numbered signaling group with the lowest interface identifier. If the system cannot find enough channels in a given signaling group with that interface identifier, it checks the next higher interface identifier. If no more interface identifiers are available in the current signaling group, the system moves its search to the channels in the next higher signaling group.

For example, if three facilities having signaling group/interface identifier combinations of 1/1, 1/2, and 2/1 were associated with a trunk group, then a call offered to that trunk group would search those facilities in the order as they were just listed. Also note that since trunks within a given facility can span several trunk groups, a single facility can be associated with several different trunk groups.

Given this facility list concept, the algorithms have the ability to search for trunks, by facility, in an attempt to satisfy the bandwidth requirements of a given wideband call. If one facility does not have enough available bandwidth to support a given call, or it is not used for a given call due to the constraints presented in the following section, then the algorithm searches the next facility in the trunk group for the required bandwidth (if there is more than one facility in the trunk group).

In addition to searching for channels based on facilities and required bandwidth, Port Network (PN) preferential trunk routing is also employed. This PN routing applies within each algorithm at a higher priority than the constraints put on the algorithm by the parameters listed later in this section. In short, all facilities that reside on the same PN as the originating endpoint are searched in an attempt to satisfy the bandwidth of a given call, prior to searching any facilities on another PN.

Direction of trunk/hunting within facilities

You can tell the system to search for available channels in either ascending or descending order. These options help you reduce glare on the channels because the system can search for channels in the opposite direction to that used by the network. If an ISDN trunk group is not optioned for wideband, then a cyclical trunk hunt based on the administration of trunks within the trunk group is still available.

H11

When a trunk group is administered to support H11, the algorithm to satisfy a call requiring 1,536 Kbps of bandwidth uses a fixed allocation scheme. That is, the algorithm searches for an available facility using the following facility-specific channel definitions:

- T1: H11 can only be carried on a facility without a D-channel being signaled in an NFAS arrangement (B-channels 1-24 are used).
- E1: Although the 1,536 Kbps bandwidth could be satisfied using a number of fixed starting points (for example, 1, 2, 3, and so forth), the only fixed starting point being supported is 1. Hence, B-channels 1-15 and 177-25 always are used to carry an H11 call on an E1 facility.

If the algorithm cannot find an available facility within the trunk that meets these constraints, then the call is blocked from using this trunk group. In this case, the call can be routed to a different trunk group preference via Generalized Route Selection (GRS), at which time, based on the wideband options administered on that trunk group, the call would be subject to another hunt algorithm (that is, either the same H11 algorithm or perhaps an N x DS0 algorithm described in a later paragraph).

Note that on a T1 facility, a D-channel is not considered a busy trunk and results in a facility with a D-channel always being partially contaminated. On an E1 facility, however, a D-channel is not considered a busy trunk because H11 and H12 calls can still be placed on that facility; an E1 facility with a D-channel and idle B-channels is considered an idle facility.

H12

Since H12 is 1,920 Kbps which is comprised of 30 B-channels, a 1,920-kbps call can only be carried on an E1 facility. As with H11, the hunt algorithm uses a fixed allocation scheme with channel 1 being the fixed starting point. Hence, an H12 call always is carried on B-channels 1 to 15 and 17 to 31 on an E1 facility (as shown in the following table). When offered any other call (other than a 1,536-kbps call), the algorithm behaves as it does when H11 is optioned.

		DS0s Comprising Each Channel	
Facility	**ISDN Interface**	**H11**	**H12**
T1 T1	23B + D 24B (NFAS)	- 1-24	- -
E1 E1	30B + D 31B (NFAS)	1-15, 17-25 1-15, 17-25	1-15, 17-31 1-15, 17-31

H0

When a trunk group is administered to support H0, the algorithm to satisfy a call requiring 384 Kbps of bandwidth also uses a fixed allocation scheme. Unlike the H11 fixed scheme which only supports a single fixed starting point, the H0 fixed scheme supports 4 (T1) or 5 (E1) starting points. The H0 algorithm searches for an available quadrant within a facility based on the direction of trunk or hunt administered. If the algorithm cannot find an available quadrant within any facility allocated to this trunk group, then the call is blocked from using this trunk group. Again, based on GRS administration, the call might route to a different trunk group preference and be subject to another algorithm based on the wideband options administered.

Note that a D-channel is considered a busy trunk and results in the top most quadrant of a T1, B-channels 19 to 24, always being partially contaminated. This is *not true* for NFAS.

If this H0 optioned trunk group is also administered to support H11, H12, or N x DS0, then the system also attempts to preserve idle facilities. In other words, when offered a narrowband, H0, or N x DS0 call, the system searches partially-contaminated facilities before it searches to idle facilities.

N x DS0

For the N x DS0 multi-rate service, a trunk group parameter determines whether a floating or a flexible trunk allocation scheme is to be used. The algorithm to satisfy an N x DS0 call is either floating or flexible.

- Floating (Contiguous) — In the floating scheme, an N x DS0 call is placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without any constraint being put on the starting channel (that is, no fixed starting point trunk).

- Flexible — In the flexible scheme, an N x DS0 call is placed on any set of B-channels as long as the requested bandwidth is satisfied. There is absolutely no constraint such as contiguity of B-channels or fixed starting points. Of course, as with all wideband calls, all the B-channels comprising the wideband call must reside on the same ISDN facility.

  Regardless of the allocation scheme employed, the N x DS0 algorithm, like the H11 and H12 algorithms, attempts to preserve idle facilities when offered B, H0, and N x DS0 calls. This is important so that N x DS0 calls, for large values of N, have a better chance of being satisfied by a given trunk group. However, if one of these calls cannot be satisfied by a partially-contaminated facility and an idle facility exists, a trunk on that idle facility is selected, thus contaminating that facility.

  There are additional factors to note regarding specific values of N and the N x DS0 service:

  - N = 1 — this is considered a narrowband call and is treated as any other voice or narrowband-data (B-channel) call.
  - N = 6 — if a trunk group is optioned for both H0 and N x DS0 service, a 384-kbps call offered to that trunk group is treated as an H0 call and the H0 constraints apply. If the H0 constraints cannot be met, then the call is blocked.
  - N = 24 — if a trunk group is optioned for both H11 and N x DS0 service, a 1,536-kbps call offered to that trunk group is treated as an H11 call and the H11 trunk allocation constraints apply.
  - N = 30 — if a trunk group is optioned for both H12 and N x DS0 service, a 1,920-kbps call offered to that trunk group is treated as an H12 call and the H12 trunk allocation constraints apply.

Glare and blocking

Glare prevention

Glare occurs when both sides of an ISDN interface select the same B-channel for call initiation. For example, a user side of an interface selects the B-channel for an outgoing call and, before Communication Manager receives and processes the SETUP message, the server also selects the same B-channel for call origination. Since any single wideband call uses more channels, the chances of glare are greater. With proper and careful administration, glare conditions can be reduced.

To reduce glare probability, the network needs to be administered so both sides of the interface select channels from opposite ends of facilities. This is called linear hunting, ascending or descending. For example, on a 23B+D trunk group, the user side could be administered to select B-channels starting at channel 23 while the network side would be administered to start selecting at channel 1. Using the same example, if channel 22 is active but channel 23 is idle, the user side should select channel 23 for re-use.

Blocking prevention

Blocking occurs when insufficient B-channels required to make a call are available. Narrowband calls require only one channel so blocking is less likely than with wideband calls which require multiple B-channels. Blocking also occurs for wideband calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible).

To reduce blocking, Communication Manager selects trunks for both wideband and narrowband calls to maximize availability of idle fixed channels for H0, H11, and H12 calls and idle floating channels for N x DS0 calls that require a contiguous bandwidth. The strategy for preserving idle channels to minimize blocking depends on the channel type. The chances for blocking are reduced if you use a flexible algorithm, assuming it is supported on the other end.

Channel Type	Blocking Minimization Strategy
H0	Preserve idle quadrants
H11	Preserve idle facilities
H12	Preserve idle facilities
Flexible NxDS0	Preserve idle facilities
Floating NxDS0	Preserve idle facilities as first priority

Administering Wideband Switching

Before you start, you need a DS1 Converter circuit pack.

To administer wideband switching:

1. On the **Access Endpoint** screen, administer all fields.

   See Access Endpoint on page 745 for more information.
2. On the **PRI Endpoint** screen, administer all fields.

   See PRI Endpoint on page 1418 for more information.
3. On the **ISDN Trunk Group** screen, administer all fields.

   See Trunk Group on page 1669 for more information.
4. On the **Route Pattern** screen, administer all fields.

   See Route Pattern on page 1444 for more information.

Considerations

- For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter.

Interactions

- Administered Connections

  Provides call initiation for WAEs. All Administered Connections that originate from WAEs use the entire bandwidth administered for WAE. The destination of an Administered Connection can be a PRI endpoint.

- Automatic Circuit Assurance

  Treats wideband calls as logical single-trunk calls so that a single ACA-referral call is made if an ACA-referral call is required. The call is referred to the lowest B-channel associated with the wideband call.

- Call Coverage

  A wideband endpoint extension cannot be administered as a coverage point in a call-coverage path.

- Call Detail Recording

  When CDR is active for the trunk group, all wideband calls generate CDR records. The feature flag indicates a data call and CDR records contain bandwidth and BCC.

- Call Forwarding

  You must block Call Forwarding through Class of Service.

- Call Management System and Basic Call Management System

  Wideband calls can be carried over trunks that are measured by CMS and BCMS. Wideband endpoints are not measured by CMS and BCMS.

- Call Vectoring

  PRI endpoints can use a vector-directory number (VDN) when dialing. For example, PRI endpoint 1001 dials VDN 500. VDN 500 points to Vector 1. Vector 1 can point to other PRI endpoints such as route-to 1002, or route-to 1003, or busy.

  Call Vectoring is used by certain applications. When an incoming wideband call hunts for an available wideband endpoint, the call can route to a VDN, that sends the call to the first available PRI endpoint.

- Class of Restriction

  COR identifies caller and called-party privileges for PRI endpoints. Administer the COR so that account codes are not required. Forced entry of account codes is turned off for wideband endpoints.

- Facility Busy Indication

  You can administer a busy-indicator button for a wideband-endpoint extension, but the button does not accurately track endpoint status.

- Facility Test Calls

  You can use Facility Test Calls to perform loop-back testing of the wideband call facility.

- Generalized Route Selection

  GRS supports wideband BCC to identify wideband calls. GRS searches a route pattern for a preference that has wideband BCC. Route preferences that support wideband BCC also can support other BCCs to allow different call types to share the same trunk group.

- CO Trunk (TTC - Japan) Circuit Pack

  This circuit pack cannot perform wideband switching. No member of the circuit pack should be a member of a wideband group.

CallVisor Adjunct-Switch Applications Interface

CallVisor Adjunct-Switch Applications Interface (ASAI) links Avaya Communication Manager and adjunct applications. The interface allows adjunct applications to access switching features and supply routing information to Communication Manager. CallVisor ASAI improves Automatic Call Distribution (ACD) agents' call handling efficiency by allowing an adjunct to monitor, initiate, control, and terminate calls on the Avaya S8XXX Server. The CallVisor ASAI interface can be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/messaging applications.

CallVisor ASAI is supported by two transport types. These are:

1. Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) transport (CallVisor ASAI-BRI)
2. LAN Gateway Transmission Control Protocol/Internet Protocol transport (Avaya LAN Gateway).

CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services. The Q.932 Facility Information Element (FIE) carries the CallVisor ASAI requests and responses across the interface. An application program can access CallVisor ASAI services by supporting the ASAI protocol or by using a third-party vendor application programming interface (API).

For a simple ASAI configuration example, see Figure 194.

Figure 194: ASAI Switch Interface Link — BRI Transport

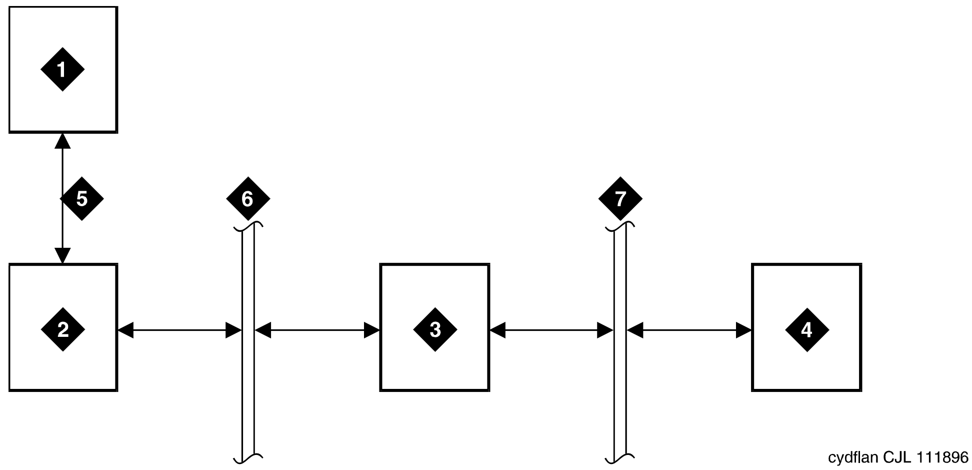


Figure notes:

1. **ASAI adjunct**
2. **ISDN Line circuit pack**
3. **Packet Controller circuit pack**
4. **Switch processing element (SPE)**
5. **ISDN-BRI**
6. **Packet bus**
7. **Memory bus**

ASAI Capabilities

For information concerning the types of associations over which various event reports can be sent, see *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Considerations

- If your system has an expansion cabinet (with or without duplication), ASAI resources should reside on the system's Processor Cabinet.

Interactions

See *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Setting up CallVisor ASAI

CallVisor Adjunct-Switch Applications Interface (ASAI) can be used in the telemarketing and help-desk environments. It is used to allow adjunct applications to monitor and control resources in Avaya Communication Manager.

Before you start

- On the **System Parameters Customer-Options (Optional Features)** screen, verify the:
  - **ASAI Link Core Capabilities** field is **y**. If not, contact your Avaya representative.
  - **Computer Telephony Adjunct Links** field is **y** if the adjunct is running the CentreVu Computer Telephony.

Instructions

To set up CallVisor ASAI:

1. Type **`add cti-link`** ***`nn`***, where ***`nn`*** is a number between **1** and **64**. Press **Enter**.

   The **CTI Link** screen appears.

Figure 195: CTI Link screen when Type field is ASAI or ADJLK

```
add cti-link next                                                 Page 1 of x
                                  CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI
     Port: 1C0501                                                    COR: 1
     Name: ASAI CTI Link 1









BRI OPTIONS
               XID? y     Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

2. In the **Type** field, type
   - **asai** if this adjunct platform is other than CentreVu Computer Telephony, for example, IBM CallPath.
   - **adjlk** (Computer Telephony adjunct link) if this is for the CentreVu Computer Telephony using the Telephony Services Application Programming Interface (TSAPI).
3. In the **Port** field, use the port address assigned to the LAN Gateway Interface circuit pack.
4. Press **Enter** to save your changes.

Chapter 16: Administering Avaya Servers

This chapter describes how to administer an Avaya S8XXX Server and an Avaya G700 Media Gateway. It is targeted for system administrators after the product is installed and tested. In a converged network where voice and data are both sent over a corporate local area network (LAN), this configuration can provide primary or standby telephony and communications-processing capabilities.

Users should have broad data networking experience with data products and technology to best understand this product. An in-depth knowledge of the call-processing engine of Avaya Communication Manager and of the Avaya P330 switches is also recommended.

Overview

To set up and maintain your Avaya S8XXX Server with a G700 Media Gateway, you need to administer:

- the G700 Media Gateway and its internal processors, typically using the P330 Device Manager Web-based tool or a command-line interface (CLI)
- the Avaya S8XXX Server using the Server Web Interface
- call-processing features using Avaya Communication Manager

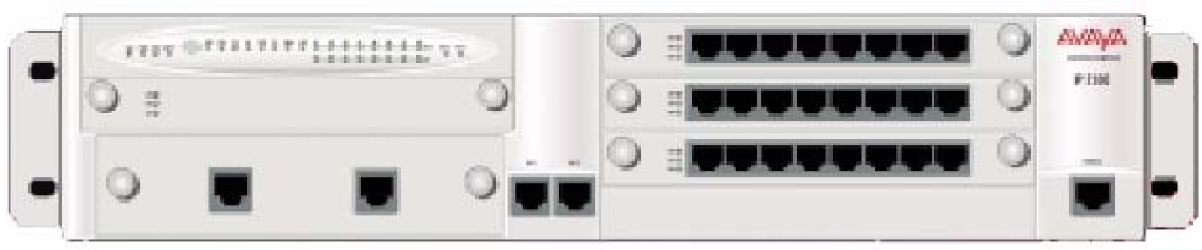
Administering the G700 Media Gateway

The hardware elements of the G700 Media Gateway are summarized in this section. For details on any of these hardware components or on different configurations, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

G700 Media Gateway physical design

The G700 Media Gateway is a19-inch, rack-mount design similar to other Avaya P330 hardware. The media gateway can be a member of a P330 stack of Layer 2 or 3 devices, reside in a stack of other media gateways, or operate as a standalone unit. Figure 196 shows the front of the media gateway.

Figure 196: The G700 Media Gateway (front view)



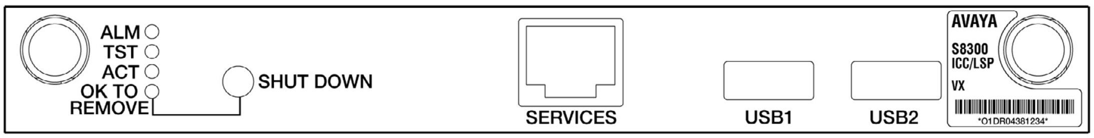
Key components of the G700 Media Gateway include:

1. LED Status Panel (top left) for the media gateway
2. S8300 Server slot (V1). The G700 Media Gateway does not have an S8300 Server in its basic configuration.
3. P330 Expansion slot (bottom left) for adding modules such as a 16-port Ethernet expansion module, fiber interfaces, ATM access or WAN access
4. Dual 10/100Base-T Ethernet Ports for connection to the corporate LAN (bottom center of chassis)
5. Up to three additional media modules in slots V2 to V4 on the right side of the chassis (slots are numbered from top to bottom)
6. 8-Pin RS-232 Serial Port (Console) for direct-connect P330 stack processor administration (typically done by Services personnel)

S8300 Server faceplate

(For S8300 Servers) Figure 197 shows the faceplate of the S8300 Server.

Figure 197: S8300 Server faceplate



The faceplate of the S8300 Server has the following components:

- The LED array on the left indicates when the S8300 is active and when it is safe to power it down. The green LED indicates that Avaya Communication Manager is running.
- A Shut Down button can be used to shut down the server (the S8300 can also be shut down and restarted using software commands).
- A Services Ethernet interface provides a direct connection to a laptop computer connected with a crossover cable. Normally only Services technicians use this interface; most ongoing S8300 administration is done using the Ethernet connection to the corporate LAN on the media gateway.
- One of the two USB connections can be used to attach an external modem, primarily used to report alarms to a Services support agency.
- The label on the right provides identification information for the S8300 Server.

G700 Media module description

(For S8300 Servers) The four slots on the G700 Media Gateway are populated with media modules according to the needs at your site. Up to four slots can be filled as follows:

- The S8300 Server is a special type of media module that hosts the call-processing (telephony) software. If present, the S8300 is installed in slot one (V1); only one can be installed per media gateway. The S8300 can act as a primary call controller for the system, or one or more S8300s can be configured to provide standby service for a primary call controller if needed. See Configuring the Local Survivable Processor on page 584 for details.
- The other media module slots are filled as needed to meet the call-processing needs at your site. Media modules can be mixed and matched. Table 1 lists the administrative name and port capacities for the media modules that can be installed in a G700 Media Gateway.

Table 8: Media Module names and port capacities

Media module type	Administration name	Number of ports
Analog Line/Trunk	analog	8
DCP Telephone	dcp	8
T1/E1	ds1	24/32 (T1=24 in robbed-bit signalling mode and E1=32 for 1 control, 1 signal and 30 voice)
VoIP	voip	N/A
S8300 Server	icc	N/A
MM270	bri	8

Configuring the Local Survivable Processor

An Avaya S8XXX Server can be configured either as the primary call-processing controller, or as a Local Survivable Processor (LSP). An LSP can take over call processing if the primary call-processing system (such as another Avaya server) is unavailable for any reason (such as a network failure or server problem). The Avaya S8XXX Server can be either the primary or LSP server; it is set up to operate as a primary or standby LSP server during the configuration process using the Server Web Interface. The license file determines the mode that the server runs in and the **Configure Server** Web page provides supplementary instruction.

If the Avaya S8XXX Server loses contact with its G700 Media Gateway, the media gateway retains its last status until the Link Loss Delay Timer (LLDT) expires. (The default for the LLDT is 5 seconds, but this interval is administrable using the **Link Loss Delay Timer (minutes)** field on the **IP-Options System Parameters** screen.) Once the LLDT expires, the system removes all boards and deletes all call processing information. However, if the G700 loses contact with the Avaya S8XXX Server, the media gateway first tries to reconnect for a period of one minute. If this fails, then the G700 then tries to connect with another server in its controller list. If the primary server was a LSP, it will start looking at the top of its MGC list in order to get back to the primary server. Otherwise, it starts down the list of alternative servers. When a functional Avaya S8XXX Server is located, the media gateway informs the server of its current call state, and the server maintains those connections until the users hang up.

If the primary call-processing server goes offline and an LSP is available as a standby unit, it will assume call processing as follows:

- IP telephones and media gateways that were previously using the primary server will try to register with the standby server for call processing, provided that they have been administered to do so in the controller list (use the `set mgc list` command).
- The standby server (LSP) will go into license error mode, then start to provide call processing. It cannot preserve any calls set up by the primary server. IP telephone connections can stay up until the call is completed if they are shuffled, but no features are supported on the call.

  Note:

  The license error mode runs for up to 30 days, and if the problem is not resolved, the system goes into No License Mode and administration and some commands are restricted.
- When the primary server is available again, it will begin handling call processing. However, those endpoints that are now using the standby server will continue to do so until the standby unit is rebooted.
- If the standby server is rebooted, all devices will return to using the primary server for call-processing service. Any calls in progress on the LSP will be dropped when the reboot occurs (the change back to the primary server is not call preserving).
- With LSP functionality, there is full functionality and feature support.

Using Device Manager to administer G700 Media Gateway components

Device Manager, also known as the P330 Embedded Web Manager, provides a browser-based graphical user interface (GUI) to assist you with ongoing media gateway administration.

Device Manager allows you to:

- View the status of different devices on the network.
- Configure or modify devices including Virtual LAN (VLAN) groupings, port mirroring, and Simple Network Management Protocol (SNMP) traps.

Accessing Device Manager from a Web browser

To access the P330 Device Manager from a Web browser:

1. Open a compatible Web browser on your computer. Microsoft Internet Explorer 5.0 (or higher) and Netscape Navigator 4.7 and 6.2 are supported. The Java Plug-in 1.3.1_02 is required.
2. In the **Address (or Location)** field of your browser, type the IP address of the P330 stack.

3. Log in as prompted.
   - The default user name is: **root**
   - The default password for read-write access is: **root**

   Note:

   You should change the default passwords after you log in, if they have not already been changed. The passwords apply to all logins for these devices, whether accessed through Device Manager or a CLI.
4. The **Welcome** screen is displayed. Proceed with P330 and media gateway device administration.

Accessing Device Manager through Network Management Console with VoIP SystemView

You can alternatively launch the P330 Device Manager from the Avaya Network Management Console with VoIP SystemView. This optional product is a complete Network Management System (NMS). Services include:

- viewing all network devices by type, subnet, or customized groupings
- logging and viewing SNMP traps and events
- launching and managing other applications including Avaya Site Administration

Contact your Avaya representative to obtain a copy of this program if desired.

Administering the command line interface

Instead of using Device Manager, you can access the server's command line interface using an SSH client, like PuTTY, and an IP address of 192.11.13.6., or use the serial interface on the front of the chassis to establish a direct connection to the P330 stack processor.

- Command line interface (CLI) access procedures are covered in *Welcome to the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*, 555-234-200.
- For a list of CLI commands, see the *Maintenance for the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*, 555-234-101.

SNMP alarms are different from server hardware- or software-generated Operations Support System (OSS) alarms that are recorded in the server logs, and might be reported to Avaya's Initialization and Administration System (INADS) or another services support agency over the server's optional modem interface or through SNMP notifications. Either method, both, or no alarm-reporting method might be used at a given site.

Administering the Avaya S8XXX Server

The Avaya S8XXX Server contains the call-processing software of Avaya Communication Manager, and controls its operation over the corporate network. Some of the primary functions controlled by the Avaya S8XXX Server are:

- Backing up and restoring call processing, server, and security data.
- Checking server and process status.
- Administering network features for the server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services, and installing license and authentication files.
- Installing new software and reconfiguring the server as needed.
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs, and running tests if needed.

Accessing the Avaya S8XXX Server Web Interface

Avaya S8XXX Server tasks are performed using the Server Web Interface. This browser-based server administration tool is an easy-to-use graphical user interface for performing the server administration tasks listed above. It contains an extensive help system that describes all Web screens and Avaya S8XXX Server procedures. This section covers highlights of Avaya S8XXX Server administration.

The Web Interface can be accessed through the corporate LAN connection to the G700 Media Gateway, or through the Services Ethernet interface on the front of the Avaya S8XXX Server connected to a laptop PC using a crossover cable. Details on how to configure a laptop for a direct connection are in the online help and in *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway*.

To access the Avaya S8XXX Server, you must log in as follows:

1. Open a compatible Internet browser on your computer. Currently only Internet Explorer 5.x (Avaya recommends 5.5 with Service Pack 2) and Netscape 4.7x are supported.
2. In the **Address (or Location)** field of your browser, type the IP address or name of the Avaya S8XXX Server. Press **Enter**.
   - LAN access by IP address. If you are logging into the administrative interface over the corporate local area network, you can type the Avaya S8XXX Server's unique IP address in standard dotted-decimal notation, such as http://192.152.254.201.
   - LAN access by server name. If the server has already been configured and if the corporate LAN includes a domain name service (DNS) server that has been administered with the servers' names, you can type the server's name into the address field instead of the IP address. Server names vary depending on local administration (such as http://media-server1.mycompany.com).

- Laptop access by IP address. If you are logging in to the services Ethernet interface from a directly connected laptop, the IP address on the server is always **192.11.13.6**. The subnet mask is always **255.255.255.252**. New servers that have not yet been configured can only be accessed in this way. Server-name login is *not* available through the services interface because this connection is a closed (private) network with no DNS. The name and IP address of the Avaya S8XXX Server are specified during initial server configuration.

Note:

If your browser does not have a valid security certificate, you will see a warning screen and instructions to load the security certificate. If you are certain your connection is secure, accept the server security certificate to access the **Logon** screen. If you plan to use this computer and browser to access this or other Avaya S8XXX Servers again, click the main menu link to **Install Avaya Root Certificate** after you log in.

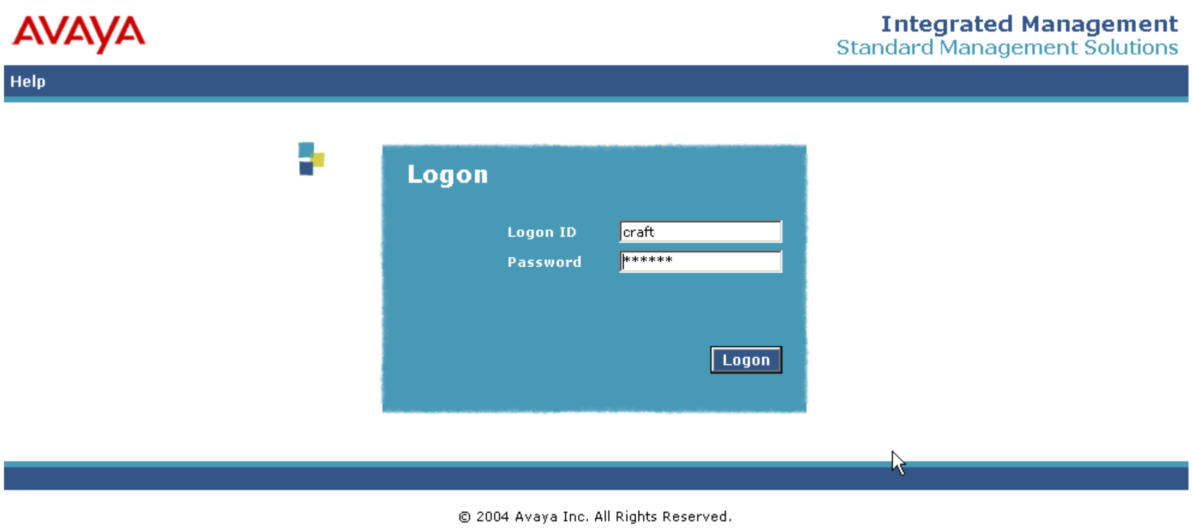
The system displays the **Logon** screen.

Figure 198: Integrated Management Logon screen

3. In the **Logon ID** field, type your user name, such as **cust**. Remember that user names and passwords are case sensitive. Enter the login ID and confirmation information in upper or lower case as required. Click the **Logon** button or press **Enter**.

   The system redisplays the **Logon** screen with a **Password** field.

Figure 199: Integrated Management Logon/Password screen



4. Type your password in the **Password** field, and click **Logon** or press **Enter**.

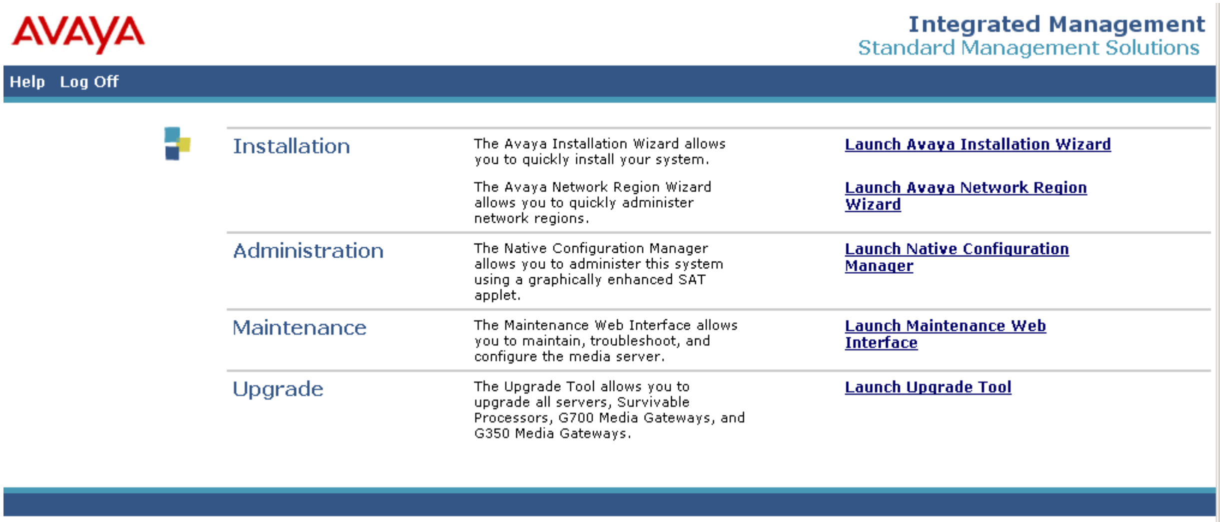
   Note:

   If your login is Access Security Gateway (ASG) protected, you will see an ASG challenge screen. Enter the correct response and click **Logon** or press **Enter**.

   The server matches your login information against authentication tables. If the server cannot recognize your user name or password, you receive an authentication failure message. Return to step 4. If you fail to enter the user name and login confirmation correctly 4 times within a few minutes, the **Logon** screen will block further attempts to log on for a while.

   After successful authentication, the system displays the Integrated Management Standard Management Solutions (SDS) home page.

Figure 200: Integrated Management SMS home page



Accessing the Maintenance Web Interface

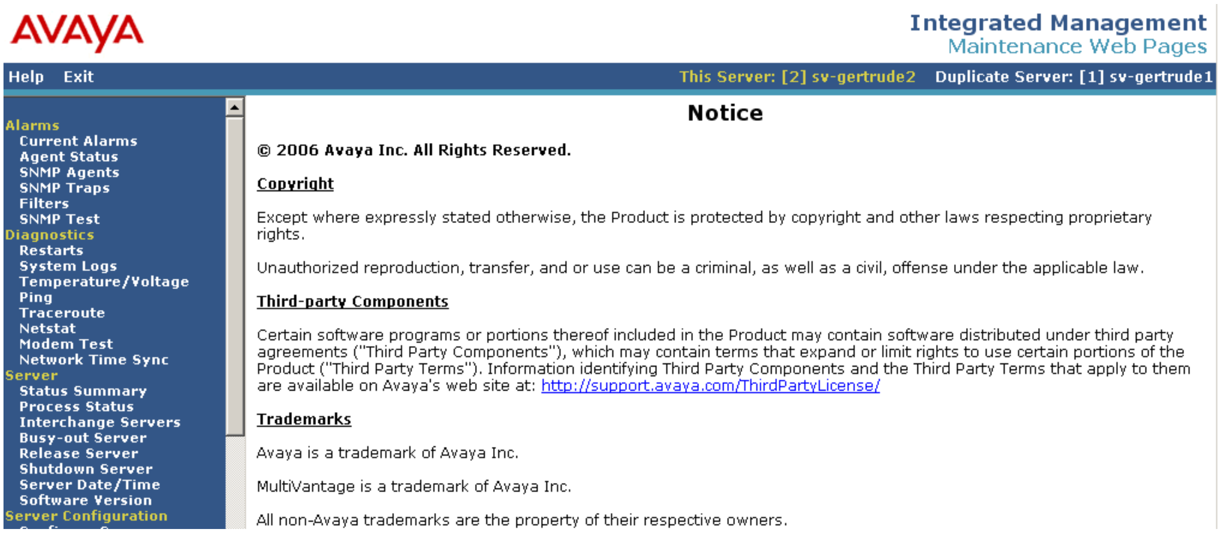
The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the Avaya S8XXX Server.

To access the Maintenance Web Pages:

1. From the Integrated Management SMS home page (see Figure 200), click **Launch Maintenance Web Interface**.

   The system displays the **Maintenance Web Pages** home page.

Figure 201: Maintenance Web Pages home page



The tasks you can perform are shown by a list of links in the panel on the left side of the screen.

For help with any of these tasks, click **Help** on this home page. Click **Help** on any of the pages accessed b y the links to go directly to the help for that specific screen.

Avaya S8XXX Server Web interface tasks

Key tasks that administrators typically perform on Avaya S8XXX Servers are summarized in this section. See online help for more detailed information.

Backing up and restoring data

One of the most important tasks is to set up a schedule to routinely back up system data to a safe location. Because the S8300 Server does not include a media storage device for backup purposes, the data must be emailed or transferred using file transfer protocol (FTP) to another server or computer for backup. The S87XX Server has a local removable storage device.

The Web interface allows you to back up call-processing data (Avaya Communication Manager "translations"), server system data, and security files. Avaya recommends that you encrypt the backup files to keep this sensitive information secure.

Upgrading software and configuration

Occasionally you might need to install new software on the Avaya S8XXX Server. The new software (and a new license file if required) must be copied to the server using one of the methods listed in Copying files to the server on page 592. The software installation process uses wizard-like screens on the Web to guide you through the process.

You might also need to update your server configuration from time to time, and reverify it after a software upgrade. IP addresses for the server and its gateway to the corporate LAN, or for the optional UPS unit, DNS servers, and modem, are specified using configuration wizard-like screens. The wizard-like screens also allow you to specify static network routes for sending information over the network, and to update or change the method by which the server keeps time.

Copying files to the server

Files must be copied to the Avaya S8XXX Server from another computer or server in the network, or uploaded from a directly connected laptop computer. Files that might be copied to the server include license and authentication files, system announcements, and files for software upgrades. Files can be copied to the server using one of the following methods:

- Upload Files to Server (via browser) link to upload one or more files from your computer to the server's FTP directory using HTTP protocol.
- Download Files to Server (from Web) link to copy files to the server from another server on the network; it works like the **Upload Files** screen.
- Transfer files from another computer or server accessible from the corporate network using FTP or Trivial FTP (TFTP). Files must be transferred in binary mode. Either a GUI or CLI FTP program can be used, depending on what is available on your computer.

Setting up SNMP

You can set up Simple Network Management Protocol (SNMP) services on the server to provide a means for a corporate NMS to monitor the server, and send alarm notifications to a services agency, to a corporate NMS, or both. For more information on administering SNMP, see Administering SNMP.

To activate SNMP alarm notification for devices, use the **SNMP Traps** screen to set up SNMP destinations in the corporate NMS. SNMP traps for other devices on the network can be administered using Device Manager. See Using Device Manager to administer G700 Media Gateway components on page 585.

Note:

UDP port 162 for snmptrap must be "opened" to allow reception of traps (from media gateways) and transmission of traps to your trap receiver. Certain trap categories from media gateways must be administered "on" by media gateway administration. Use media gateway commands **`set snmp trap enable auth`** and **`tcp syn-cookies`** for this. For more information on media gateways, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431 and *Maintenance Procedures for Avaya Communication Manager, Media Gateways and Servers*, 03-300432.

Setting up Processor Ethernet

Much like a CLAN board, Processor Ethernet (PE) provides connectivity to IP endpoints, gateways, and adjuncts. The PE interface is a logical connection in the Communication Manager software that uses a port on the NIC in the server. There is no additional hardware needed to implement Processor Ethernet, but the feature must be enabled via license file. Type **`display system-parameters customer-options`** to verify that the **Processor Ethernet** field on the **System Parameters Customer-Options (Optional Features)** is set to **y**. If this field is not set to **y**, contact your Avaya representative.

During the configuration of a server, the PE is assigned to a Computer Ethernet (CE). The PE and the CE share the same IP address but are very different in nature. The CE interface is a native computer interface while the PE interface is the logical appearance of the CE interface within Communication Manager software. The interface that is assigned to the PE can be a control network or a corporate LAN. The interface that is selected determines which physical port the PE uses on the server.

Note:

The PE interface is enabled automatically on an LSP or an ESS server. Do not disable the PE interface on an LSP or an ESS server. Disabling the PE interface disables the LSP or ESS server's ability to register with the main server. The LSP or ESS server will not work if the PE interface is disabled.

Three adjuncts are supported for connectivity to the PE interface of an LSP or a simplex ESS server. The three adjuncts are Call Management System (CMS), Call Detail Recording (CDR), and Application Enablement Services (AE Services).

High-level steps to setting up the PE interface

This section contains general, high-level steps for configuring and administering the PE interface. As each system may have unique configuration requirements, contact your Avaya representative if you have questions.

1. Load the Communication Manager software.
2. Load the license file.
3. Configure the PE interface on the server using the server's maintenance Web interface:
   a. Select the interface that will be used for PE in the **Set Identities** page. The **Set Identities** page can be found on the server's maintenance Web interface under **Server Configuration > Configure Server**.
   b. If this is an ESS server or an LSP, enter the additional information in the **Configure LSP or ESS** screen:
      - **Registration address at the main server** field. Enter the IP address of a CLAN or PE interface on the main server to which the LSP or ESS server will connect. The IP address is used by the LSP or ESS server to register with the main server. In a new installation, where the LSP or the ESS server has not received the initial translation download from the main server, this address will be the only address that the LSP or the ESS server can use to register with the main server.
      - **File synchronization address of the main cluster**: Enter the IP address of a server's NIC (Network Interface Card) connected to a LAN to which the LSP or the ESS server is also connected. The ESS server or the LSP must be able to ping to the address. Consideration should be given to which interface you want the file sync to use. Avaya recommends the use of the customer LAN for file sync.
4. On the Communication Manager System Access Terminal (SAT), enter the name for each ESS server, LSP, and adjunct in the IP Node Names screen. The SAT command is **`change node-name`**. You do not have to add the PE interface (**procr**) to the **IP Node Names** screen. Communication Manager adds the PE interface automatically.
5. For a simplex main server, use the IP Interfaces screen to enable H.248 gateway registration, H.323 endpoint registration, gatekeeper priority, network regions, and target socket load. On some platform types, the **IP Interfaces** screen is already configured. Use the SAT command **`display ip-interfce procr`** to see if the PE interface is already configured. If it is not, use the SAT command **`add ip-interface procr`** to add the PE interface.
6. Use the Processor Channel Assignment screen (command **`change communication-interface processor-channels`**) and the IP Services screen (**`change ip-sevices`**) to administer the adjuncts that use the PE interface on the main server:
   - Enter **p** in the **Interface Link** field on the **Processor Channel Assignment** screen.
   - Enter **procr** in the **Local Node** field on the **IP Services** screen.

7. For adjunct connectivity to an ESS server or an LSP, use the **Survivable Processor - Processor Channels screen** to:
   - Use the same processor channels information as the main server by entering **i(nherit)** in the **Enable** field.
   - Use different translations than that of the main server by entering **o(verwrite)** in the **Enable** field. After entering **o(verwrite)** you can enter information specific to the ESS server or the LSP in the remaining fields.
   - Disable the processor channel on the ESS server or the LSP by entering **n(o)** in the **Enable** field.
8. Execute a **`save translations all`**, **`save translations ess`**, or **`save translations lsp`** command to send (file sync) the translations from the main server to the ESS or LSP server.

Defining network port usage

The main server(s), LSPs, and each ESS server, use specific TCP/UDP ports across a customer's network for registration and translation distribution. Use the information in Table 9 to determine which TCP/UDP ports must be open in your network for an LSP or an ESS server. You must check the firewalls on your network to open the required TCP/UDP ports.

Table 9: Network port usage

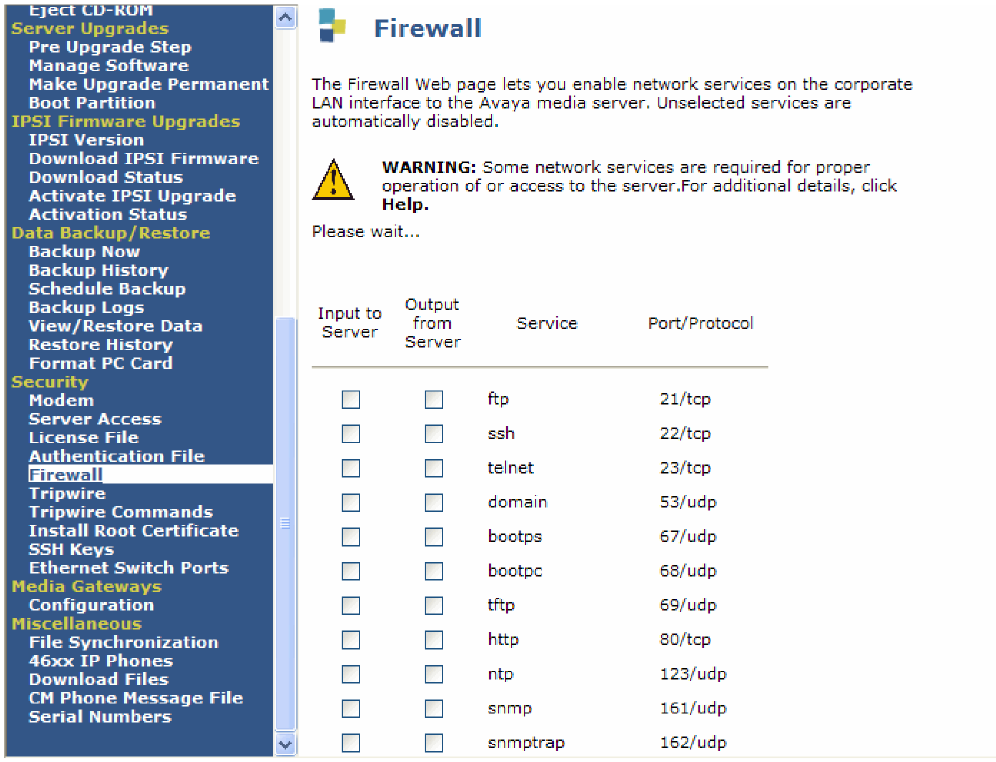
Port	Used by:	Description
20	ftp data	
21	ftp	
22	ssh/sftp	
23	telnet server	
68	DHCP	
514	This port is used in Communication Manager 1.3 to download translations.	
1719 (UDP port)	The survivable server(s) to register to the main server(s).	UDP outgoing and incoming
1024 and above	Processor Ethernet	TCP outgoing
1039	Encrypted H.248	TCP incoming
		1 of 2

Table 9: Network port usage (continued)

Port	Used by:	Description
1720	H.323 host cell	TCP incoming and outgoing
1956	Command server - IPSI	
2312	Telnet firmware monitor	
2945	H.248 message	TCP incoming and outgoing
5000 to 9999	Processor Ethernet	TCP incoming
5010	IPSI/Server control channel	
5011	IPSI/Server IPSI version channel	
5012	IPSI/Server serial number channel	
21873 (TCP port)	The main server(s) running Communication Manager 2.0 to download translations to the LSP(s).	Prior to an upgrade to Communication Manager 3.0 or later, servers running Communication Manager 2.x used port 21873 to download translations to the LSP(s). Once the upgrade to 3.0 is complete and all servers are running versions of Communication Manager 3.0 or later, the main server(s) uses port 21874 to download translations and port 21873 will no longer be needed.
21874 (TCP port)	The main server(s) to download translations to the survivable servers.	A main server(s) uses port 21874 to download translations to the ESS server(s) and the LSP(s) on Communication Manager 3.0 and later loads.
		2 of 2

To configure the ports on your server, click **Firewall** under the **Security** heading in the maintenance Web interface (Figure 202).

Figure 202: Firewall screen in the maintenance Web interface



Configuring the PE Interface

Use the information in this section to configure the PE interface on the server. This section does not contain complete information on how to configure the Communication Manager server. For information on how to configure the Communication Manager server, see the installation documentation for your server type. The documentation can be found at http://support.avaya.com.

Setting identities

Using the Web interface, configure Processor Ethernet on the server as follows:

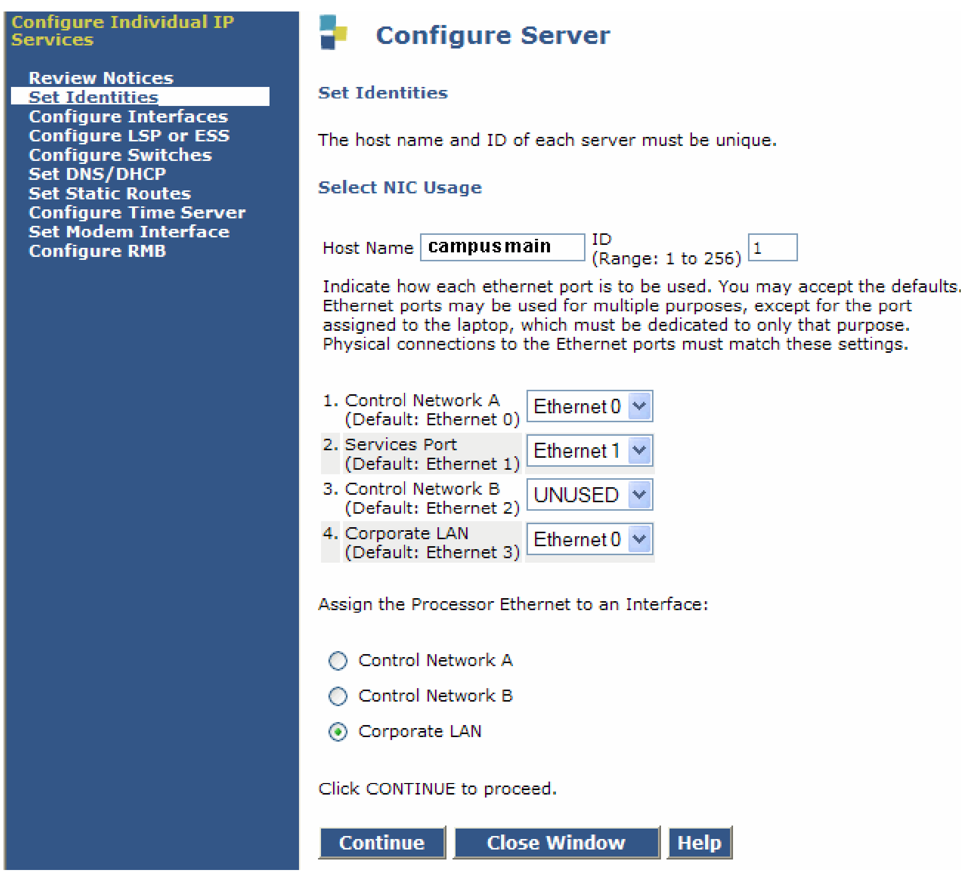
1. On the **Configure server - Set Identities page**, select the interface that will be used for Processor Ethernet.

   The interface you select is based on what the PE interface is used for and the topology of your network. For example, if the PE interface is used for CMS and CDR connectivity, then the PE interface must connect to the LAN that the CMS and CDR are connected to.

   To register with the main server, an ESS server uses its PE interface to connect to a CLAN in a port network. The PE interface on the ESS server must therefore have connectivity to the CLAN that has connectivity to the main server.

   In some configurations, the CLAN is connected to the enterprise LAN to allow telephony users access to the CLAN. In this case, the PE interface must also be assigned to a NIC that can connect to the same enterprise LAN. However, CLANs can be connected to different isolated LANs. In order to select the correct interface for the PE interface to NIC assignment, you must know and understand your network topology.

Figure 203: Configure server - set identities page for an S8500 Server

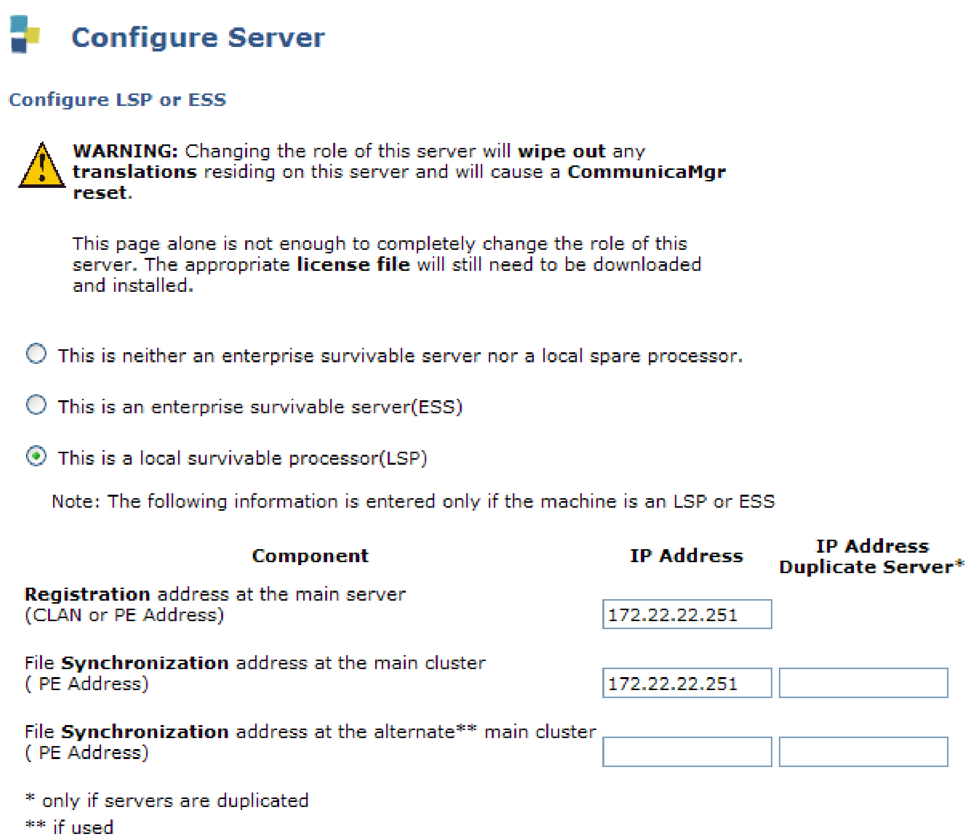


2. If this is a main server, this is the only configuration screen that you need to configure for the PE interface.
3. When configuring an ESS or an LSP server, you must complete the **Configure Server - Configure LSP or ESS** screen in addition to the **Set Identities** screen.

Configuring an LSP or an ESS server

When configuring an ESS server or an LSP you must complete the **Configure Server - Configure LSP or ESS** screen in addition to the **Set Identities** screen. Figure 204 shows an example of the **Configure LSP or ESS** screen.

Figure 204: Configure server - configure an LSP or an ESS server



Complete the following fields in the **Configure LSP or ESS** screen:

1. Select the radio button next to the correct entry to indicate if this is an ESS server, an LSP, or neither an ESS server or an LSP.
2. In the **Registration address at the main server** field, enter the IP address of the CLAN or PE interface of the main server that is connected to a LAN to which the LSP or ESS server is also connected. The IP address is used by the LSP or ESS server to register with the main server. In a new installation, where the LSP or the ESS server has not received the initial translation download from the main server, this address will be the only address that the LSP or the ESS server can use to register with the main server.
3. **File synchronization address of the main cluster**: Enter the IP address of a server's NIC connected to a LAN to which the LSP or the ESS server is also connected. The ESS server or the LSP must be able to ping to the address. Consideration should be given to which interface you want the file sync to use. Avaya recommends the use of the customer LAN for file sync.

Adding the PE as a controller for the H.248 gateways

Use the command **`set mgc list`** on an H.248 gateway when adding a PE-enabled S8500 or S8300 Server as the primary controller, or as an alternate controller for the gateway. The first media gateway controller on the list is the primary controller (gatekeeper).

For example, if during configuration a NIC card with IP address 132.222.81.1 is chosen for the PE interface, the **`set mgc list`** command would be:

`set mgc list 132.222.81.1`, ***`<alt_ip-address_1>`***, ***`<alt ip-address 2>`***

Administering PE in Communication Manager

Processor Ethernet administration is always performed on the main server. The LSP or the ESS server receives the translations from the main server during registration or when you perform a **`save translations lsp`**, **`save translations ess`**, or **`save translations all`** command on the SAT of the main server.

When communication with the main server is lost, you can perform administration on an active LSP or an active ESS server. In this case, the administration is temporary until the communication to the main server is restored. At that time, the LSP or the ESS server registers with the main server and receives the file sync. The file sync will overwrite any existing translations.

This section outlines the screens used in the administration of Processor Ethernet. For more information on these screens, see Chapter 19: Screen Reference.

- **IP Node Names** screen

  If the PE interface is enabled in the license file, the PE interface (**procr**) automatically appears on the **IP Node Names** screen. You cannot add the PE interface to the **IP Node Names** screen.

- **IP Interfaces** screen

  Administer the PE interface and the CLAN interface on the **IP Interfaces** screen. It is possible to have both the PE interface and one or more CLAN boards administered on the same system. On some server types the PE interface is automatically added. To see if the PE interface is already added to your system, use the command **`display ip-interface procr`**. To add the PE interface, use the command **`add ip-interface procr`**.

  Administer the PE interface on the main server if the main is an S8300, S8400, or an S8500 and one or more of the following entities use the main server's PE interface to register with the main server:

  - AE Services, CMS, CDR adjuncts
  - H.248 gateways
  - H.323 gateways or endpoints.

  For configurations that do not use the PE interface on the main server, you do not need to administer the **IP Interfaces** screen. This is true even if the ESS server or the LSP is using the PE interface. The **IP Interfaces** screen is automatically populated for an ESS or LSP.

- **Survivable Processor** screen

  The **Survivable Processor** screen is used to add a new LSP and also provides a means to connect one of the three supported adjuncts (CMS, CDR, AESVCS) to an LSP or an ESS server. The **Survivable Processor** screen is administered on the main server. The translations are sent to the ESS server or LSP during a file sync. After the file sync, the information on the **Survivable Processor** screen is used by the LSP or the ESS server to connect to a CMS, an AESVCS, or a CDR.

Administering ESS servers for PE

If there is an ESS server in the configuration, you must add the ESS server using the **System Parameters - ESS** screen. Information from the **System Parameters ESS** screen is automatically copied to the **Survivable Processor** screen. You do not add an ESS server using the **Survivable Processor** screen. The **Survivable Processor** screen is used only when you want to use the PE interface of the ESS server to connect to one of the three supported adjuncts (CMS, CDR, AESVCS). For more information on administering the ESS server on the **Survivable Processor** screen, see Survivable Processor on page 1558.

Starting with Communication Manager 3.1, node names are used in place of IP addresses for ESS servers on pages one through five of the **System Parameters ESS** screen. When upgrading an ESS configuration to Communication 3.1or later, the following events occur:

- When you upgrade an ESS configuration from a release prior to Communication Manager 3.1, the system automatically creates a node name for each administered ESS server. The node name is combination of:
  - The string **ESSCid** followed by the ESS server's cluster ID
  - The string **Sid** followed by the ID of the ESS server

    For example, if the ESS server had an IP address of 111.222.333.123, a cluster ID of 4, and a server ID of 2, the node name automatically assigned to the server would be, ESSCid004Sid002. This node name would have the IP address of 111.222.333.123.
- A survivable processor entry is created for each ESS server.
  - The **Enable** field in the **Survivable Processor** screen defaults to **i(nherit)** for all administered adjunct links.

Administering LSP servers for PE

Local Survivable Processors (LSPs) are administered using the **Survivable Processor** screen. For more information on administering an LSP, see Survivable Processor on page 1558.

Working with adjuncts

For the simplex main server, adjuncts that use the CLAN can use the PE interface of the main server for connectivity to the main server. For the LSP and the ESS server, there are three adjuncts, the CMS, AESVCS, and the CDR, that are supported using the LSP or the ESS server's PE interface. This section provides a high-level overview of the adjuncts supported by the ESS server and the LSP and how they are administered to use the PE interface.

- **Survivable CMS**

  Starting with CMS Release 13.1, you can use a Survivable CMS co-located at the site of the ESS server or LSP. A Survivable CMS is a standby CMS that collects data from an LSP or an ESS server when the main server is not operational or when the customer is experiencing a network disruption. A Survivable CMS should not be located at the same location as the main server.

  During normal operations, the Survivable CMS has a connection to the ESS server or the LSP, but does not collect data or support report users. Only the main CMS server collects data. When an ESS server assumes control of one or more port networks, or an LSP is active, the ESS server and/or the LSP sends data to the Survivable CMS.

- **CDR**

  The server initiates the connection to the CDR unit and sends call detail information over the configured link. The link remains active at all times while the CDR unit waits for data to be sent by a connected server. In the case of an ESS server or LSP, data will not be sent until the survivable server becomes active. Some CDR units can collect data from multiple servers in a configuration, separately or all at once. For information on the capability of your CDR unit, check with your CDR vendor.

  The CDR unit is administered on the **IP Services** screen. To use the PE interface, **procr** must be entered in the **Local Node** field.

- **AESVCS**

  AESVCS (Application Enablement Services) supports connectivity to a maximum of 16 servers. Since AESVCS cannot tell which server is active in a configuration, it must maintain a constant connection to any server from which it might receive data. An Avaya S8XXX Server "listens" for AESVCS after it boots up. The AESVCS application establishes the connection to the server.

  If the adjunct terminates solely on the main server's PE interface, you do not have to administer the **Survivable Processor** screen. If AESVCS connects to an LSP or an ESS server, you must administer the **Survivable Processor** screen in addition to the **IP Services** screen.

Using load balancing

You can load balance the H.323 endpoint traffic across multiple IP interfaces. The **IP Interfaces** screen contains the fields needed to load balance the IP interface.

Note:
The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Use the following guidelines to load balance the H.323 endpoints:

1. Load balancing starts with placing the CLANs and the PE interface into a network region using the **Network Region** field.
2. Within the network region, further load balancing is done by entering a priority in the **Gatekeeper Priority** field. This field appears only if the **Allow H.323 Endpoint** field is set to **y**. You can have more than one IP interface administered at the same value in the **Gatekeeper Priority** field within a region. For example, you could have two CLANs administered as a **1** in the **Gatekeeper Priority** field.

   Valid values for the **Gatekeeper Priority** field range from **1** to **9**, with **1** being the highest. Within a network region, the system uses the highest Gatekeeper Priority IP interface first.
3. The number that is entered in the **Target socket load** or the **Target socket load and Warning level** field is the maximum number of connections you want on the interface. A socket represents a connection of an endpoint to the server. As endpoints connect, the load balancing algorithms direct new registrations to interfaces that are less loaded. The current

load is unique to each interface and is the ratio of currently used sockets to the number administered in this field. Communication Manager tries to keep the ratio used by each interface the same. Note that this is a "target" level, and that Communication Manager might use more sockets than specified in the field.

If there is only one ip-interface within a priority, the **Target socket load** or the **Target socket load and Warning level** field is no longer used for load balancing. A number can be entered in this field to receive an error or a warning alarm if the targeted value is exceeded.

Setting Alternate Gatekeeper List (AGL) priorities

The alternate gatekeeper list is used for H.323 endpoints when they cannot reach their primary gatekeeper. The **Gatekeeper Priority** field and the **Network Region** field on the **IP Interfaces** screen determines the priority of the PE interface or the CLAN on the alternate gatekeeper list. For more information about the **Gatekeeper Priority** field, see Using load balancing on page 603.

Administering Call-processing

The telephony features of the S8300 Server are administered using the same commands and procedures as an S87XX Server or a legacy DEFINITY Enterprise Communications System.

Accessing Avaya Communication Manager

Avaya Communication Manager resides on the Avaya S8XXX Server. It can be accessed through either Avaya Site Administration (ASA) or the System Access Terminal (SAT) program.

Avaya Site Administration

Avaya Site Administration features a graphical user interface (GUI) that provides access to SAT commands as well as wizard-like screens that provide simplified administration for frequently used features. You can perform most of your day-to-day administration tasks from this interface such as adding or removing users and telephony devices. You can also schedule tasks to run at a non-peak usage time. ASA is available in several languages.

Note:

In order for ASA to work properly with the ASG Guard II, the **Write (ms)** field on the **Advanced** tab of the **Connection Properties** screen must be set to a value of **5** (i.e., delay of 5 ms). ASG Guard II is an outboard appliance providing access security for Avaya products that do not have Access Security Gateway (ASG) software as a native application. For more information on ASG Guard II, contact your Avaya technical support representative.

For more information, see Using Avaya Site Administration in Chapter 1: System Basics.

System Access Terminal

The System Access Terminal (SAT) program uses a Command Line Interface (CLI) interface for telephony administration. SAT is available through the Avaya Site Administration package.

Security Considerations

Levels of security for administration of the G700 Media Gateway are the same as traditionally for Avaya Communication Manager. This means that administration login passwords are passed in plain text with no encryption. Exceptions to this no-encryption policy include:

- The ASG program that is installed on all Avaya S8XXX Servers.
- An encrypted Web interface to the Avaya S8XXX Server (see the security certificate information in the server online help)
- Optional encryption for data backups (see Backing up and restoring data on page 591).
- Support for RADIUS authentication for media gateways and P330 stack elements using the P330 Device Manager. See Using Device Manager to administer G700 Media Gateway components on page 585.

Command syntax changes for media modules

The syntax for using the SAT commands for a G700 Media Gateway or Avaya S8XXX Server has changed. In a traditional DEFINITY system, ports are identified by the cabinet number, carrier, slot, and port. For example: 02A0704

Because this numbering convention does not make sense for media modules, a new convention was developed. The numbering convention for the media modules uses the same seven-character field as does a traditional system, but the fields represent the media gateway number, media module slot (V1 to V9), and port number (00 to 99 are supported; the actual number of ports that can be specified depends on the type of media module).

Example: 001V205

In this example, the 001 represents the media gateway number, the V2 represents the slot number (possibly V1 through V9), and 05 is the port number.

Accessing the Communication Manager SAT CLI

You can access the command line interface (CLI) of the Avaya Communication Manager SAT using any of the following methods:

- Using Secure Shell for remote login on page 606

- Using Telnet over the Customer LAN on page 607
- Using Avaya Terminal Emulator for LAN connection to Communication Manager on page 613
- Using Windows for PPP modem connection (Windows 2000 or XP) on page 610

  This connection requires you to have a modem on your PC. It also requires you to do the following first:

  - Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP) on page 608
  - Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 609
- Using Avaya Terminal Emulator for Modem Connection to Communication Manager on page 614

  This connection requires you to have a modem on your PC. It also requires you to do the following first:

  - Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP) on page 608
  - Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 609
- Configuring Avaya Site Administration on page 27

Using Secure Shell for remote login

You can log in remotely to the following platforms using Secure Shell (SSH) as a secure protocol:

- G350 Media Gateway
- S8300, S8400, S8500, or S87XX Server Linux command line
- Communication Manager System Administration Terminal (SAT) interface on an Avaya S8XXX Server using port 5022.

The SSH capability provides a highly secure method for remote access. The capability also allows a system administrator to disable Telnet when it is not needed, making for a more secure system. For details on disabling Telnet, see Turning off Telnet for increased security on page 608.

Note:

The client device for remote login must also be enabled and configured for SSH. Refer to your client P.C. documentation for instructions on the proper commands for SSH.

Enabling and disabling SSH or SFTP sessions on the CLAN or VAL circuit packs

Prerequisites:

- TN799BP (CLAN) with Release 3.0 firmware
- VAL with Release 3.0 firmware
- Communication Manager, Release 3.0 or later

To enable a secure FTP (SFTP) session on a CLAN or VAL circuit pack:

1. Type **`enable filexfr [board location]`**. Press **Enter**. The Enable File Transfer screen displays.

Figure 205: Enable File Transfer screen

```
enable filexfer                                                        Page 1
                              ENABLE FILE TRANSFER
             Login:______
          Password:___________
          Password:___________
            Secure?
```

2. Type a 3-6 alphabetic character login in the **Login** field.
3. Type a 7-11 character password (one character must be a number) in the first **Password** field.
4. Retype the same password in the second **Password** field.
5. Type **y** in the **Secure?** field. Press **Enter**.

SFTP is enabled on the circuit pack, and the login/password are valid for 5 minutes.

To disable a secure FTP (SFTP) session on a CLAN or VAL circuit pack:

1. Type **`disable filexfr [board location]`**. Press **Enter**.

SFTP is disabled on the circuit pack.

Using Telnet over the Customer LAN

Note:

For ease of administration, it is recommended that, whenever possible, you use the Avaya Terminal Emulator, or access the server's command line interface using an SSH client, like PuTTY, and an IP address of 192.11.13.6., instead of Telnet.

To use Telnet over the customer LAN:

1. Make sure you have an active Ethernet (LAN) connection from your computer to the Avaya S8XXX Server.
2. Access the telnet program; for example:
   - On a Windows system, go to the **Start** menu and select **Run**.
   - Type `telnet <server_IP_address> 5023`. You might also type the server name if your company's DNS server has been administered with the Avaya S8XXX Server name.
3. When the **login** prompt appears, type the appropriate user name (such as *cust* or *craft*).
4. When prompted, enter the appropriate password or ASG challenge.
5. If you log in as **craft**, you are prompted to suppress alarm origination. Generally you should accept the default value (yes).
6. Enter your preferred terminal type.

   The system displays the SAT command line.

Turning off Telnet for increased security

If you are using SSH to log in to the server or gateway, you might wish to turn off Telnet for an extra level of security. To turn off Telnet:

1. On the Maintenance Web page main menu, under **Security**, click on **Firewall**.

   The system displays the **Firewall** page.
2. Click the **Telnet Input to Server** boxes so that the check mark disappears.
3. Click the **Telnet Output from Server** box so that the check mark disappears.
4. Click the **Submit** button.

Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP)

Note:

The remote dial-up PC must be configured for PPP access. Also, Avaya Terminal Emulator does *not* support Windows XP.

To set up Windows for Modem Connection:

1. Right-click **My Network Places** and click **Properties**.
2. Click **Make New Connection** and follow the Network Connection Wizard:
3. Select **Dial-up to private network** on the **Network Connection Type** screen.
4. In the **Phone number** field, enter the appropriate telephone number inserting special digits such as 9 and 1 or *70, if necessary.

5. On the **Connection Availability** screen, click **For all users** or **Only for myself**, as appropriate.
6. On the **Completing the Network Connection Wizard** screen, type the name you want to use for this connection. This name will appear in the **Network and Dial-up Connections** list.
7. Check **Add a shortcut to my desktop**, if desired, and click **Finish**.
8. If a **Connect** screen appears, click **Cancel**.

Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA)

To configure the remote PC for PPP modem connection:

1. On your PC's desktop, right-click **My Network Places** and click **Properties**.

   The system displays the **Network and Dial-up Connections** screen.
2. Double-click the connection name you made in the previous task, Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP) on page 608.

   Note:

   Depending on your system, the **Connect** screen might appear, from which you must select **Properties**.
3. Click the **Security** tab.
4. Select the **Advanced (custom settings)** radio button.
5. Check the **Show terminal window** checkbox.
6. Click the **Networking** tab.
7. In the **Components** box, verify that **Internet Protocol (TCP/IP)** and **Client for Microsoft Networks** are both checked.
8. Select **Internet Protocol (TCP/IP)** and click **Properties**.
9. Click the **Advanced** button.
10. Uncheck (clear) the **Use default gateway on remote network** box.
11. Click **OK** three times to exit and save the changes.

Using Windows for PPP modem connection (Windows 2000 or XP)

This connection requires you to have a modem on your PC. It also requires you to do the following first:

- Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP) on page 608
- Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 609.

To use Windows for PPP modem connection:

1. Return to the **Network and Dial-up Connections** screen and right-click the connection you just created.
2. Select **Connect**.
3. Leave the **User Name**, **Password**, and **Domain** fields blank. If the **Dial** field is blank, enter the appropriate telephone number.
4. Click the **Dial** button. When the Avaya S8XXX Server's modem answers, the system displays the **After Dial Terminal** screen.
5. Log on to the LAN.
   a. Enter your remote access login name and password.
   b. When the **Start PPP Now!** message appears, click **Done**.
   c. The system displays a small double-computer icon in the lower right portion of your screen.
6. Double-click the double-computer icon.
7. The system displays the connection's **Dialup Status** box.
8. Click on the **Details** tab.
9. Note the **Server** IP address.
10. Open a telnet session to the Avaya S8XXX Server:

    Type **`telnet <ip-address>`**, where ***`<ip-address>`*** is the Server IP address as noted in the **Dialup Status** box from Step 9.
11. Access SAT or use the CLI commands as needed.

International modem settings

You can change modem settings for a specific country using the Linux command line. To enter AT commands from a Linux command prompt:

1. On the command line, type (without quotes): **cu -l /dev/ttys1**, where -l is a lowercase L. Replace "ttyS1" with the tty port for the USB (e.g., /dev/usb/ttyACM0 for USB port 1). Press **Enter**.

   The system displays a "Connected" message.

2. Type `ATI`. Press **Enter**.

   The system displays an **OK** message.

3. Type the appropriate AT command to change the modem setting for the desired country. For example, for Japan, type **AT%T19,0,10&W0**. Press **Enter**.

   The system displays an **OK** message.

4. To return to the command prompt, type "**~**" (without quotes).

The following table shows AT commands and result codes for country-specific modem configuration

:

Country	AT Command (Hexadecimal)	Result Code (Decimal)
Argentina	AT%T19,0,34	52
Australia	AT%T19,0,01	1
Austria	AT%T19,0,34	52
Belgium	AT%T19,0,34	52
Brazil	AT%T19,0,34	52
Canada	AT%T19,0,34	52
China	AT%T19,0,34	52
Cyprus	AT%T19,0,34	52
Czech Republic	AT%T19,0,25	37
Denmark	AT%T19,0,34	52
Finland	AT%T19,0,34	52
France	AT%T19,0,34	52
Germany	AT%T19,0,34	52
		1 of 3

Country	AT Command (Hexadecimal)	Result Code (Decimal)
Greece	AT%T19,0,34	52
Hong Kong	AT%T19,0,30	48
Hungary	AT%T19,0,30	48
Iceland	AT%T19,0,34	52
India	AT%T19,0,30	48
Indonesia	AT%T19,0,30	48
Ireland	AT%T19,0,34	52
Italy	AT%T19,0,34	52
Japan	AT%T19,0,10	16
Korea	AT%T19,0,30	48
Liechtenstein	AT%T19,0,34	52
Luxembourg	AT%T19,0,34	52
Mexico	AT%T19,0,34	52
Netherlands	AT%T19,0,34	52
New Zealand	AT%T19,0,09	9
Norway	AT%T19,0,34	52
Philippines	AT%T19,0,30	48
Poland	AT%T19,0,30	48
Portugal	AT%T19,0,34	52
Russia	AT%T19,0,34	52
Singapore	AT%T19,0,30	48
South Africa	AT%T19,0,35	53
Slovak Republic	AT%T19,0,34	52
Slovenia	AT%T19,0,30	48
Spain	AT%T19,0,34	52
Sweden	AT%T19,0,34	52
		2 of 3

Country	AT Command (Hexadecimal)	Result Code (Decimal)
Switzerland	AT%T19,0,34	52
Turkey	AT%T19,0,34	52
United Kingdom	AT%T19,0,34	52
United States	AT%T19,0,34	52
		3 of 3

Enabling transmission over IP networks for modem, TTY, and fax calls

Prerequisites

The ability to transmit fax, modem, and TTY calls over IP trunks or LANs and WANs assumes that the endpoints sending and receiving the calls are connected to a private network that uses H.323 trunking or LAN connections between gateways and/or port networks. This type of transmission also assumes that calls can either be passed over the public network using ISDN-PRI trunks or passed over an H.323 private network to Communication Manager switches that are similarly enabled.

As a result, it is assumed that you have assigned, or will assign, to the network gateways the IP codec you define in this procedure. For our example, the network region 1 will be assigned codec set 1, which you are enabling to handle fax, modem, and TTY calls.

To enable transmission over IP networks for modem, TTY, and fax calls:

1. Type **`change ip-codec-set 1`**. Press **Enter**.

   The **IP Codec Set** screen appears.

Complete the fields as required for each media type you want to enable. Press **Enter**.

For more information on modem/fax/TTY over IP, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Using Avaya Terminal Emulator for LAN connection to Communication Manager

You can download the Avaya Terminal Emulator from the main menu for the VisAbility Management Suite. Simply click **Download** next to the Administration menu item and follow the instructions.

Once the Terminal Emulator is installed on your PC, use the following steps to establish a LAN connection to your Avaya S8XXX Server.

1. Double-click the Terminal Emulator icon on your desktop. Alternatively, go to the Start menu, select Programs, then Avaya, then Terminal Emulator.

   The system displays the Terminal Emulator.

2. From the menu bar across the top of the screen, select **Phones**, then select **Connection List**.

   The system displays the **Connections** screen.

3. From the menu bar across the top, select **Connection**, then select **New Connection**.

   The system displays the **Connection Settings** screen.

4. Put in a name for the connection. Usually, this will be the name of your Avaya S8XXX Server.

5. On the **Host** screen, click **Telnet**.

6. Click the **Emulation** tab at the top.

   The system displays the **Emulation** tab.

7. From the Emulator dropdown box, select the emulator you desire, usually **513BCT** (default), **AT&T 4410, AT&T**, or **DECVT100**.

8. In the **Keyboard** window, select **pbx**.

9. Click the **Network** tab.

   The system displays the **Network** tab.

10. In the **IP address** field, type the IP address of the Avaya S8XXX Server.

11. In the **TCP/IP port number** field, type **5023** to log in directly to the Communication Manager SAT command line.

12. Click **OK**.

    The **Connection Settings** screen disappears.

13. On the **Connections** screen, double-click the name of the connection you just set up.

    The Login prompt for the Communication Manager software appears.

14. Log in to Communication Manager to access the SAT command screen.

Using Avaya Terminal Emulator for Modem Connection to Communication Manager

This connection requires you to have a modem on your PC. It also requires you to do the following first:

- Setting up Windows for Modem Connection to the Avaya S8XXX Server (Windows 2000 or XP) on page 608

- Configuring the remote PC for PPP modem connection (Windows 2000 or XP, Terminal Emulator, or ASA) on page 609

Once the Terminal Emulator is installed on your PC and you have a modem attached and configured to both your PC and the Avaya S8XXX Server, use the following steps to establish a modem connection to your Avaya S8XXX Server.

1. Double-click the Terminal Emulator icon off of your desktop. Alternatively, go to the Start menu, select **Programs**, then select **Avaya**, and finally select **Terminal Emulator**.

   The system displays the Terminal Emulator.

2. From the menu bar across the top of the screen, select **Phones**, then select **Connection List**.

   The system displays the **Connections** screen.

3. From the menu bar across the top, select **Connection**, then select **New Connection**.

   The system displays the **Connection Settings** screen.

4. Put in a name for the connection. Usually, this will be the name of your Avaya S8XXX Server.

5. In the Host window, click **Telnet**.

6. Click the **Emulation** tab at the top.

   The system displays the **Emulation** tab.

7. From the Emulator dropdown box, select the emulator you desire, usually 513BCT (default), AT&T 4410, AT&T or DECVT100.

8. In the Keyboard window, select **pbx**.

9. Click the **Modem** tab.

   The system displays the **Modem** tab.

10. In the **IP address** field, type the IP address of the connection **Dialup Status** box as noted in Step 9.

11. In the **TCP/IP port number** field, type **5023** to log in directly to the Communication Manager SAT command line.

12. In the **Modem** field, use the dropdown box to select the type of modem that your PC uses.

13. In the **Serial port** field, select the COM port you are using for your modem connection.

14. In the **Baud rate** field, select 9600 from the dropdown box.

15. Click the **Dial Numbers** tab.

    The system displays the **Display Numbers** tab.

16. Type the telephone number of the Avaya S8XXX Server as appropriate. Enter **1** in the **Country Code** field for long-distance.

17. Click **OK**.

18. On the **Connections** screen, double-click. the name of the connection you just set up.

19. The PC dials up the Avaya S8XXX Server, and when connected, the login prompt for the Communication Manager software appears.
20. Log in to Communication Manager to access the SAT command prompt screen.

Logging in to the Avaya S8XXX Server with ASA

To start Avaya Site Administration, click **Start > Programs > Avaya > Site Administration**. Avaya Site Administration supports a terminal emulation mode, which is directly equivalent to SAT command interface. Avaya Site Administration also supports a whole range of other features, including the GEDI and Data Import. For more information refer to the Online Help, Guided Tour, and Show Me accessed from the Avaya Site Administration Help menu.

To use Avaya Site Administration, open the application and select the Avaya S8XXX Server you want to access. When prompted, log in.

When you are logged in, click **Start GEDI**.

Screen and command summary

The following screens are used to administer G700 Media Gateways, Avaya S8XXX Servers, and other media modules.

Media Gateways

The commands and screens for media gateways include:

The **Media-Gateway** administration screen is used to administer G700 Media Gateways and their media modules. Information is similar to the **list media-gateway** screen (next item), but also includes MAC address, network region, location and site data.

Note:

For more information about the **Media-Gateway** screen, and a description of commands, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

- The **`list media-gateway ['print' or 'schedule']`** command shows the list of currently administered gateways. Information includes the media gateway number, name, serial number, IP address, and whether or not this media gateway is currently registered with the call controller. The IP address field is blank until the media gateway registers once, then remains populated.
- The **`list configuration media-gateway`** ***`x`*** command allows you to list all the assigned ports on the media modules for the G700 Media Gateway specified by its number (***x***).

System-Parameters Customer-Options (Optional Features) screen

See **System Parameters Customer-Options (Optional Features)** for a complete description of this screen.

- The OPTIONAL FEATURES section contains a **Local Survivable Processor (LSP)** field. If it displays a **y** (yes), this Avaya S8XXX Server is configured to provide standby call processing in case the primary server is unavailable. See Configuring the Local Survivable Processor on page 584 for details. This display-only field can be set only by the license file.
- Two additional fields in this section indicate if the primary call-processing controller is an S8300 Server. If traditional port networking is disabled and Processor Ethernet is enabled, an S8300 Server is controlling telecommunications.
  - **Port Network Support**: set to **n** indicates that traditional port networking is disabled. An S8300 Server is the primary call controller.
  - **Processor Ethernet**: set to **y** indicates the presence of an S8300 Server.

Quality of Service Monitoring

Several screen changes allow you to monitor Quality of Service (QoS) on an Avaya S8XXX Server with a G700 configuration. The media gateway can send data to a real-time control protocol (RTCP) server, which in turn monitors the network region's performance. Screens include:

- An RTCP MONITOR SERVER section on the **IP-Options System Parameters** screen allows you to enter a single default IP address, server port, and RTCP report period that can be utilized by all administered network regions. This means you do not have to re-enter the IP address each time you access the **IP Network Region** screen.
- The **IP Network Region** screen also must be administered for QoS monitoring (for details, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504). If the **RTCP Enabled** field is left at default (**y**), then be sure to set a valid IP address in the **IP-Options System Parameters** screen. For situations that require customization, this screen is administered on a per IP network regional basis. Items to customize include:
  - Enabling or disabling of RTCP monitoring
  - Modifications to the report flow rate
  - Changes to the server IP address and server port
- The `list ip-network-region qos` and `list ip-network-region monitor` commands list quality of service and monitor server parameters from the **IP Network Region** screen as follows:
  - **qos** displays VoIP media and call control (and their 802.1p priority values), BBE DiffServ PHB values, RSVP profile and refresh rate.

- **monitor** displays RTCP monitor server IP address, port number, report flowrate, codec set, and UDP port range parameters.

Media Gateway serviceability commands

Additional commands related to media gateways appear in *Maintenance for the Avaya G700 Media Gateway controlled by an Avaya S8300 Media Server or an Avaya S8700 Media Server*. These include:

- The **`status media-gateways`** command provides an alarm summary, busyout summary, and link summary of all configured media gateways.
- Several commands have been modified to support the media gateway port identification format described in Command syntax changes for media modules on page 605. These include:
  - Message Sequence Trace (mst)
  - display errors
  - display alarms

Administering SNMP

The SNMP protocol provides a simple set of operations that allow devices in a network to be managed remotely. Communication Manager 4.0 and later releases supports the following versions of SNMP:

- SNMP Version 1 (SNMP v1) and SNMP Version 2c (SNMP v2c): SNMP v1 was the initial version of SNMP. Security in SNMP v1 and SNMP v2c is based on plan-text strings known as communities. Communities are passwords that allow any SNMP-based application to gain access to a device's management information.
- SNMP Version 3 (SNMP v3): SNMP v3 provides additional security with authentication and private communication between managed entities.

The server's maintenance web interface is used to perform the following functions for SNMP:

- Administer an SNMP trap: For more information, see Administering traps on page 619.
- Administer an SNMP agent: For more information, see Administering SNMP agents on page 624.
- Administer a filter: For more information, see Administering filters on page 628.
- View the G3-Avaya-MIB: For more information, see Administering SNMP agents on page 624.

- Enable the network ports needed for SNMP: For more information on the ports that need to be enabled for SNMP, see **Turning on access for SNMP ports at the network level on page 619**.

Turning on access for SNMP ports at the network level

CAUTION:

For SNMP to work, the Master Agent must be in an "Up" state and the SNMP ports must be enabled through the firewall. Use the information in this section to enable the ports needed for SNMP. To check the status of the Master Agent, select **Agent Status** on the server's web interface. To start the Master Agent, click **Start Agent.**

You must turn on network access for SNMP ports to allow SNMP access to Communication Manager. Use the following steps to turn on the network ports:

1. On the server's maintenance web page, click **Firewall** under the **Security** heading.

   The system displays the **Firewall** screen.
2. On the bottom of the **Firewall** screen, click **Advanced Setting**.
3. Scroll down and find the following three ports used by SNMP:
   - snmp 161/tcp
   - snmp 161/udp
   - snmptrap 1
4. On all three ports listed above, click the check boxes in both the **Input to Server** and **Output to Server** columns.
5. To save the changes, click **Submit**.

Administering traps

Use this section to administer the following actions for an SNMP trap destination:

- Add a trap destination on page 620
- Displaying an administered trap on page 623
- Changing an administered trap on page 623
- Deleting an administered trap on page 624

Add a trap destination

Use the following steps to add a trap destination for SNMP.

Instructions

To add a trap destination:

1. On the server's maintenance web page, click **SNMP Traps** under the **Alarms** heading.

   The **SNMP Trap** screen appears as shown in example Figure 206.

Figure 206: SNMP Traps screen

2. Check the status of the Master Agent.
   - If the status of the Master Agent is "Up": Select **Agent Status** from the navigation bar and click **Stop Agent**. Once the Master Agent is in a "Down" state, return to the **SNMP Trap** screen by clicking **SNMP Traps** on the navigation bar. If the status of the Master Agent is "Down," continue to step 3.
3. On the bottom of the screen, click **Add**.

   The **Add Trap Destination** screen appears as shown in Figure 207 and Figure 208.

Figure 207: Add Trap Destination screen

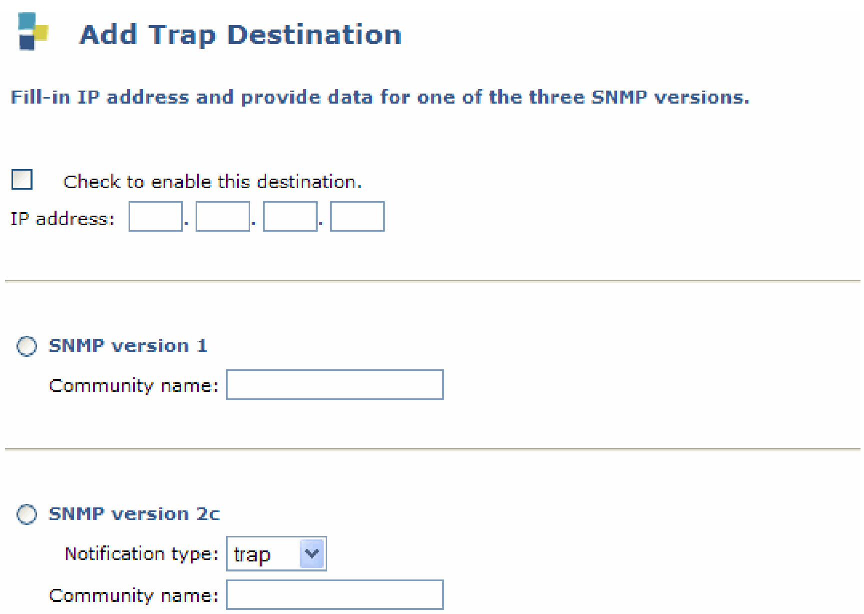
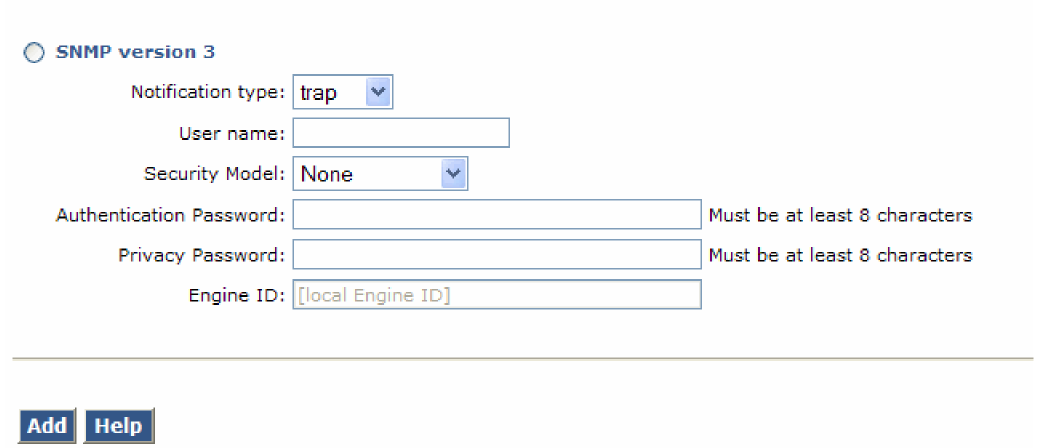


Figure 208: Add Trap Destination screen part 2



4. Click the **Check to enable this destination** box.

   Note:

   If you do not enable this destination, you can still enter the destination information and click **Add**. The system saves the data and displays the information with the status of disabled.

5. In the **IP address** field, enter the IP address for this destination.

6. Communication Manager supports SNMP v1, SNMP v2c, and SNMP v3. Select the SNMP version you are using.
7. Complete the fields associated with each version of SNMP that you select:
   - **SNMP version 1**: In the **Community name** field, enter the SNMP community name.
   - **SNMP version 2c**:
     a. In the **Notification type** field: Select between trap or inform. A trap is sent without notification of delivery. An inform is sent with a delivery notification to the sending server. If a delivery notification is not received, the inform is sent again.
     b. In the **User name** field: Enter the SNMP user name that the destination recognizes.
     c. In the **Security Model** field, select from one of the following options:
        - **none**: Traps are sent in plan text without a digital signature.
        - **authentication**: When authentication is selected, an authentication password must be given. SNMP v3 uses the authentication password to digitally "sign" v3 traps using MD5 protocol (associate them with the user).
        - **privacy**: When privacy is selected, both an authentication password and a privacy password is used to provide user-specific authentication and encryption. Traps are not only signed as described when using authentication, but also encrypted using Data Encryption Standard (DES) protocol.
     d. **Authentication Password** field: If you selected authentication as your security model, enter an authentication password. The password must be at least eight characters in length and can contain any characters except: ’\ & , ‘ ".
     e. **Privacy Password** field: If you selected privacy for your security model, first complete the **Authentication Password** field as described in the previous paragraph, then enter a password in the **Privacy Password** field. The password must be at least eight characters in length and can contain any characters except: ’\ & , ‘ ".
     f. **Engine ID** field: A unique engine ID is used for identification. Enter the engine ID of the designated remote server. An engine ID can be up to 24 characters in length consists of the following syntax:
        - IP address: The IP address of the device that contains the remote copy of SNMP.
        - Udp-port: (Optional) Specifies a User Datagram Protocol (UDP) port of the host to use.
        - udp-port-number: (Optional) The socket number on the remote device that contains the remote copy of SNMP. The default number is 161.
        - vrf: (Optional) Instance of a routing table.
        - vrf-name: (Optional) Name of the VPN routing/forwarding (VRF) table to use for storing data.
        - engineid-string: The name of a copy of SNMP.

8. Click **Add** to save the trap.

   The **SNMP Traps** screen re-appears displaying the trap.

9. To add another trap, follow steps 3 through 8.

10. If you are finished adding trap destinations, you must start the Master Agent. To start the Master Agent, select **Agent Status** from the navigation bar and click **Start Agent.**

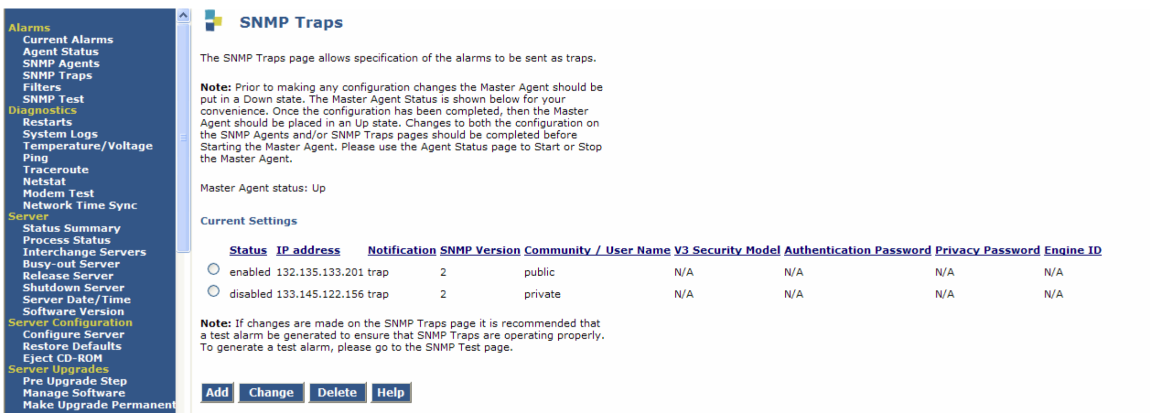
Displaying an administered trap

Use the following steps to display traps administered on the server:

1. On the server's maintenance interface, click **SNMP Traps**.

   The **SNMP Traps** screen displays as shown in example Figure 209.

Figure 209: SNMP Traps screen after an addition



The administered traps display under the **Current Settings** heading.

Changing an administered trap

Use the following steps to change parameters on an administered trap:

1. On the server's maintenance web interface, click **SNMP Traps**.

2. Check the status of the Master Agent. The Master Agent must be in a "Down" state before you make changes to the **SNMP Traps** screen.

   - If the status of the Master Agent is "Up": Select **Agent Status** from the navigation bar and click **Stop Agent**. Once the Master Agent is in a "Down" state, return to the **SNMP Traps** screen by clicking **SNMP Traps** on the navigation bar. If the status of the Master Agent is "Down," continue with 3.

3. Under the **Current Settings** heading on the **SNMP Traps** screen, click the radio button associated with the trap that you wish to change.

   The **Change Traps Destination** screen appears.

4. Make the changes to the trap destination and click **Change**.

   The **SNMP Traps** screen appears displaying the changes made to the selected trap.

5. If you are finished changing the trap destinations, you must start the Master Agent. To start the Master Agent, select **Agent Status** from the navigation bar and click **Start Agent.**

Deleting an administered trap

Use the following steps to delete an administered trap:

1. On the server's maintenance web interface, click **SNMP Traps**.
2. Check the status of the Master Agent. The Master Agent must be in a "Down" state before you make changes to the **SNMP Traps** screen.
   - If the status of the Master Agent is "Up": Select **Agent Status** from the navigation bar and click **Stop Agent**. Once the Master Agent is in a "Down" state, return to the **SNMP Traps** screen by clicking **SNMP Traps** on the navigation bar. If the status of the Master Agent is "Down," continue with 3.
3. Under the **Current Settings** heading on the **SNMP Traps** screen, click the radio button associated with the trap that you want to delete.

   The **Delete SNMP Trap Destination** screen appears.

4. Click **Delete.**

   The **SNMP Traps** screen appears displaying the updated trap destination list.

5. If you are finished deleting the trap destinations, you must start the Master Agent. To start the Master Agent, select **Agent Status** from the navigation bar and click **Start Agent.**

Administering SNMP agents

The **SNMP Agents** screen allows you to restrict SNMP services at the application level.

CAUTION:

The **Firewall page** - **Advanced Settings**, is used to inhibit the reception of SNMP messages at the network level. For SNMP to work, the Master Agent must be in an "Up" state and the SNMP ports must be enabled through the firewall. For more information on the **Firewall** page, see Turning on access for SNMP ports at the network level **on page 619**. For more information on how to check the status of the Master Agent, see step 2 under Instructions on page 625.

The **SNMP Agent** screen is divided into the following sections:

- A link to view the G3-Avaya-MIB: A management information base (MIB) contains definitions and information about the properties of managed sources and services that an SNMP agent(s) supports. The G3-Avaya-MIB is used for Communication Manager. The G3-Avaya-MIB contains:
  - Object identifiers (IDs) for every Avaya object
  - A list of MIB groups and traps along with their associated varbinds
  - Configuration, fault and performance data associated with the Communication Manager server

  To view the MIB, click **G3-Avaya-MIB**.

  The G3-Avaya-MIB appears on the screen.
- IP Addresses for SNMP Access: Use this section to restrict access by all IP addresses, allow access by all IP addresses, or list IP address from which SNMP is allowed.
- SNMP User/Communities: Use this section to enable and administer the version of SNMP that you are using. Communication Manager supports SNMP v1, SNMP v2c, and SNMP v3. Each SNMP version can be enabled and disabled independently of the other versions.

Instructions

CAUTION:

On the S87XX Servers (duplicated servers), you must administer an SNMP agent exactly the same on both servers.

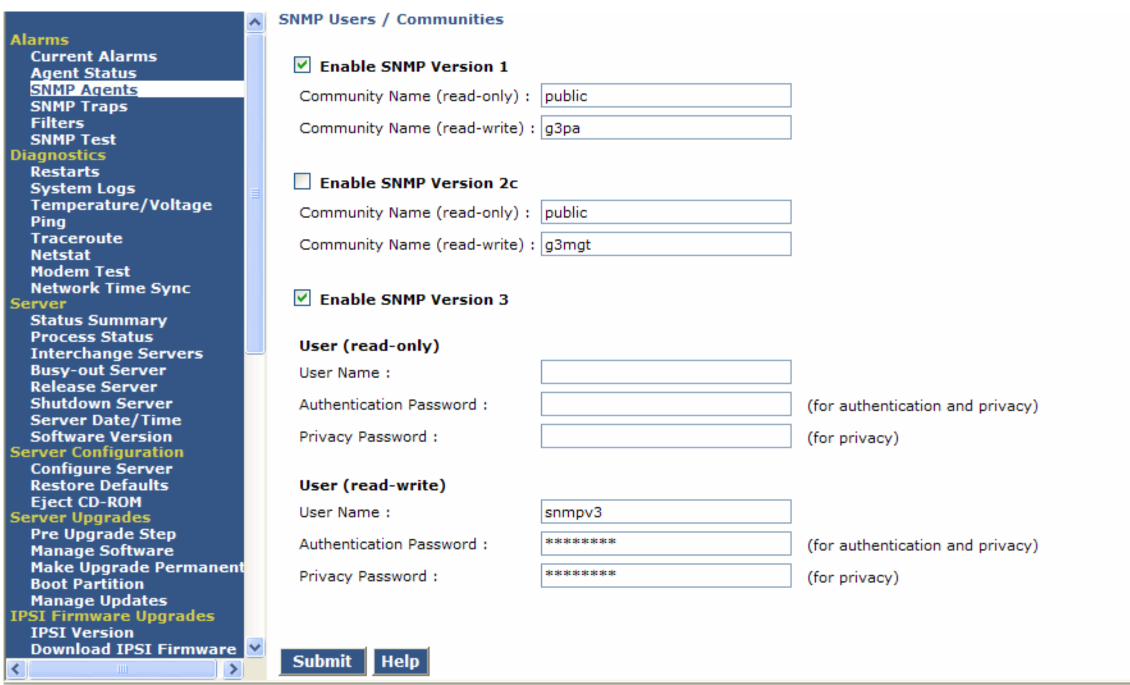
Use the following steps to administer an SNMP Agent:

1. On the server's maintenance web interface, click **SNMP Agents**.

   The **SNMP Agents** screen appears as shown in example Figure 210 and Figure 211.

Figure 210: SNMP Agents screen part 1

Figure 211: SNMP Agents screen part 2



2. Check the status of the Master Agent. If the status of the Master Agent is "up": Select **Agent Status** from the navigation bar and click **Stop Agent**. Once the Master Agent is in a "Down" state, return to the **SNMP Traps** screen by clicking **SNMP Traps** on the navigation bar. If the Master Agent is in a "Down" state, continue with step 3.
3. In the **IP Addresses for SNMP Access** section:
   a. Select the radio button associated with one of the following options:
      - No access: This option restricts all IP address from talking to the agent.
      - Any IP access: This option allows all IP addresses to access the agent.
      - Following IP addresses: You can specify up to five individual IP addresses that has permission to access the agent.
4. In the SNMP users/communities section: Select one or more versions of SNMP by clicking on the **Enable** box associated with the version.
   - **SNMP Version 1**:
     a. **Enable SNMP Version 1**: Check this box to enable SNMP v1. If the SNMP v1 box is enabled, SNMP v1 can communicate with the SNMP agents on the server.
     b. **Community Name (read-only)**: When this option is selected the community or the user can query for information only (SNMPGETs).
     c. **Community Name (read-write)**: When this option is selected the community or the user can not only query for information but can also send commands to the agents (SNMPSETs).
   - **SNMP Version 2**: Check this box to enable SNMP v2. If the SNMP v2 box is enabled, SNMP v2 can communicate with the SNMP agents on the server.
     a. **Enable SNMP Version 2**: Check this box to enable SNMP v2.
     b. **Community Name (read-only)**: When this option is selected the community or the user can query for information only (SNMPGETs).
     c. **Community Name (read-write)**: When this option is selected the community or the user can not only query for information but can also send commands to the agents (SNMPSETs).
   - **SNMP Version 3**: SNMP v3 provides the same data retrieval facilities as the previous versions with additional security. A User Name, authentication password, and privacy password are used to provide a secure method of authenticating the information so the device knows whether to respond to the query or not.
     a. **Enable SNMP Version 3**: Check this box to enable SNMP v3. If the SNMP v3 box is enabled, SNMP v3 can communicate with the SNMP agents on the server.

User (read-only) - Entering a user name, authentication password, and security password in this section provides the user with read functionality only.

b. **User Name**: Enter a User Name. The User Name can be a maximum of any 50 characters with the exception of quotation marks.

c. **Authentication Password**: Enter a password for authenticating the user. The authentication password must be a maximum of any 50 characters with the exception of quotation marks.

d. **Privacy Password**: Enter a password for privacy. The privacy password can contain any 8 to 50 characters with the exception of quotation marks.

User (read-write) - Entering a user name, authentication password, and security password in this section provides the user with read and write functionality.

e. **User Name**: Enter a User Name. The User Name can be a maximum of any 50 characters with the exception of quotation marks.

f. **Authentication Password**: Enter a password for authenticating the user. The authentication password must be a maximum of any 50 characters with the exception of quotation marks.

g. **Privacy Password**: Enter a password for privacy. The privacy password can contain any 8 to 50 characters with the exception of quotation marks.

5. To save the changes, click **Submit**.
6. Once you are finished adding the SNMP Agent, you must start the Master Agent. To start the Master Agent, select **Agent Status** from the server's maintenance web page and click **Start Agent.**

Important:
You can use the **Agent Status** screen to change the state of the Master Agent and to check the state of the subagents. If the subagent is connected to the Master Agent, the status of each subagent is "Up." If the status of the Master Agent is "Down" and the status of the subagent is "Up," the subagent is not connected to the Master Agent.

Administering filters

Use the **SNMP Filters** screen to perform the following tasks:

- Adding a filter on page 629
- Changing a filter on page 633
- Deleting one or all filters on page 634
- Customer Alarm Reporting Options on page 634

The filters are used only for Communication Manager and determine which alarms are sent as traps to the trap receiver(s) that are administered using the **SNMP Traps** page. For more information on how to administer an SNMP trap, see Administering traps on page 619.

Important:

Filters created by Fault and Performance Manager (FMP) do not display on the **SNMP Filters** screen. If you are using FMP, create the filters using the FMP application. The FMP application provide some additional capabilities that are not available using the **SNMP Filters** screen.

Adding a filter

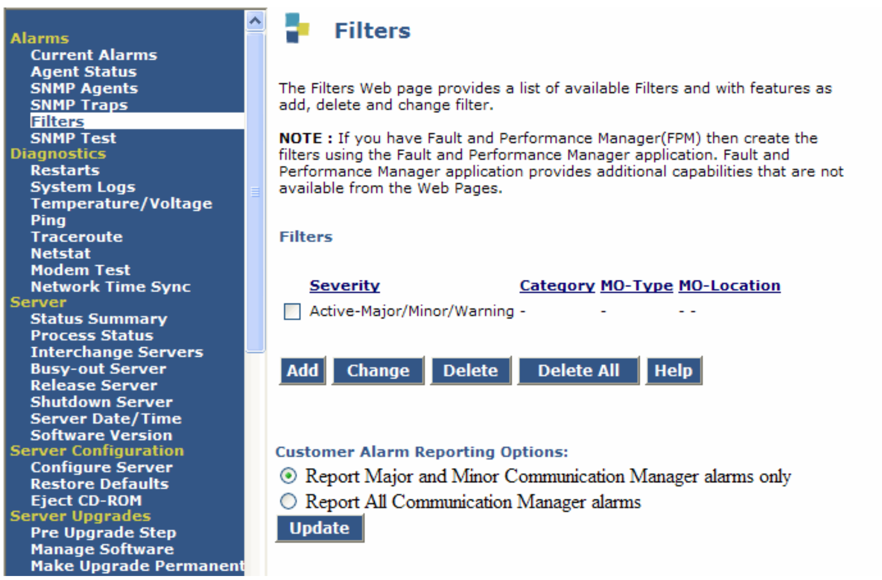
Use the following steps to add a filter.

Instructions

1. On the server's maintenance web interface, click **SNMP Filters** under the **Alarms** heading.

   The **SNMP Filters** screen appears as shown in example Figure 212.

Figure 212: SNMP Filters screen



2. Click **Add**. The **Add Filter** screen appears as shown in example Figure 213.

Figure 213: SNMP Filters screen - Add Filter

3. **Severity**: Select from one or more of the following alarm severities that will be sent as a trap:
   - Active
   - Major
   - Minor
   - Warning
   - Resolved
4. **Category and MO-Type**: Select the alarm category for this filter from the drop-down menu. The **MO-Types** that display are based on the **Category** that you select. The available categories with their associated MO-Types are listed in Table 10.

Table 10: Category with associated MO-Type table

Category	MO-Type
adm-conn	ADM-CONN
announce	ANN-PT, ANN-BD, ANNOUNCE
atm	ATM-BCH, ATM-DCH, ATM-EI, ATM-INTF, ATM-NTWK, ATM PNC-DUP, ATM-SGRP, ATM-SYNC, ATM-TRK, ATM-WSP
bri/asai	ASAI-ADJ, ASAI-BD, ASAI-PT, ASAI-RES, ABRI-PORT, BRI-BD, BRI-PORT, BRI-SET, LGATE-AJ, LGATE-BD, LGATE-PT
cdr	CDR-LINK
	1 of 4

Table 10: Category with associated MO-Type table (continued)

Category	MO-Type
data-mod	BRI-DAT, DAT-LINE, DT-LN-BD, PDMODULE, TDMODULE
detector	DTMR-PT, DETR-BD, GPTD-PT, TONE-BD
di	DI-BD, DI-PT
environ	AC-POWER, CABINET, CARR-POW, CD-POWER, EMG-XFER, EXT-DEV, POWER, RING-GEN
esm	ESM
exp-intf	AC-POWER, CARR-POWER, DC-POWER, EPN-SNTY, EXP-INTF, MAINT, SYNC, TDM-CLK, TONE-BD
ext-dev	CUST-ALM
generatr	START-3, SYNC, TDM-CLK, TONE-PT, TONE-BD
inads-link	INADS
infc	EXP-INTF
ip	MEDPRO, IPMEDPRO, MEDPORPT, H323-SGRP, H323-BCH, H323-STN, DIG-IP-STN, RDIG-STA, RANL-STA, NR-CONN, REM-FF, ASAI-IP, ADJLK-IP, SIP-SGRP
lic-file	NO-LIC, LIC-ERR
maint	MAINT
misc	CONFIG, ERR-LOG, MIS, PROC-SAN, SYSTEM, TIME-DAY
mmi	MMI-BD, MMI-LEV, MMI-PT, MMI-SYNC
mnt-test	M/T-ANL, M/T-BD, M/T-DIG, M/T-PT
modem	MODEM-BD, MODEM-PT
pkt	M/T-PKT, PKT-BUS
pms/jrnl	JNL-PRNT, PMS-LINK
pns	DS1C-BD, DS1-FAC, EXP-INTF, FIBER-LK, PNC-DUP, SN-CONF, SNC-BD, SNC-LINK, SNC-REF, SNI-BD, SNI-PEER
pncmaint	DS1C-BD, DS1-FAC, EXP-INTF, FIBER-LK, PNC-DUP, SN-CONF,SNC-BD, SNC-LINK, SNC-REF, SNI-BD
pnc-peer	SNI-PEER
procr	PROCR
	2 of 4

Table 10: Category with associated MO-Type table (continued)

Category	MO-Type
quick-st	ABRI-PT, ADXDP-PT, ANL-16-LINE, ANL-LINE, ANL-NE-LINE, ANN-PT, ANNOUNCE, ASAI-ADJ, AUDIX-PT, AUX-TRK, BRI-PT, BRI-SET, CDR-LINK, CLSFY-PT, CO-DSI, CO-TRK, CONFIG, DAT-LINE, DID-DS1, DID-TRK, DIG-LINE, DIOD-TRK, DS1-FAC, DS1C-BD, DTMR-PT, EPN-SANITY, EXP-INTF, EXP-PN, FIBER-LINK, GPTD-PT, HYB-LINE, ISDN-LNK, ISDN-TRK, JNL-PRNT, MAINT, MET-LINE, MODEM-PT, OPS-LINE, PDATA-PT, PDMODULE, PKT-BUS, PKT-INT, PMS-LINK, PMS-PRNT, PNC-DUP, PRI-CDR, S-SYN-PT, SN-CONF, SNC-BD, SNC-LNK, SNC-REF, SNI-BD, SNI-PEER, SYS-PRNT, SYSLINK, SYSTEM, TDM-BUS, TDM-CLK, TDMODULE, TIE-DS1, TIE-TRK, TONE-BD, TTR-LEV
sch-adj	SCH-ADJ
s-syn	S-SYN-BD, S-SYN-PT
stabd	ABRI-PORT, ADXDP-BD, ADXDP-PT, ANL-16-LINE, ANL-BD, ANL-LINE, ANL-NE-LINE, ASAI-ADJ, AUDIX-BD, AUDIX-PT, BRI-BD, BRI-PORT, BRI-SET, DIG-BD, DIG-LINE, HYB-BD, HYB-LINE, MET-BD, MET-LINE
stacrk	ADXDP-PT, ANL-LINE, ANL-16-LINE, ANL-NE-LINE, AUDIX-PT, DIG-LINE, HYB-LINE, MET-LINE, OPS-LINE
stations	ABRI-PORT, ADXDP-PT, ANL-16-LINE, ANL-LINE, ANL-NE-LINE, ASAI-ADJ, AUDIX-PT, BRI-PORT, BRI-SET, DIG-LINE, HYB-LINE, MET-LINE, OPS-LINE
sys-link	SYS-LINK
sys-prnt	SYS-PRNT
tdm	TDM-BUS
tone	CLSFY-BD, CLSFY-PT, DETR-BD, DTMR-PT, GPTD-PT, START-E, SYNC, TDM-CLK, TONE-BD, TONE-PT, TTR-LEV
trkbd	AUX-BD, AUX-TRK, CO-BD, CO-DS1, CO-TRK, DID-BD, DID-DS1, DID-TRK, DIOD-BD, DIOD-TRK, DS1-BD, ISDN-TRK, PE-BCHL, TIE-BD, TIE-DS1, TIE-TRK, UDS1-BD, WAE-PT
trkcrk	AUX-TRK, CO-DS1, C9-TRK, DID-DS1, DID-TRK, DIOD-TRK, ISDN-LNK, ISDN-TRK, TIE-DS1, TIE-TRK
trunks	CO-TRK, SUX-TRK, CO-DS1, DID-DS1, DID-TRK, DIOD-TRK, ISDN-LNK, ISDN-TRK, PE-BCHL, TIE-DS1, TIE-TRK, WAE-PORT
vc	VC-BD, VC-DSPPT, VC-LEV, VC-SUMPT
	3 of 4

Table 10: Category with associated MO-Type table (continued)

Category	MO-Type
vsp	MMI-BD, MMI-PT, MMI-LEV, MMI-SYNC, VC-LEV, VC-BD, VC-SUMPT, VC-DSPPT, VP-BD, VP-PT, VPP-BD, VPP-PT, DI-BD, DI-PT, MEDPRO, IPMEDPRO, MEDPROPT
wide-band	PE-BCHL, WAE-PORT
wireless	RC-BD, RFP-SYNC, WFB, CAU, WT-STA
	4 of 4

5. MO Location: Select an MO Location from the following list:
   - Media Gateway
   - Cabinet
   - Board
   - Port
   - Extension
   - Trunk Group/Member
6. To add the filter, click **Add**.

   The **Filters** screen appears displaying the new filter.

Changing a filter

Use the following steps to change a filter:

1. From the server's maintenance web interface, click **SNMP Filter** under the **Alarms** heading.

   The **Filters** screen appears.
2. Click the box associated with the filter you wish to change and press **Change**.

   The **Change Filters** screen appears.
3. Make the desired changes to the filter and press **Change**.

   The **Filters** screen appears displaying the changes made to the filter.

Deleting one or all filters

Use the following steps to delete one or all filters:

1. To delete all the filters, click **Delete All**.

   The system displays a warning message asking if you are sure. If you wish to continue, click **OK**. The **Filters** screen appears.

2. To delete one filter, click the box associated with the filter you wish to delete and press **Delete**.

   The system displays a warning message asking if you are sure. If you wish to continue, click **OK**. The **Filters** screen appears with the updated list of filters.

Customer Alarm Reporting Options

The **Customer Alarm Reporting Options** sections allows you to select from one of the following reporting options:

- Report Major and Minor Communication Manager alarms only
- Report All Communication Manager alarms

Use the following steps to set the **Customer Alarm Reporting Options**:

1. Click the radio button associated with the desired reporting option.
2. Click **Update**

   The **Filters** screen appears displaying the selected reporting option.

Chapter 17: Collecting Call Information

Collecting Information About Calls

Call Detail Recording (CDR) collects detailed information about all incoming and outgoing calls on specified trunk groups. If you use Intra-switch CDR, you can also collect information about calls between designated extensions on Avaya Communication Manager. Communication Manager sends this information to a printer or to some other CDR output device that collects call records and that might also provide reports.

You can have a call accounting system directly connected to your Avaya S8XXX Server running Communication Manager. If you are recording call details from several servers, Communication Manager can send the records to a collection device for storage. A system called a poller can then take these records and send them to the call accounting system. The call accounting system sorts them, and produces reports that you can use to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls.

Before you start

The call accounting system that you use might be sold by Avaya, or it might come from a different vendor. You need to know how your call accounting system is set up, what type of call accounting system or call detail recording unit you are using, and how it is connected to the server running Communication Manager. You also need to know the format of record that your call accounting system requires.

CAUTION:

When migrating a platform from a legacy system to a Linux-based system of Communication Manager 3.0 or newer, where both the old and new systems utilize CDR, ensure that the older CDR parsing scripts correctly utilize all of the characters identified in each of the fields contained in the applicable format table (see the Format Tables in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205).

Instructions

In this example, we are going to establish call detail recording for all calls that come in on trunk group 1 (our CO trunk). We are going to set up CDR so that any call that is handled by an attendant produces a separate record for the attendant part of the call.

1. Type **change trunk-group n**. Press **Enter**.

   The Trunk Group screen - page 1 appears.

Figure 214: Trunk Group screen

```
add trunk-group next                                            Page   1 of   x
                                TRUNK GROUP

Group Number: 1                    Group Type: co            CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1         TN: 1         TAC:
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0          Country: 1             Incoming Destination:
   Comm Type: voice                Auth Code? n   Digit Absorption List:
    Prefix-1? y                  Trunk Flash? n         Toll Restricted? y

         Trunk Type:
```

2. In the **CDR Reports** field, type **y**.

   This tells Communication Manager to create call records for calls made over this trunk group.

3. Press **Enter** to save your changes.

4. Type **change system-parameters cdr**. Press **Enter**.

   The CDR System Parameters screen appears.

Figure 215: CDR System Parameters

```
change system-parameters cdr                                Page   1 of   x
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                     CDR Date Format: month/day
      Primary Output Format: printer       Primary Output Endpoint: CDR1
    Secondary Output Format:
           Use ISDN Layouts? n                   Enable CDR Storage on Disk? n
       Use Enhanced Formats? n      Condition Code 'T' for Redirected Calls? n
     Use Legacy CDR Formats? y                  Remove # From Called Number? n
Modified Circuit ID Display? n                             Intra-switch CDR? n
                  Record Outgoing Calls Only? y     Outg Trk Call Splitting? n
  Suppress CDR for Ineffective Call Attempts? y          Outg Attd Call Rec? y
      Disconnect Information in Place of FRL? n      Interworking Feat-flag? n
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
 Record Agent ID on Incoming? n      Record Agent ID on Outgoing? n
      Inc Trk Call Splitting? n
   Record Non-Call-Assoc TSC? n
       Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
    Privacy - Digits to Hide: 0               CDR Account Code Length: 4
```

5. In the **CDR Format** field, type **month/day**.

   This determines how the date will appear on the header record.

6. In the **Primary Output Format** field, enter **Unformatted**.

   This is the record format that our call accounting system requires. Check with your call accounting vendor to determine the correct record format for your system.

7. In the **Use Legacy CDR Formats** field, enter **y** to use CDR formats from Communication Manager 3.1 and earlier. Enter **n** to use formats from Communication Manager 4.0 and later. (For more information, see the *Screen Reference* chapter, Use Legacy CDR Formats field.)

8. In the **Primary Output Ext.** field, type **2055**.

   This is the extension of the data module that we use to connect to our call accounting system.

9. In the **Record Outgoing Calls Only** field, type **n**.

   This tells Communication Manager to create records for both incoming and outgoing calls over all trunk groups that use CDR.

10. In the **Outg Trk Call Splitting** and **Inc Trk Call Splitting** fields, type **y**.

    This tells the system to create a separate record for any portion of an incoming or outgoing call that is transferred or conferenced.

11. In the **Outg Att Call Record** and **Inc Att Call Record** fields, type **y**.

    This tells the system to create a separate record for the attendant portion of any incoming or outgoing call.

More information

You can also administer Communication Manager to produce separate records for calls that are conferenced or transferred. This is called Call Splitting. There are many other variations that you can administer for CDR, and these are described in the screens section of this book.

Related topics

For additional information on Call Detail Recording (CDR), see *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Setting up Intra-switch CDR

Call detail recording generally records only those calls either originating or terminating outside the server running Communication Manager. There might be times when you need to record calls between users on the local server. Intra-switch CDR lets you track calls made to and from local extensions.

Instructions

In this example, we administer Communication Manager to record all calls to and from extensions 5100, 5101, and 5102.

1. Type ***`change system-parameters cdr`***. Press **Enter**.

   The [CDR System Parameters](#) screen appears.

2. In the **intra-switch CDR** field, type **y**.
3. Press **Enter** to save your changes.
4. Type **`change intra-switch-cdr`**. Press **Enter**.

   The [Intra-Switch CDR screen](#) appears.

Figure 216: Intra-Switch CDR screen

```
change intra-switch-cdr                                          Page   1 of   3
                           INTRA-SWITCH CDR

                                Assigned Members:   0    of 5000   administered
      Extension            Extension          Extension           Extension

    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
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    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________
    _____________        _____________      _____________       _____________

Use 'list intra-switch-cdr' to see all members.
Use 'change intra-switch-cdr <ext>' to add/change/remove other members.
```

5. In the first three available slots, type **5100**, **5101**, and **5102**.
6. Press **Enter** to save your changes.
7. Communication Manager will now produce call records for all calls to and from these extensions, including those that originated on the local server.

Related topics

See Intra-Switch CDR on page 1164 for more detailed information.

Tracking Calls by Account Code

You can have your users to enter account codes before they make calls. By doing this, you can have a record of how much time was spent on the telephone doing business with or for a particular client.

Instructions

In this example, we are going to set up the system to allow the user at extension 5004 to enter a 5-digit account code before making a call.

1. Type **`change system-parameters cdr`**. Press **Enter**.

   The **CDR System Parameters** screen appears.

Figure 217: CDR System Parameters screen

```
change system-parameters cdr                                Page   1 of   x
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                     CDR Date Format: month/day
      Primary Output Format: printer       Primary Output Endpoint: CDR1
    Secondary Output Format:
           Use ISDN Layouts? n                   Enable CDR Storage on Disk? n
       Use Enhanced Formats? n      Condition Code 'T' for Redirected Calls? n
     Use Legacy CDR Formats? y                  Remove # From Called Number? n
Modified Circuit ID Display? n                             Intra-switch CDR? n
                  Record Outgoing Calls Only? y     Outg Trk Call Splitting? n
  Suppress CDR for Ineffective Call Attempts? y         Outg Attd Call Rec? y
      Disconnect Information in Place of FRL? n      Interworking Feat-flag? n
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
 Record Agent ID on Incoming? n      Record Agent ID on Outgoing? n
      Inc Trk Call Splitting? n
   Record Non-Call-Assoc TSC? n
       Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
    Privacy - Digits to Hide: 0              CDR Account Code Length: 4
```

2. In the **CDR Account Code Length** field, type **5**.
3. Press **Enter** to save your changes.
4. Assign an account button on the **Station** screen for extension 5004. See Adding Feature Buttons on page 129 for more information.

5. Provide your users with a list of account codes to use.
6. You can also assign a feature access code and give this to your users.

Forcing Users to Enter Account Codes

Forced Entry of Account Codes is another form of account code dialing. You can use it to allow certain types of calls only with an account code, to track fax machine usage by account, or just to make sure that you get account information on all relevant calls.

Before you start

Before you can administer Forced Entry of Account Codes, it must be enabled on the **System Parameters Customer-Options (Optional Features)** screens. If it is not, please contact your Avaya representative.

Instructions

In this example, we administer the system to force users in our North American office to enter an account code before making international calls.

1. Type **`change system-parameters cdr`**. Press **Enter**.

   The **CDR System Parameters** screen appears.
2. In the **Force Entry of Acct Code for Calls Marked on Toll Analysis Form** field, type **y**.
3. In the **CDR Account Code Length** field, type **5**.
4. Press **Enter** to save your changes.
5. Type **`change toll 0`**. Press **Enter**.
6. The **Toll Analysis** screen appears.

Figure 218: Toll Analysis screen

```
change toll n                                                       Page  1 of  x
                                TOLL ANALYSIS                     Percent Full:_
                               Location:
                      Total          Toll  CDR  <--Unrestricted Call List-->
    Dialed String    Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
_____________________ ___  ___   _      _     _   _  _  _  _  _  _  _  _  _  _
```

7. In the first available **Dialed String** field, type **011**.

   This is the international access code for this office.
8. In the **Total Min** and **Max** columns, type **10** and **18**, respectively.

   This is the minimum and maximum number of digits the system will analyze when determining how to handle the call.
9. In the **Toll List** and **CDR FEAC** columns, type **x**.
10. Press **Enter** to save your changes.

More information

You can also establish a class of restriction with **Forced Entry of Account Codes** set to **y**, and assign this class of restriction (COR) to trunks or other facilities that you want to restrict. With this method, all users with this COR must enter account codes before making any outgoing trunk calls. See Class of Restriction on page 834 for more information.

Receiving Call-Charge Information

Avaya Communication Manager provides two ways to receive information from the public network about the cost of calls. Note that this service is not offered by the public network in some countries, including the US.

- Advice of Charge (AOC, for ISDN trunks) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it might be recorded as either a charging or currency unit.
- Periodic Pulse Metering (PPM, for non-ISDN trunks) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Before you start

You need to request either AOC or PPM service from your network provider. In some areas, your choice might be limited. Your Avaya technical support representative can help you determine the type of service you need.

Note:

This service is not offered by the public network in some countries, including the U.S.

Collecting call charge information over ISDN

In this example, we administer the system to provide Advice of Charge over an existing ISDN trunk group, at the end of a call. This information will appear on CDR reports.

1. Type **`change trunk-group 2`**.

   The **ISDN Trunk Group screen** appears with existing administration for this ISDN trunk group.

Figure 219: ISDN Trunk Group screen

```
add trunk-group next                                                Page   1 of  xx
                                TRUNK GROUP

Group Number: 1                   Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC:
   Direction: outgoing       Outgoing Display? n        Carrier Medium: PRI/BRI
 Dial Access? n                Busy Threshold: 255
Queue Length: 0
Service Type:                                              TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

2. In the **CDR Reports** field, type **y**.

   This ensures that the AOC information appears on the CDR report.

3. Verify that **Service Type** is **public-ntwrk**.
4. In the **Supplementary Service Protocol** field, type **a**.
5. The **Charge Advice** field, type **end-on-request**.

   This ensures that Communication Manager will place one request for charge information. This reduces the amount of information passed to Communication Manager and consumes less processor time than other options.

6. Press **Enter** to save your changes.

Administering Charge Advice for QSIG trunks

Use the QSIG Supplementary Service - Advice of Charge feature to extend charging information from the public network into the private network. The charging information that many service providers supply is extended from a gateway enterprise system to the end user's enterprise system. The charging information can then be displayed on the user's desktop.

Information can be extended and displayed either:

- At intervals during the call and at the end of the call, or
- Only at the end of the call

QSIG stands for Q-Signaling, which is a common channel signal protocol based on ISDN Q.931 standards and used by many digital telecommunications systems. Only charge information received from the public network with ETSI Advice of Charge, and Japan Charge Advice is extended into the QSIG private network.

To administer charge advice for a trunk group with Supplementary Services Protocol **b**:

1. On the **Trunk Group** screen, for Group Type **ISDN**, <tab> to the **Charge Advice** field.

2. Select from the following options:
   - during-on-request - to request that charging information be provided at intervals during a call, and also at the end of the call
   - end-on request - to request that charging information be provided only at the end of a call
   - none - no charging information will be requested for the trunk group

   Note:

   Receipt of charge advice on the QSIG trunk group is also dependent on Charge Advice administration at the PSTN trunk group involved on the call, and whether charges are received from the public network.

- On the **Trunk Group** screen, administer the **Decimal Point** field.
   - period (.) -This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point.
   - comma (,) - If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point.

If the received charge contains no decimals, no decimal point is displayed (i.e., the administered decimal point is ignored for charge information received with no decimals). On an upgrade from a QSIG trunk group with the **Decimal Point** field administered as **none**, the field defaults to **period**.

Receiving call-charge information over non-ISDN trunks

In this example, we will administer an existing Direct Inward and Outward Dialing (DIOD) trunk to receive PPM from the public network.

1. Type **`change trunk-group 3`**.

   Page 1 of the **Trunk Group** screen appears with existing administration for this trunk group. Click the numbered page tabs or **Next Page** to find fields that appear on subsequent pages of the **Trunk Group** screen.
2. In the **CDR Reports** field, type **y**.

   This ensures that the PPM information appears on the CDR report.
3. In the **Direction** field, type **two-way**.
4. Click Next Page to find the **PPM** field.
5. In the **PPM** field, type **y**.
6. In the **Frequency** field, type **50/12**.

   This is the signal frequency (in kHz). The frequency you will use depends on what the circuit pack you use is able to accept. See Tone Generation on page 1663 for more information.
7. In the **Administrable Timers** section, set the **Outgoing Glare Guard** timer to **5** seconds.

8. Press **Enter** to save your changes.
9. You also need to ensure that the values of the **Digital Metering Pulse Minimum**, **Maximum** and **Value** on the **DS1 Circuit Pack** screen are appropriate to the values offered by your service provider. See DS1 Circuit Pack on page 945 for more information.

Related topics

For more information about AOC and PPM, see "Call Charge Information" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Viewing Call Charge Information

Avaya Communication Manager provides two ways for you to view call-charge information: on a telephone display or as part of the Call Detail Recording (CDR) report. From a display, users can see the cost of an outgoing call, both while the call is in progress and at the end of the call.

Instructions

In this example, we administer extension 5040 to be able to view the charge of a call in progress. The charges will appear in currency units (in this case, Lira) on the user's telephone display.

1. Type `change trunk-group 2`. Press **Enter**. Click **Next Page** until the **Trunk Features** section appears.
2. In the **Charge Conversion** field, type **200**.

   This indicates that one charge unit sent from the service provider is equal to 200 units, in this case, Lira.
3. In the **Decimal Point** field, type **none**.
4. In the **Charge Type** field, type **Lira**.
5. Press **Enter** to save your changes.
6. Type `change system-parameters features`. Press **Enter**.

   The **Feature-Related System Parameters** screen appears.
7. In the **Charge Display Update Frequency (seconds)** field, type **30**.

   Frequent display updates might have considerable performance impact.

8. Press **Enter** to save your changes.
9. Now assign extension 5040 a **disp-chrg** button to give this user the ability to control the charge display. See Adding Feature Buttons on page 129 for more information.

If you want end users to control when they view this information, you can assign a display button that they can press to see the current call charges. If you want call charges to display automatically whenever a user places an outgoing call, you can set **Automatic Charge Display** to **y** on the user's **COR** screen.

Setting up Survivable CDR

The Survivable CDR feature is used to store CDR records to a server's hard disk. For ESS servers and LSPs, the Survivable CDR feature is used to store the CDR records generated from calls that occur when an LSP or ESS server is controlling one or more gateways or port networks. For a man server, the Survivable CDR feature provides the ability to store CDR records on the server's hard disk.

When the Survivable CDR feature is enabled, the CDR records are saved in a special directory named /var/home/ftp/CDR on the server's hard disk. The CDR adjunct retrieves the Survivable CDR data files by logging into the server and copying the files to its own storage device. The CDR adjunct uses a special login that is restricted to only accessing the directory where the CDR records are stored. After all the files are successfully copied, the CDR adjunct deletes the files from the server's hard disk and processes the CDR records in the same manner that it does today.

Note:

This feature is available on main servers and ESS servers that are Communication Manager 5.0 and later releases only. It is available on LSP platforms running Communication Manager 4.0 and later.

The CDR adjunct must poll each main, LSP, and ESS server regularly to see if there are any new data files to be collected. This is required even when an LSP or ESS server is not controlling a gateway or a port network because the CDR adjunct has no way of knowing if a LSP or ESS server is active.

The Survivable CDR feature utilizes the same CDR data file formats that are available with legacy CDR.

CDR files

When Survivable CDR is enabled, the server writes the CDR data to files on the hard disk instead of sending the CDR data over an IP link. The Survivable CDR feature creates two types of CDR data files, a Current CDR data file that the server uses to actively write CDR data and a set of archive files containing CDR data that the server collected earlier but has not yet been collected and processed by the CDR adjunct. The naming convention for both file types are similar, however the name of the Current CDR file is always prefixed by a 'C-' (for more information see CDR file naming convention on page 648). The CDR Current file remains active until one of the following events happen:

- The server's system clock reaches 12:00 midnight
- The Current CDR file reaches or exceeds 20 megabytes. A 20 megabyte file may contain up to 140K CDR records depending on the CDR format used.
- A filesync, a reset system 2 (cold restart), or a reset system 4 (reboot) occurs

After one of the above events occur the following actions take place:

- The Current CDR file is closed and it becomes an archive CDR file
- The file permissions change from read/write (rw) for root and read only for members of the CDR_User group to:
  - Owner (root): Read/Write/Execute (rwx)
  - Group (CDR_User): Read/Write (rw-)
  - World: none (---)
- The 'C-' prefix is removed from the front of the file name
- For a main server, a new Current CDR file is created
- For an LSP or an ESS server, a new Current CDR file is created only if the LSP or the ESS server is controlling one or more gateways or port networks.

CDR file naming convention

The CDR data files have the following naming convention:

tssssss-cccc-YYMMDD-hh_mm

where:

- t is populated with an L for an LSP, an E for an ESS server, or an S for a main server
- ssssss is populated with the least significant six digits of the System ID or SID. The SID is a unique number in the RFA license file used to identify the system. The SID for a server can be viewed by using one of the following methods:
  - Use the **statuslicense -v** BASH command.
  - Use the command **display system-parameters customer-options** on the SAT.
- cccc is populated with the least significant four digits of the Cluster ID (CL ID) or Module ID (MID). To display the MID for the server:
  - Use the **statuslicense -v** BASH command.
- YY is populated with the two digit number of the year the file was created.

- MM is populated with the two digit number of the month the file was created.
- DD is populated with the two digit day of the month the file was created.
- hh is populated with the hour of the day the file was created based on a 24 hour clock.
- mm is populated with the number of minutes after the hour when the file was created.

The Current CDR file uses the same naming convention except the name is prefixed with a ’C-’.

Removing CDR files

The CDR files can be removed by:

- The Survivable CDR feature: The Survivable CDR feature on the main, LSP, or ESS server automatically removes the oldest CDR data achieve file anytime the number of archived files exceed 20. The Current CDR file is not an archived file and therefore not counted in the 20 files allowed on the hard disk.
- CDR adjunct: In a normal operating environment, the CDR adjunct has the responsibility to delete the CDR data files after they are copied and verified that they are correct.

Access to CDR files

A special user group called CDR_User exists that allows the administrator to identify all users authorized to access the CDR storage directory. The archived CDR files are stored in /var/home/ftp/CDR.

High level administration

Use the following high level steps to administer the Survivable CDR feature:

1. Create a new user account to allow the CDR adjunct access and permissions to retrieve CDR data files (see Creating a new user account on page 650).
2. Enable CDR storage on the hard disk (see Survivable CDR administration for the main server on page 651).
3. If using this feature on the main server: Administer the **Primary Output Endpoint** field on the main’s **`change system-parameters cdr`** SAT form to be **DISK** (see Survivable CDR administration for the main server on page 651). When using Survivable CDR, only the **Primary Output Endpoint** field is available. Administration of the **Secondary Output Endpoint** field is blocked.
4. If you are using this feature on an LSP and an ESS server: Administer the "**Enable CDR Storage on Disk**" field on the **`change survivable-processor`** form (see Survivable CDR administration for an LSP or ESS server on page 651).

Detailed administration

Creating a new user account

For the CDR adjunct to access the CDR data files, a new user account must be created on the main server. The new account is pushed to the LSP and/or ESS server when a filesync is performed.

To create a new user account:

1. On the server's **Integrated Management Maintenance Web Page**, click **Administrator Accounts** under the **Security** heading.

   The **Administrator Accounts** page displays.
2. Enter the login ID for the new user in the **Enter Login ID or Group Name** field.
3. Click the **Add Login** radio button and then click **Submit**.

   Administrator Logins -- Add Login page displays.
4. Enter the data in Table 11 in each field of the **Administrator Logins -- Add Login** page.

Table 11: CDR adjunct user account recommended options

Field Name	Recommended Option
Login Name	Any valid user name chosen by the administrator or installer
Login group	CDR_User
Shell:	Select CDR access only by clicking the associated radio button.
Lock this account	Leave blank
Date on which the account is disabled	Leave blank
Select type of authentication	Password
Enter key or password	Any valid password chosen by the administrator or installer
Re-enter key or password	Re-enter the above password
Force password/key change on first login	no
	1 of 2

Table 11: CDR adjunct user account recommended options (continued)

Field Name	Recommended Option
Maximum Number of days a password may be used (PASS_MAX_DAYS)	99999
Minimum number of days allowed between password changes (PASS_MIN_DAYS)	0
Number of days warning given before a password expires (PASS_WARN_AGE)	7
Days after password expires to lock account	-1
	2 of 2

5. Click **Add** to create the new user account.

Survivable CDR administration for the main server

Use the following steps to administer the Survivable CDR feature for a main server:

1. On page 1 of the CDR System Parameters screen:
   a. **Enable CDR Storage on Disk?**: Possible entries for this field are yes or no. Entering yes (**y**) in this field enables the Survivable CDR feature for the main, LSP, and ESS servers. If this field is set to no (**n**), the CDR functionality remains as legacy CDR.
   b. **Primary Output Endpoint**: Possible entries for this field are **CDR1**, **CDR2**, and **DISK**. For the main server, the **Primary Output Endpoint** field must be set to **DISK**. When Survivable CDR is administered as **Disk** on the **Primary Output Endpoint** field, the **Secondary Output Endpoint** field is blocked.

Survivable CDR administration for an LSP or ESS server

Use the following steps to administer the Survivable CDR feature for an LSP or an ESS server:

Note:
The Survivable CDR feature is administered on the main server for the LSP and ESS servers.

Important:
An LSP or ESS server only stores Survivable CDR records if it is administered to support Survivable CDR and if it is controlling one or more gateways or port networks.

1. On the **CDR System Parameters** screen:
   a. **Enable CDR Storage on Disk?**: Possible entries for this field are yes or no. Entering yes in this field enables the Survivable CDR feature for the main, LSP, and ESS servers. If this field is set to no, the CDR functionality remains legacy CDR.
2. **Survivable Processor - IP-Services screen**, page 3:
   a. **Service Type**: The **Service Type** field must be set to **CDR1** or **CDR2** to enable entries to the **Store to Dsk** field.
   b. **Store to Dsk**: Enter **y** to enable Survivable CDR for this LSP or ESS server. When the **Service Type** field is set to **CDR1** or **CDR2** and the **Store to Dsk** field is set to **y**, all CDR data for the specific LSP or ESS server being administered will be sent to the hard disk rather than output to an IP link. LSP or ESS servers will only store CDR records to hard disk when the LSP or ESS server is controlling a gateway or port network.

Important:
You must complete the **Survivable Processor - IP Services** screen for ***each*** LSP or ESS server that will utilize the Survivable CDR feature.

Note:
The **Enable** field for a given line in the **Survivable Processor - IP Services** screen must be set to **o** (overwrite) to allow changes for that row.

Chapter 18: Telephone Reference

This reference section describes many of the telephones and adjuncts that you can connect to servers running Avaya Communication Manager.

Use this section to:

- determine where to connect a telephone—is it analog, digital, hybrid, or IP? Is it designed for a 2-wire or 4-wire environment?
- determine whether a telephone will fit your users' needs—does it accommodate enough feature buttons? does it include a display?
- understand how to assign the user's feature buttons—how do the buttons on the **Station** screen map to buttons on the physical telephone.

Because you need to know how the physical buttons on the telephone relate to your button assignments on the **Station** screen, this section includes figures for those telephones to which you can assign feature buttons.

This section includes descriptions of the following telephones:

500 telephones

The 500 telephones are single appearance analog rotary-dial telephones which provides cost-effective service wherever it is located. It provides limited access to features because the rotary dial has no * or # positions.

2400 telephones

2402 Telephone

The 2402 Telephone is a low-cost, low function DCP telephone. The 2402 has a 2 x 24 display, and four groups of permanently-labeled feature buttons (Speaker, Mute, Volume, Hold, Conference, Transfer, Redial, Drop, Messages, a, b, Feature). The 2402 Telephone is a Class B device.

2410 telephone

The 2410 Telephone is a mid-range, two-wire, circuit-switched digital telephone. The 2410 has a display that is similar to the 4610, but is used in character mode (text only). A single set type will be able to handle both Eurofont and Katakana fonts. The 2410 is electrically similar to the 2420. The LCD can display a minimum of 29 characters, in five lines. There are six programmable call appearance/feature buttons (two pages of six buttons for a total of 12 programmable buttons). There are four applications buttons (Call Log, Speed Dial, Options, Label), a built-in speakerphone, fixed feature buttons, and downloadable firmware. The user interface of the 2410 is similar to the 2420. The telephone is a Class B device. The 2410 does not have an EU24 port or modular capability.

2420 telephone

The 2420 telephone is a digital telephone with an optional feature expansion module and downloadable call appearance/feature buttons information. The 2420 Digital Communications Protocol (DCP) telephone does not need paper labels. The button information appears on a screen on the telephone. The firmware for the 2420 can be upgraded over its DCP connection to the switch. A new firmware image first must be downloaded into switch memory from a local trivial file transfer protocol (TFTP) server.

Avaya recommends that you administer call appears only on the first 8 call appearance/feature buttons on the 2420 telephone.

2500-series telephones

The 2500-series telephones consist of single appearance analog telephones with conventional touch-tone dialing. You can allow 2500-series telephones users to access features by giving them the appropriate feature access codes. For more information about providing feature access codes to your users, see Changing system parameters on page 58.

4600-series IP telephones

The 4600-series telephones are DCP telephones that use Internet Protocol (IP) technology with Ethernet line interfaces. IP (Internet Protocol) telephones obtain their operational characteristics from your central telephone server rather than residing in the phone unit itself. Updates and new features are downloaded to your phone without intervention or the need for phone replacement.Firmware for these telephones can be downloaded from the internet. The 4600 series includes a 2-button set, 6-button sets, a 12-button set, and a 24-button set.

Each of the 4600-series telephones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

The 4604, 4612, and 4624 telephones do not have a standard Drop button, but you can assign a drop button to any feature button. The 4600-series display telephones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these telephones.

4601 IP telephone

The 4601 is a low-cost Avaya VoIP telephone. It is based on the 4602, but does not have an LCD display. The 4601 supports H.323 v2, except for automatic unnamed registration. It supports two call appearances, and has no speakerphone. The 4601 looks similar to the 4602, but has some changes to the buttons (no SPKR, no MUTE), LEDs, and bezel. The 4601 is available in both a gray and white version. The power options are the same as the 4602; Power over Ethernet 802.3af. The telephone stand assembly is identical to that of the 4602. The 4601 Telephone is a Class B device.

Note:

When adding a new 4601 IP telephone, you must use the 4601+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4601, you receive a warning message, stating that you should upgrade to the 4601+ station type in order to access the Automatic Callback feature.

4602 IP telephone

The 4602 IP Telephone is an entry-level telephone with two programmable call appearance/ feature keys, ten fixed feature buttons, and display. The 4602SW offers the same functionality plus an integrated two-port Ethernet switch. Both the 4602 and 4602SW can run either the H.323 protocol, for integration with traditional Avaya IP Telephony Servers and Gateways, or the SIP Enablement Services (SES) protocol, for support of SES Communications Servers such as the Avaya Converged Communications Server.

Note:

When adding a new 4602 IP telephone, you must use the 4602+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4602, you receive a warning message, stating that you should upgrade to the 4602+ station type in order to access the Automatic Callback feature.

4602SW IP telephone

The 4602 and 4602SW are identical telephones from the point of view of user interface, capabilities, administration, etc. The only differences between the sets are due to differing electrical design - the 4602SW has an Ethernet switch, while the 4602 has an internal shared repeater.

4606 IP telephones

The 4606 IP telephone is a multi-appearance digital telephone with six call appearance/feature buttons: a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

4610SW IP telephone

The 4610SW IP Telephone provides a medium screen graphic display, paperless button labels, call log, speed dial, 12 programmable feature keys, Web browser, and full duplex speakerphone. It also includes a two-port Ethernet switch. The 4610SW supports unicode with R2.1 firmware. The 4610SW is also a functional application platform, supporting 3rd party applications that can push content, such as emergency alerts, to the displays or audio path. Emergency alerts and other applications are now supported.

4612 IP telephone

The 4612 IP telephone is a multi-appearance digital telephone with 12 call appearance/feature buttons. With the 4612 IP telephone the end-user can access 12 features with the softkeys and display control buttons.

4620SIP telephone

The 4620SIP IP Telephone provides access to the World Wide Web while offering the latest features and applications. The large display area allows up to 12 application-specific buttons to be presented and labeled at one time. Additionally, 12 Line/Feature buttons, 4 softkeys, and other fixed buttons provide access to powerful capabilities such as:

- local telephone and call server-based features
- speed dialing
- a Call Log
- a Wireless Markup Language (WML) browser

4620SIPCC telephone

The 4620SIPCC telephone is a new multiline SIP deskphone for Expert Agent Selection (EAS) agents that works with Avaya Communication Manager call center systems. The 4620SIPCC includes a Phone screen to view and manage calls, a Contacts list, a Call Log, feature buttons to change agent work mode and status, and a menu of options and settings to customize the telephone.

The 4620SIPCC also has softkeys that can be used to select an option or action that is displayed on the Phone screen.

4620SW IP telephone

Effective December 5, 2005, Avaya will no longer make 4620 IP telephones commercially available. The 4621SW IP telephone is an appropriate replacement. The 4621SW IP telephone, generally available since May 2005, offers the same functionality as the 4620 and adds a backlit display. For more information, see 4621SW IP telephone.

4621SW IP telephone

The 4621SW IP Telephone is cost effective and provides a large screen grayscale graphic display, paperless button labels, call log, speed dial, 24 programmable feature keys, Web browser, and full duplex speakerphone. The 4621SW also supports Unicode, IRDA, and is the only IP telephone to support the EU24 (expansion module). It also includes a 2 port Ethernet switch. The 4621SW is also a functional application platform, supporting 3rd party applications that can push content, such as emergency alerts, to the displays or audio path. Emergency alerts and other applications are now supported.

4622SW IP telephone

The 4622SW IP telephone is similar to the 4620, but with a backlit grayscale display. The 4622SW also supports Unicode. The 4622SW does not provide support for the IR interface, nor for a handset or a physical switch-hook. The 4622SW does, however, provide support for a one-way speakerphone, as well as two headset jacks, an advantage in a call center environment.

4624 IP telephone

The 4624 IP telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 4624 telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

4625SW IP telephone

The 4625SW IP telephone is similar to the 4620, but with a backlit ¼ VGA 256 color display and a slightly different stand. The 4625SW does not support the IR interface or multibyte languages. The 4625SW otherwise supports all the applications and options supported on the 4620.

4690 IP telephone

The Avaya 4690 IP Speakerphone provides the convenience and productivity benefits inherent in a purpose-built hands-free conference phone. It also delivers the extensive set of Avaya Communication Manager features directly to the conference room. It offers many of the same features as other Avaya speakerphones (360 degree coverage, two optional extended microphones for expanded coverage, full-duplex operation) and adds to them some additional capabilities. These include downloadable software upgrades and simplified wiring to IP network via Ethernet LAN connectivity.

6200-series telephones

6210 telephone

The 6210 telephones are single-line analog models. They have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6218 telephone

The 6218 telephones are single-line analog models. They have 10 programmable dialing buttons. These telephones also have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6220 telephone

The 6220 telephones are single-line analog models. These telephones also have fixed Flash, Redial, Hold, Mute, and Speakerphone (**Spkr**) feature buttons and a message waiting light. They also have handset volume control, ringer volume control, timed switch-hook disconnect, 10 programmable dialing buttons, repertory keylock, set personalized ring, and system hold.

Figure 220: The 6210 telephone

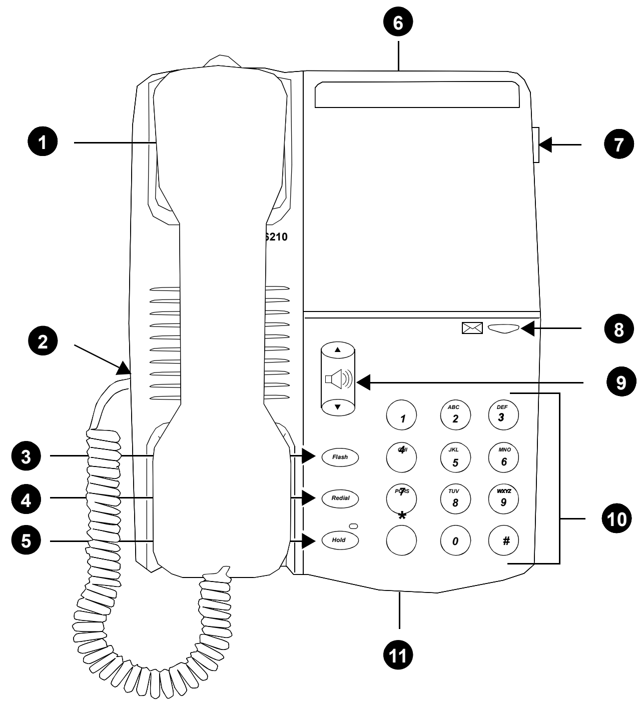


Figure notes:

1. **Handset**
2. **Handset cord jack**
3. **Flash button**
4. **Redial button**
5. **Hold button**
6. **DATA jack**
7. **Ringer volume control**
8. **Message light**
9. **Handset volume control**
10. **Dial pad**
11. **LINE jack (on bottom of telephone)**

Figure 221: The 6218 telephone

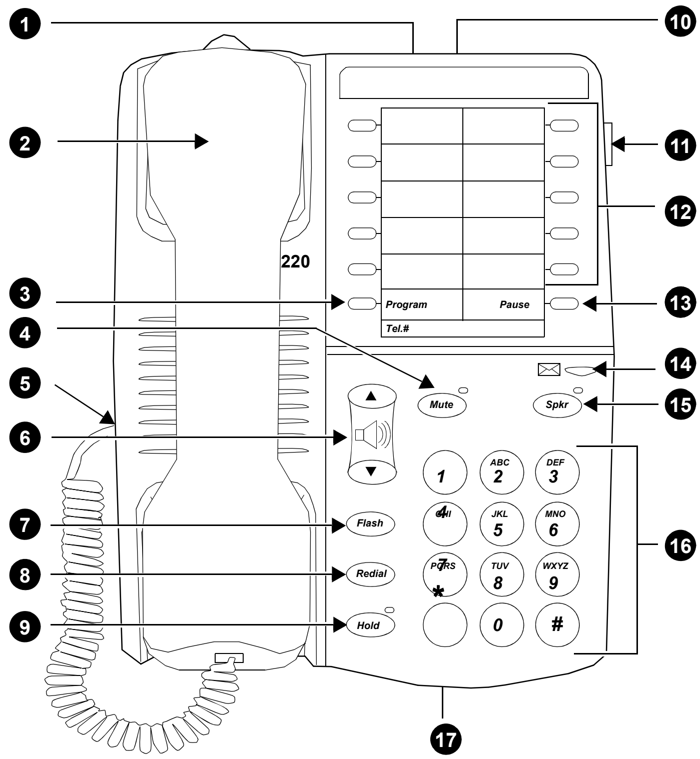


Figure notes:

1. Handset parking tab
2. Handset
3. Program button
4. Handset cord jack
5. Flash button
6. Redial button
7. Hold button
8. Data jack
9. Personalized ring
10. Ringer volume control
11. 10 programmable dialing buttons
12. Pause button
13. Message light
14. Handset/speakerphone volume control
15. Dial pad
16. LINE jack (on bottom of telephone)

Figure 222: The 6220 telephone

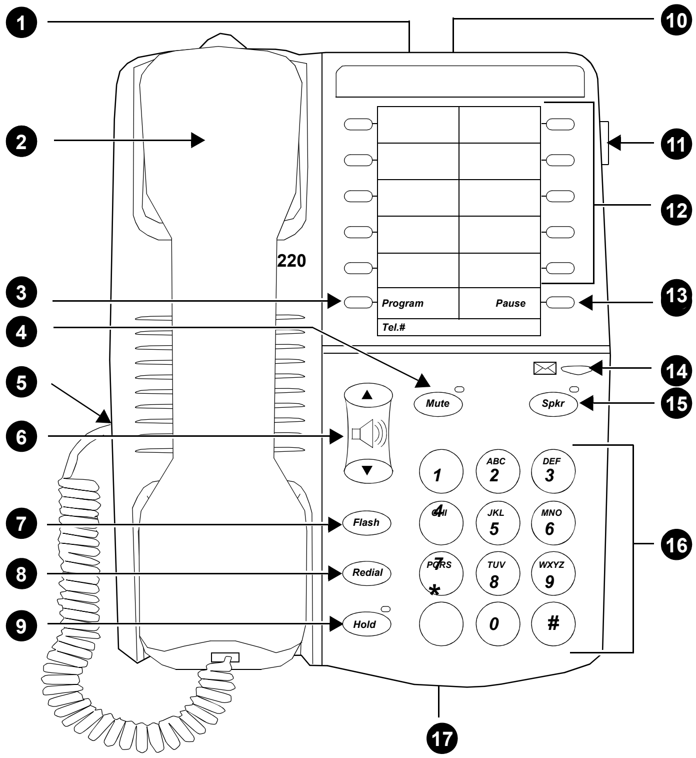


Figure notes:

1. DATA jack
2. Handset
3. Program button
4. Mute button
5. Handset cord jack
6. Handset/speakerphone volume control
7. Flash button
8. Redial button
9. Hold button
10. Personalized ring
11. Ringer volume control
12. 10 programmable dialing buttons
13. Pause button
14. Message light
15. Speakerphone button
16. Dial pad
17. LINE jack (on bottom of telephone)

6400-series telephones

The 6400-series telephones are DCP 2-wire telephones that work with Avaya Communication Manager. The last two digits of the 6400-series model number identify the number of call appearances (2-lamp buttons) for that model. For example, the 6424D has 24 call appearances. The 6400 series includes two single-line sets (6402 and 6402D), 8-button sets, a 16-button set, a 24-button set, and a 24-button expansion module (for the 6416D+ and 6424D+ telephones).

Each of the 6400-series telephones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

These telephones do not have a standard Drop button, but you can assign a drop button to any feature button. The 6400-series display telephones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these telephones.

6402/6402D telephones

The 6402 is a single-line telephone with six fixed feature buttons, a listen-only Speaker button, a Feature button that allows you to use the dial pad keys for up to 12 features assigned by the system manager, a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

The 6402D is the same as a 6402, but also has a 2-line by 16-character LCD display.

The 6402 and 6402D can be used in a 2-wire environment only.

6408/6408D telephones

The 6408 is a multi-appearance digital telephone with eight call appearance/feature buttons.

The 6408 telephone is available in the following four models:

- 6408 telephone—includes a 1-way, listen-only speaker, and no display
- 6408+ telephone—includes a 2-way speakerphone, and no display
- 6408D telephone—includes a 1-way, listen-only speaker, and a 2-line by 24-character display
- 6408D+ telephone—includes a 2-way speakerphone, and a 2-line by 24-character display.

With the 6408D and 6408D+ telephones, the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6408, 6408+, 6408D, and 6408D+ telephones can work only in 2-wire environments.

6416D+ telephone

The 6416D+ telephone is a multi-appearance digital telephone with 16 call appearance/feature buttons.

With the 6416D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

Note:

You can connect an XM24 expansion module to the 6416D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

6416D+M telephone

The 6416D+M telephone is similar to the 6416D, but with modular capabilities. You can install a module in the telephone’s desktop stand for increased set functionality. Note that these modules can only be used in the desktop position; they cannot be used if the telephone is wall-mounted.

6424D+ telephone

The 6424D+ telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 6424D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6424D+ telephone can work in both 4-wire and 2-wire environments.

Note:

You can connect an XM24 expansion module to a 6424D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

6424D+M telephone

The 6424D+M telephone is similar to the 6424D, but with modular capabilities. You can install a module in the telephone's desktop stand for increased set functionality. Note that these modules can only be used in the desktop position; they cannot be used if the telephone is wall-mounted.

Figure 223: 6402D+ telephone

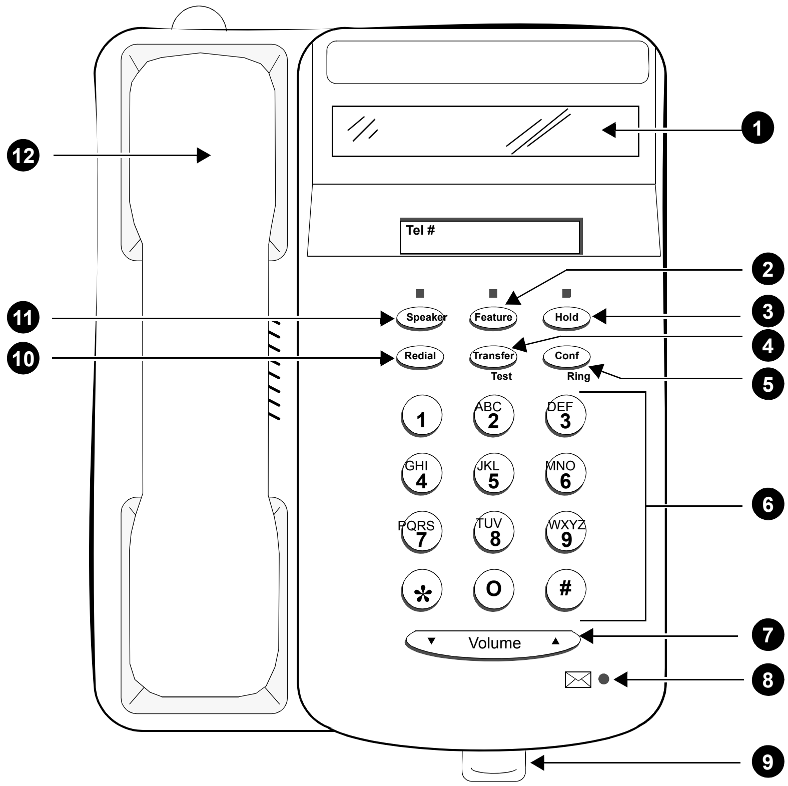


Figure notes:

1. Display
2. Feature button
3. Hold button
4. Transfer/Test button
5. Conf/Ring button
6. Dial pad
7. Volume control button
8. Message light
9. Tray handle (includes reference cards)
10. Redial button
11. Speaker button
12. Handset

Figure 224: 6408D telephone

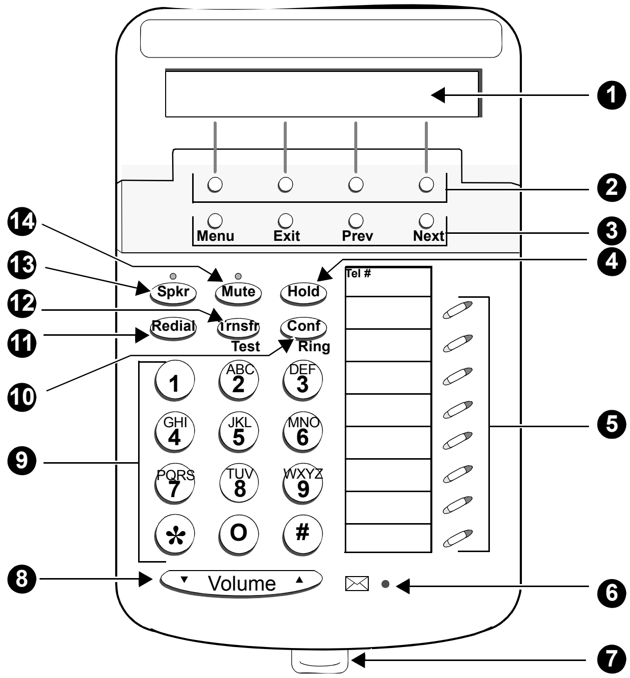


Figure notes:

1. Display
2. Softkeys
3. Display control buttons
4. Hold
5. Call appearance/feature buttons
6. Message light
7. Tray handle (includes reference cards)
8. Volume control button
9. Dial pad
10. Conf/Ring button
11. Redial button
12. Transfer/Test button
13. Speaker button
14. Mute button

7100-series telephones

7101A telephone

The 7101A telephone is a single-line analog model that is equipped with a Message Waiting light and a handy Recall button for activating the system's special features. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7102A telephone

The 7102A (7102A01A) and 7102+ (7102A01B), called the 7102 Plus. The front of the two sets is exactly the same in appearance. The only difference is that the 7102A01B is equipped with an adjunct jack. This jack allows speakerphone/headset capability.

7103A telephone

The 7103A fixed feature telephone is a single-line analog model which has been discontinued. The feature buttons on this telephone must be programmed by the system manager. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7104A telephone

The 7104A telephone is a single-line analog model that is equipped with a display that is used to display stored numbers. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7300-series telephones

7302H MERLIN telephone

The 7302H is a 5-button telephone that can be desk or wall mounted. This set can no longer be ordered.

Administer 7302H telephones (5-Button) as a 7303S model.

7303H MERLIN telephone

The 7303H is a 10-button telephone that can be desk or wall mounted.

Administer 7303H telephones (10-Button) as a 7305S model. You can administer only the first 12 feature function buttons. Of these 12 buttons, 8 have two lamps and 4 have no lamps. However, the system treats the 8 double-lamp buttons as though they have a single (green) lamp. Administer the 4 buttons (with no lamps) with features that do not require status indications.

7303S telephone

The 7303S is a multi-appearance hybrid telephone which provides access to 10 line appearances or selected programmable features. The 7303S telephone is also equipped with six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

7305H MERLIN telephone

The 7305H series is a 34-button telephone. This telephone can be equipped with or without different features such as built-in speakerphone or display. Some versions of the 34-button series can be ordered using PEC code 3162 and the appropriate suffix.

Administer 7305H telephones (34-Button) as a 7305S model. The system treats the telephone's 24 feature function buttons (two lamps each) as single (green) lamp function buttons.

7305S telephone

The 7305S telephone is a multi-appearance hybrid telephone which provides access to 10 line appearances. The 10 line appearance buttons can also be used as programmable feature buttons. The 7305S telephone is also equipped with 24 programmable feature buttons and six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

Figure 225: 7303S telephone

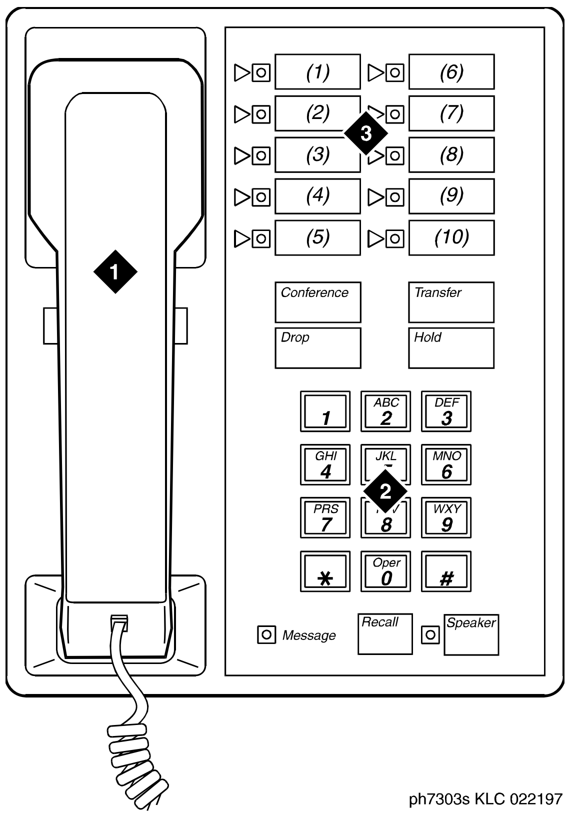


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Figure 226: 7305S telephone

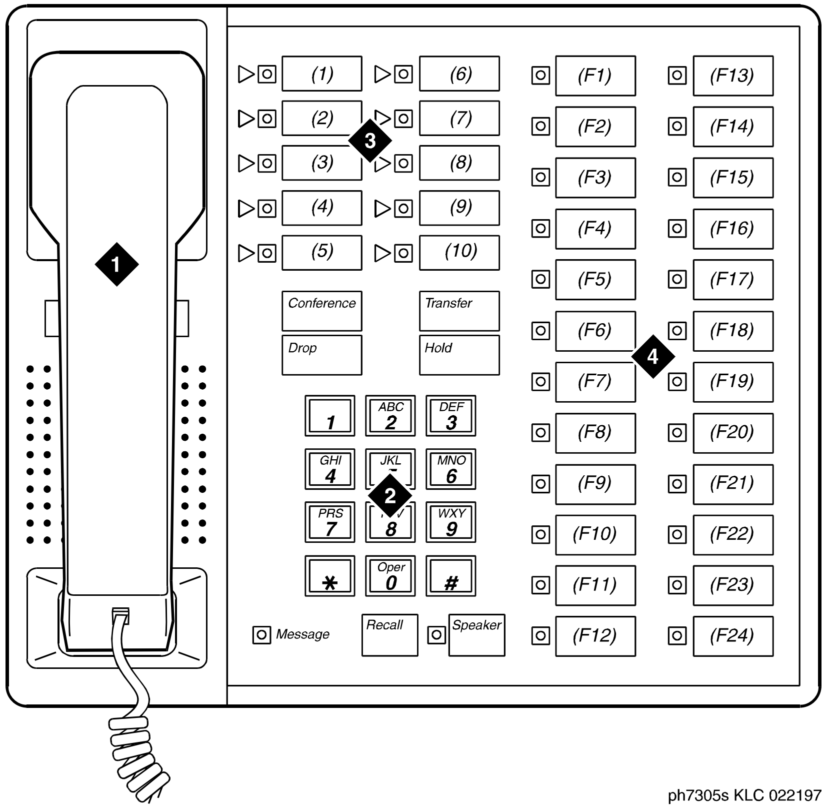


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

731x-series hybrid telephones

You should note that the following restrictions apply to administering hybrid telephones in the system:

- Hybrid telephones equipped with displays cannot be used as Automatic Call Distribution (ACD) agents and do not allow your end-users to access the Directory.
- The following fixed feature buttons do not operate on Hybrid telephones: STOP, PAUSE, RECALL, MESSAGE, HFAI, and HFAI/MIC. If you want users to have Hands Free Automatic Answer on Intercom (HFAI), assign Internal Automatic Answer (IAA) to a lamp button.
- These telephones support Leave Word Calling (LWC), but users cannot retrieve messages with the display.

Figure 227: 7313H telephone (BIS 10)

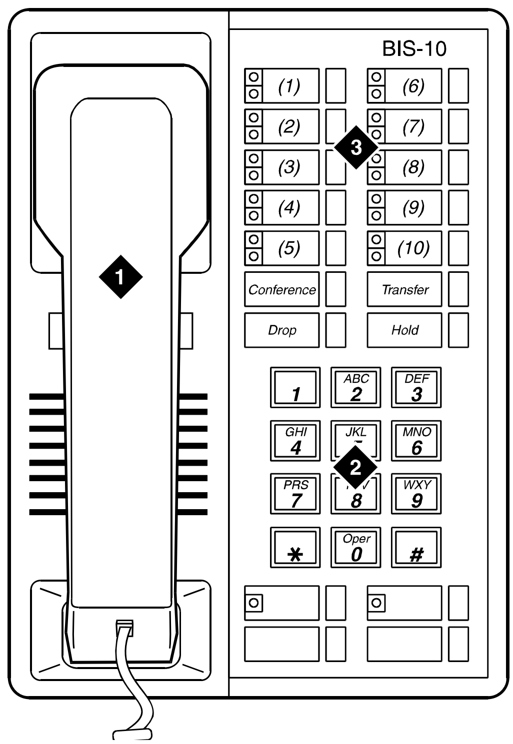


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Figure 228: 7314H telephone (BIS 22)

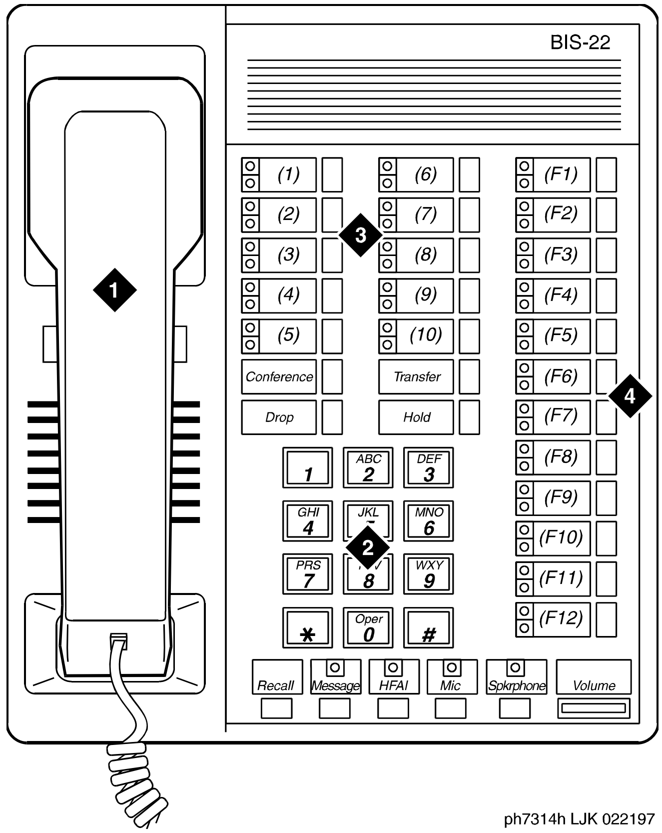


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **12 feature buttons**

Figure 229: 7315H telephone (BIS 22D)

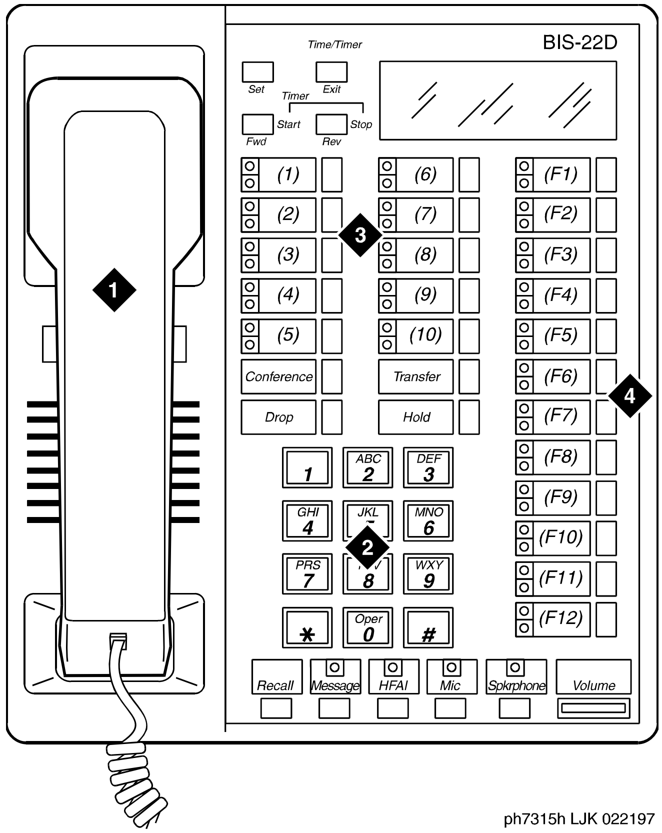


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **12 feature buttons**

Figure 230: 7316H telephone (BIS 34)

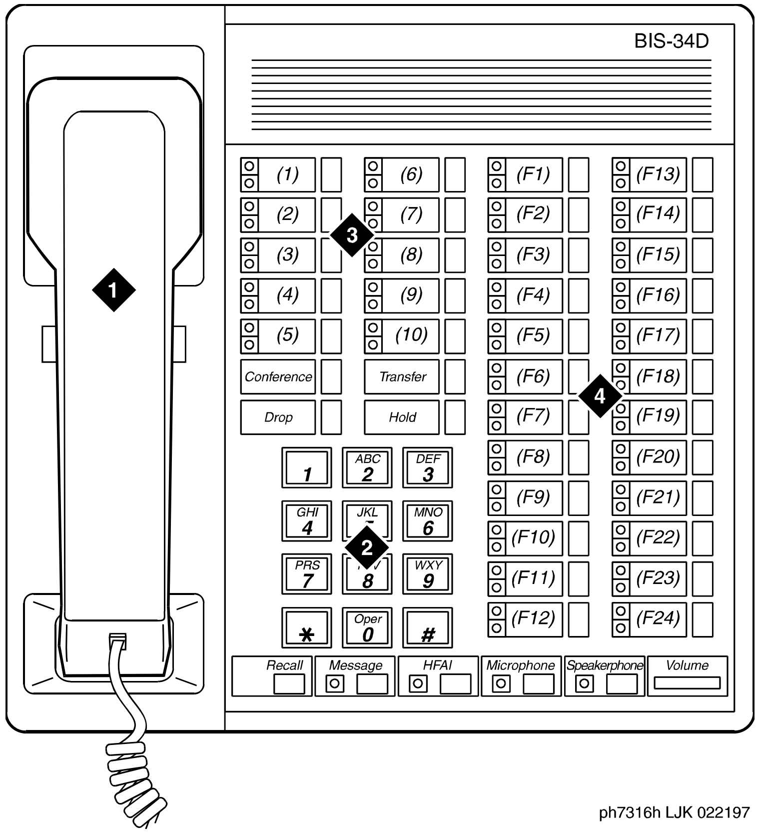


Figure notes:

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 231: 7317H telephone (BIS 34D)

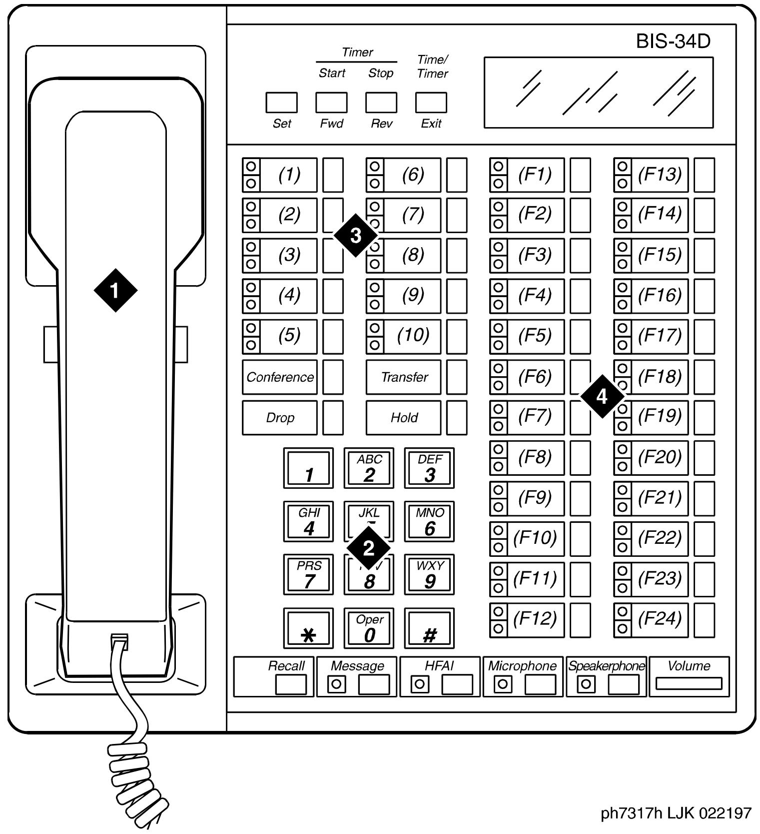


Figure notes:Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

7400-series telephones

7401D telephone

The 7401D (7401D01A) and the 7401+ (7401D02A) are both single appearance digital telephones which have no call appearance buttons or lights, but have two virtual call appearances. Depending on how the 7401D or the 7401+ telephone is administered, the second call appearance might be restricted to incoming priority calls and outgoing calls only.

7403D telephone

The 7403D is a multi-appearance digital telephone which has 10 buttons available for line appearances, one-touch feature access, or Speed Dialing. In addition, the 7403D telephone can be equipped with a Digital Terminal Data Module (DTDM) which attaches to the right side and allows the connection of a EIA RS-232C data terminal.

7404D telephone

The 7404D is a multi-appearance digital telephone which provides simultaneous voice/ asynchronous data transmission. All the transmission is done over the same two pairs of wire.

7405D telephone

The 7405D telephone is a multi-appearance digital telephone which allows features to be added as the user needs them. The Digital Display can be added to provide access to the Message Center. A Digital Terminal Data Module or 7400B can be added to enable the user of a 7405D telephone to transmit or receive data with an associated data terminal.

The basic 7405D provides 10 call appearance/feature buttons with lights that can be assigned to call appearances or system features. It has 24 programmable feature buttons and six fixed-feature buttons. The 7405D can also have a function key module which adds 24 feature buttons and a call coverage module (when no display module is used) which adds 20 call appearance/feature buttons.

7406 telephones

The 7406D telephone (7406D01A, 7406D02A, 7406D03A, and 7406D04A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable feature buttons with no lights, four programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button, and a green Message light.

The 7406BIS telephone (7406D05A and 7406D06A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable buttons with no lights, two programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button with a green light, a MUTE button with a red light, a SPEAKER VOLUME "arrow" button, and a red Message light.

The 7406+ telephone (7406D07A and 7406D08A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, three shiftable (2-level) programmable feature buttons with a green light, six shiftable (2-level) programmable feature buttons without lights, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SELECT button with a green light, a SPEAKER/RESET SPKR button with a green light, a MUTE button with a red light, a VOLUME "arrow" button, and a red Message light.

7407+ telephone

The 7407D, Enhanced 7407D, and 7407+ telephones are multi-appearance digital telephones which provide digital voice, display, and data capabilities (the latter with the 7400B+ Data Module).

There are three versions of the 7407D telephone:

- 7407+ (7407D02D) — offers 10 call appearance buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), nine feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven buttons for display features), 22 flexible feature buttons with no lights, a Message light, personalized ringing, a built-in speakerphone with a reset and listen-only option, and a built-in 2-line by 40-character display.
- The 7407D (the 7407D01B)—offers 10 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (CALCULATOR/SELECT RING, SPEAKERPHONE, and MICROPHONE), nine programmable feature buttons with lights (the two uppermost buttons can be used for voice or display features, the lower seven for display features), 11 dual-function buttons, 22 programmable feature-only buttons without lights, a Message light, a DISPLAY button that turns the display on and off, personalized ringing, a built-in speakerphone, a 2-line by 40-character liquid crystal display, and a built-in calculator. This set is AC powered.

- Enhanced 7407D (the 7407D02C)—offers 10 call appearance buttons, each with a red in-use light and green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (SELECT RING, SPEAKER, and MUTE), 22 programmable feature-only buttons without lights, nine programmable feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven for display features), a Message light, personalized ringing, a built-in speakerphone, a connection for an adjunct speakerphone or headset, a speakerphone with spokesman, and Mute option, and a 2-line by 40-character display.

7410D and 7410+ telephones

The 7410D (7410D01A) and 7410+ (7410D02A) are both multi-appearance digital telephones with 10 line appearances, four standard fixed feature buttons and a Select Ring button. The 7410D and 7410+ telephones also provide a Message light, Ringer Volume control, a Speakerphone/headset adapter jack. You can activate a Self-Test feature to test the lights and tone ringer on the telephone. In addition, the 7410+ provides a Speaker for listening-only functions.

7434D telephone

The 7434D is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. The 7434D telephone supports an adjunct display module or a call coverage module.

7444D telephone

The 7444 telephone is a multi-appearance digital telephone that offers 34 call appearance/ feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. It is powered from the switch.

Note:

The 7444 is powered by the switch, however, to use the display, you must connect an auxiliary power supply to the telephone.

Figure 232: 7401D telephone

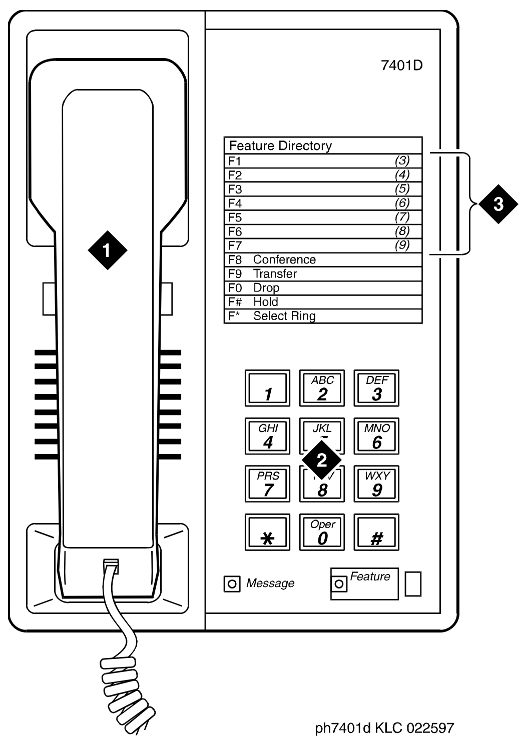


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Access codes card**

Figure 233: 7404D telephone

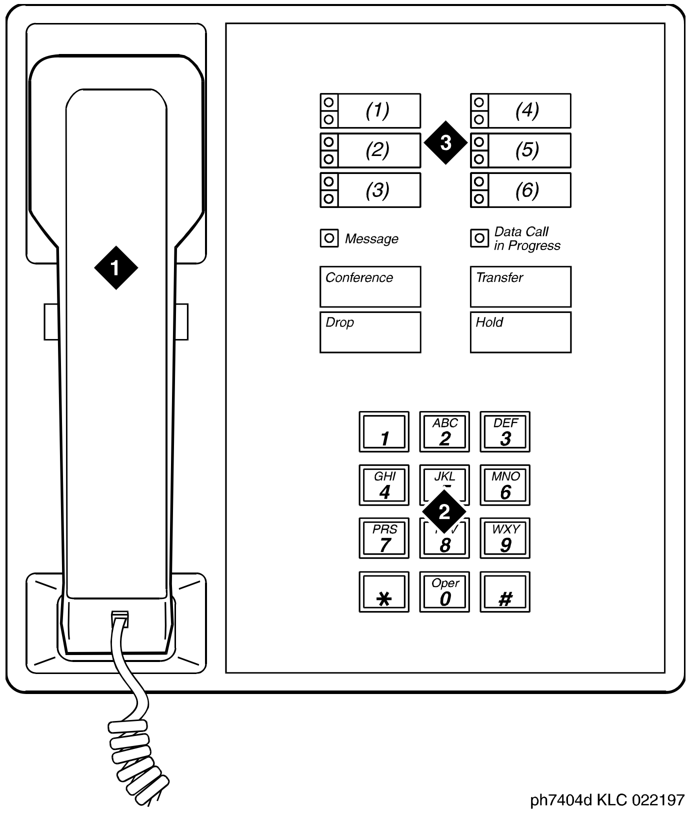


Figure notes:

1. Handset
2. Dial pad
3. 6 programmable buttons

Figure 234: 7405D telephone

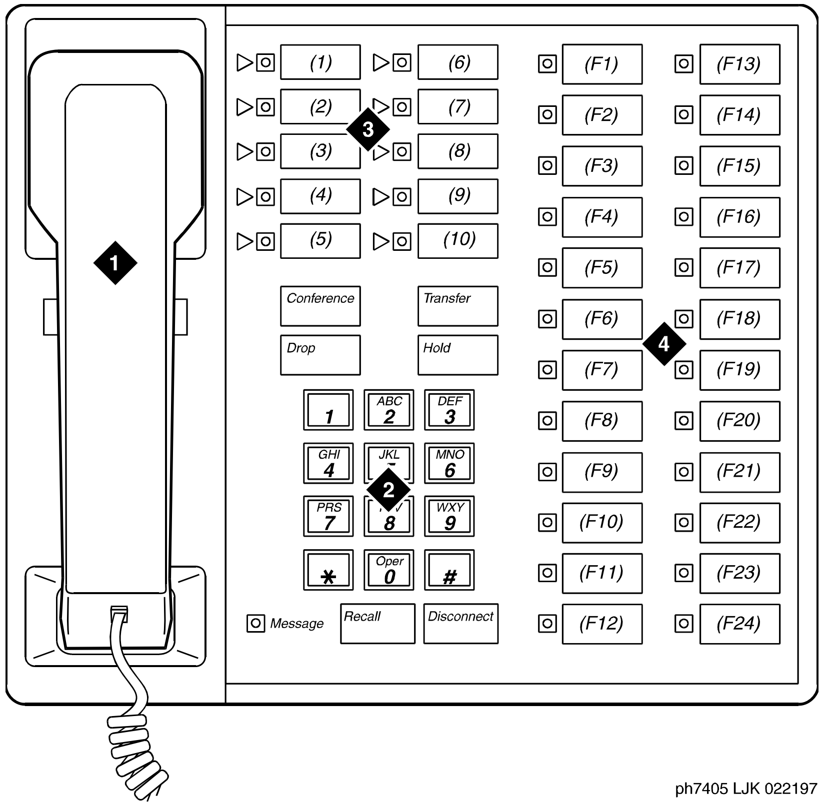


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

Figure 235: 7405D telephone with optional function key module

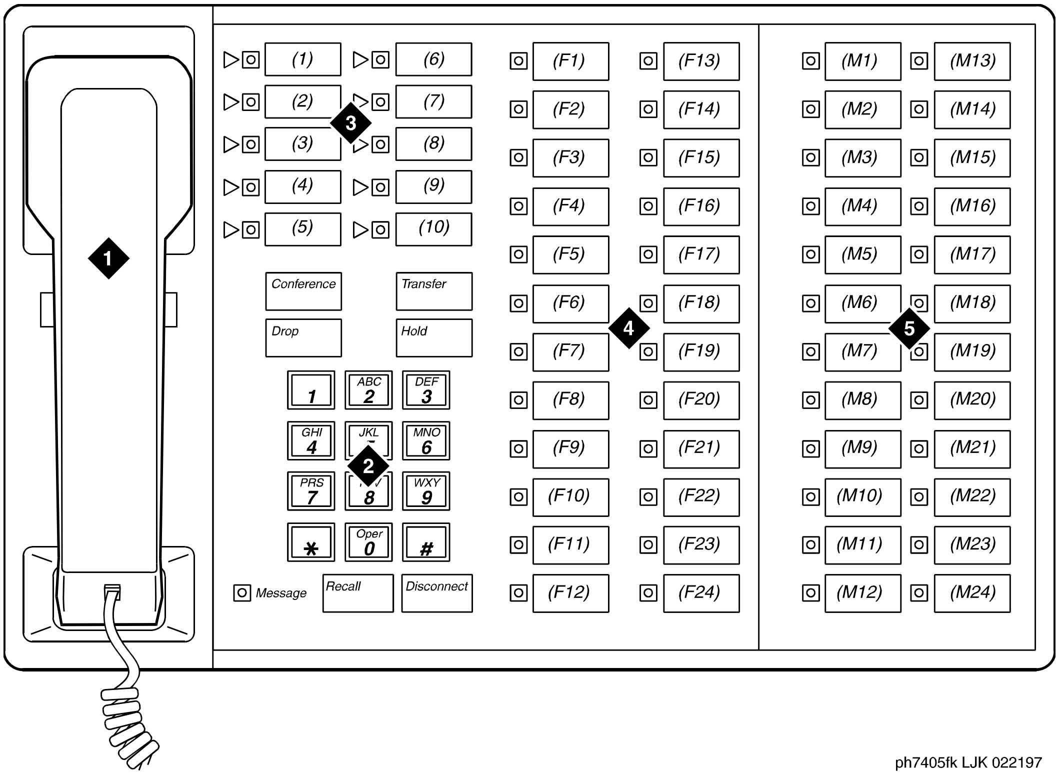


Figure notes:

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons
5. Function key module with 24 feature module buttons

Figure 236: 7405D telephone with optional digital display module

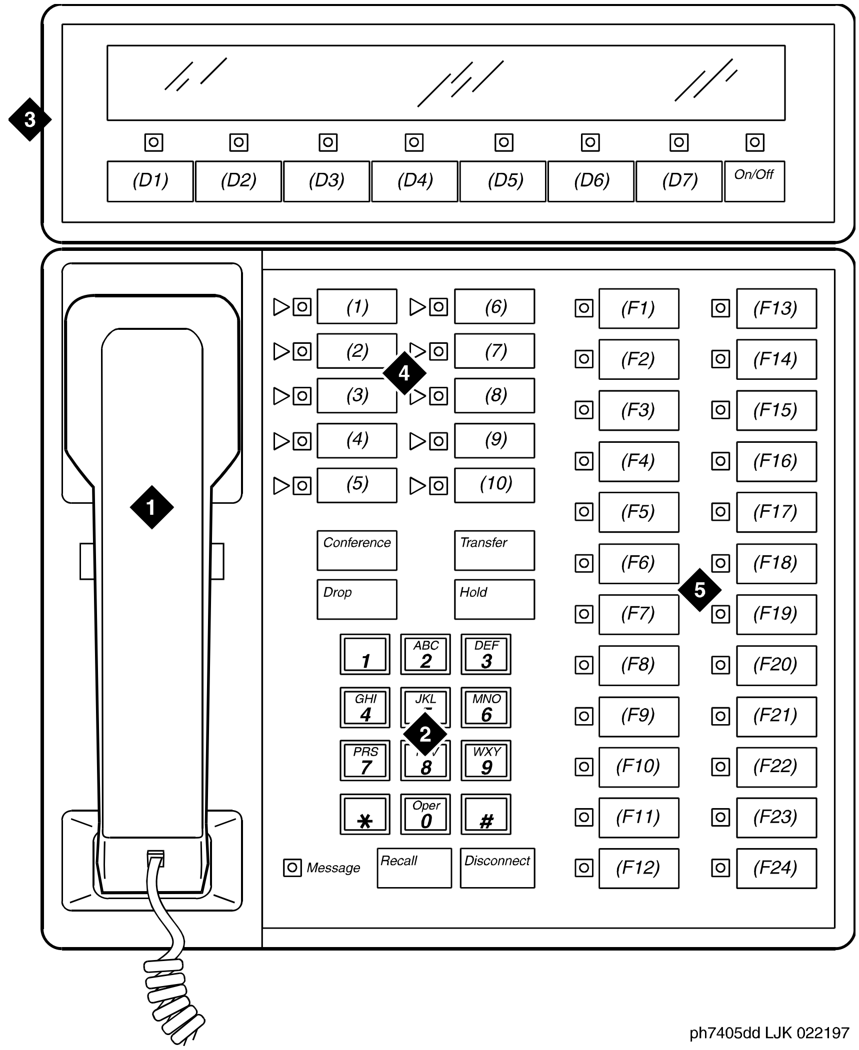


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Digital display module with 7 display buttons**
4. **10 programmable buttons**
5. **24 feature buttons**

Figure 237: 7405D telephone with optional call coverage module

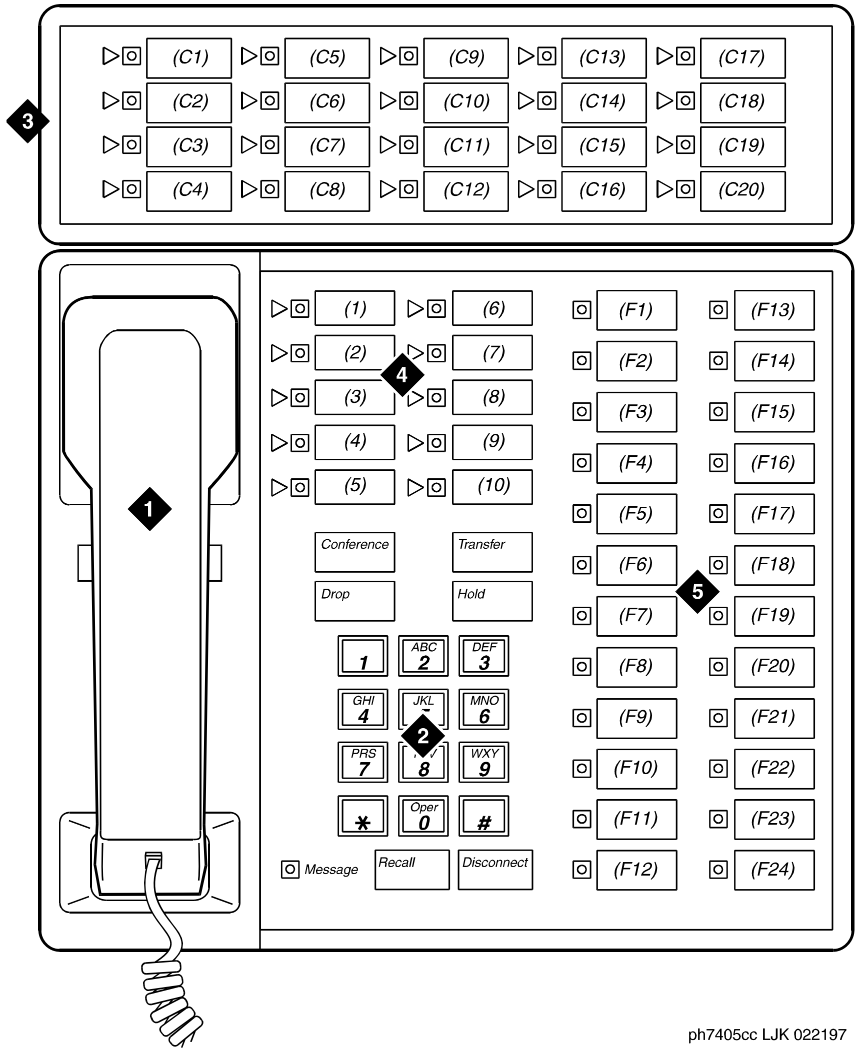


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Call coverage module with 20 coverage module buttons and status lamps**
4. **10 programmable buttons**
5. **24 feature buttons**

Figure 238: 7405D telephone with optional digital terminal data module

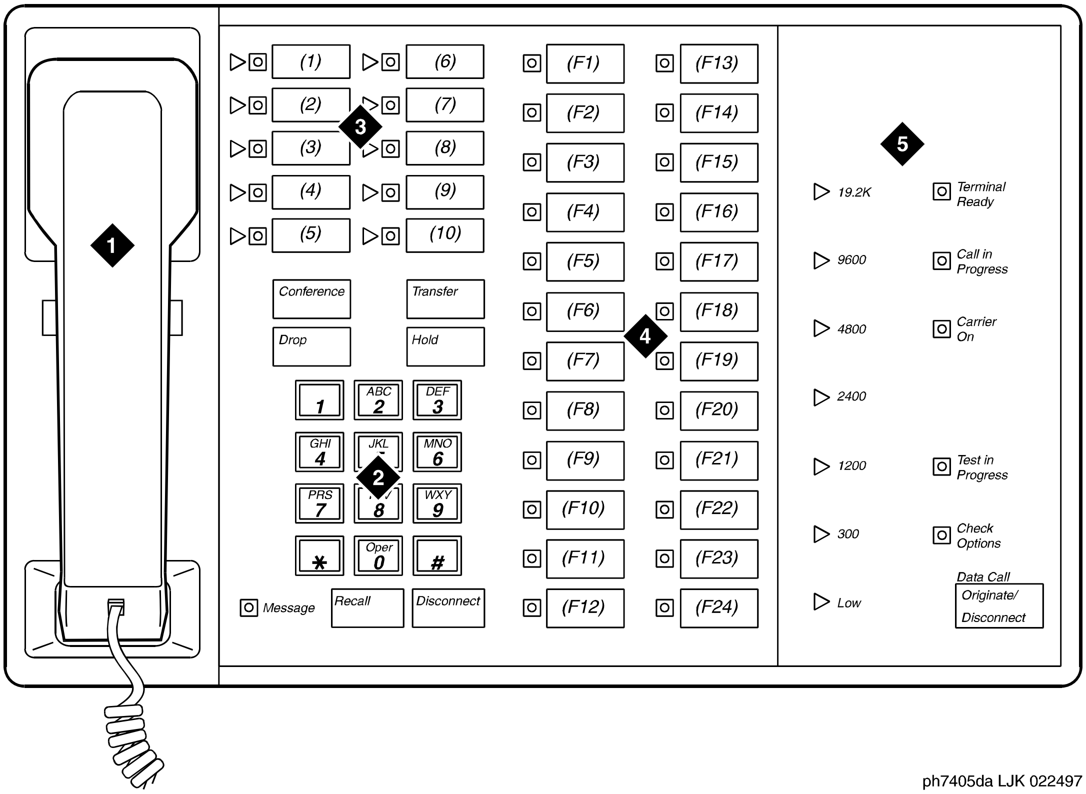


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**
5. **Digital terminal data module**

Figure 239: 7406D+ telephone

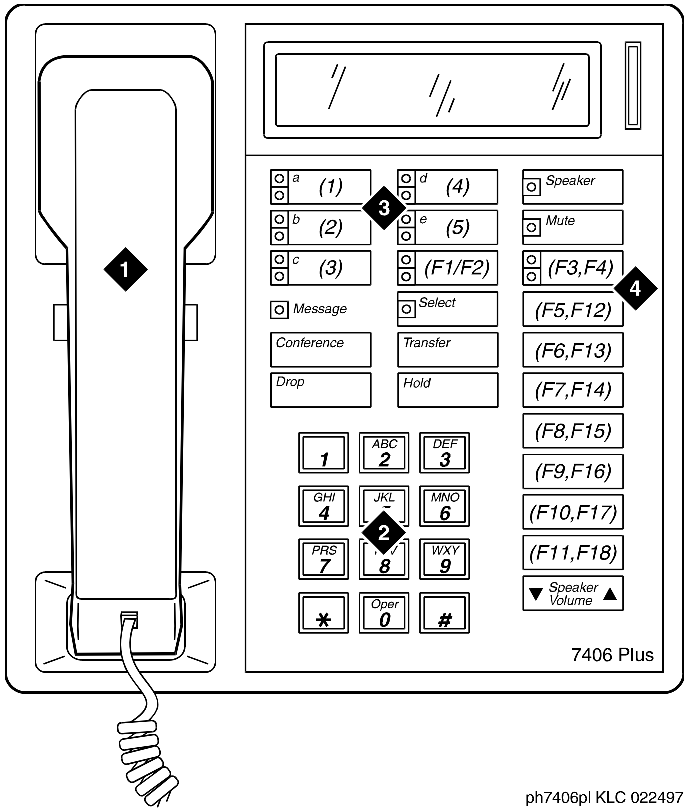


Figure notes:

1. **Handset**
2. **Dial pad**
3. **5 programmable buttons**
4. **18 feature buttons (feature buttons F2, F4, and F12 to F18 are enabled with the Shift key)**

Figure 240: 7406D telephone

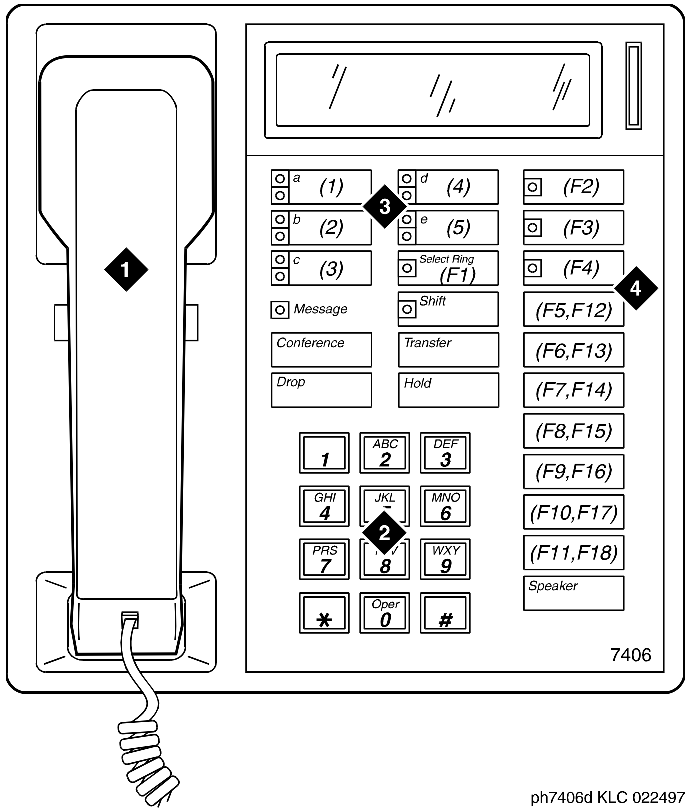


Figure notes:

1. **Handset**
2. **Dial pad**
3. **5 programmable buttons**
4. **18 feature buttons (feature buttons F12 to F18 are enabled with the Shift key)**

Figure 241: 7407D telephone

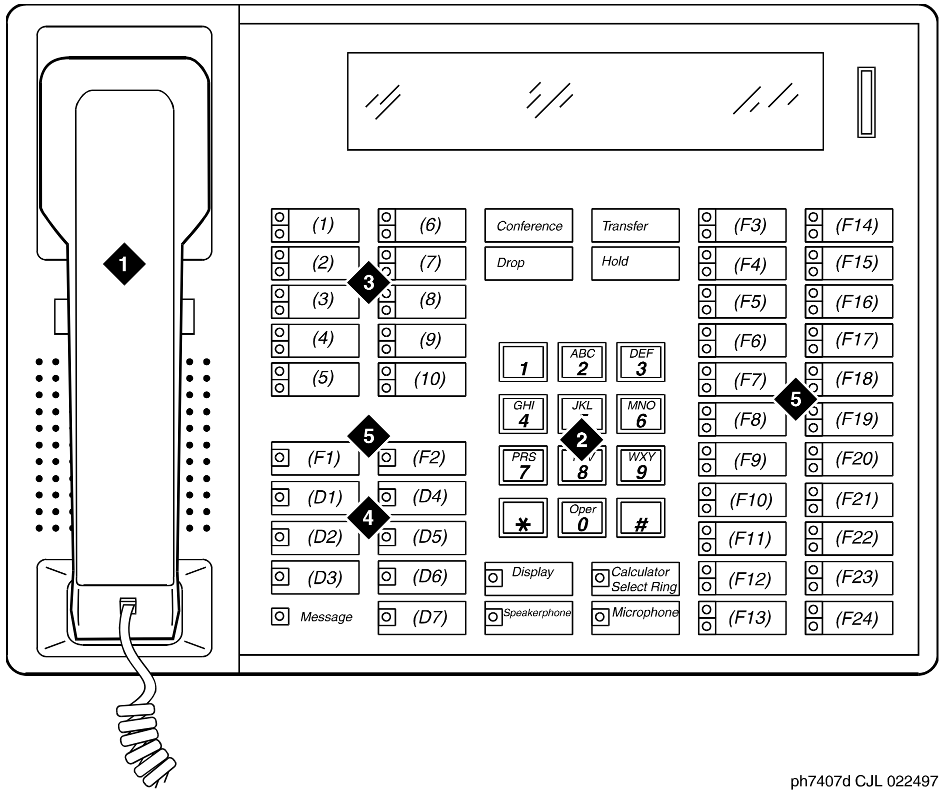


Figure notes:

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 7 display buttons
5. 24 feature buttons

Figure 242: 7434D telephone

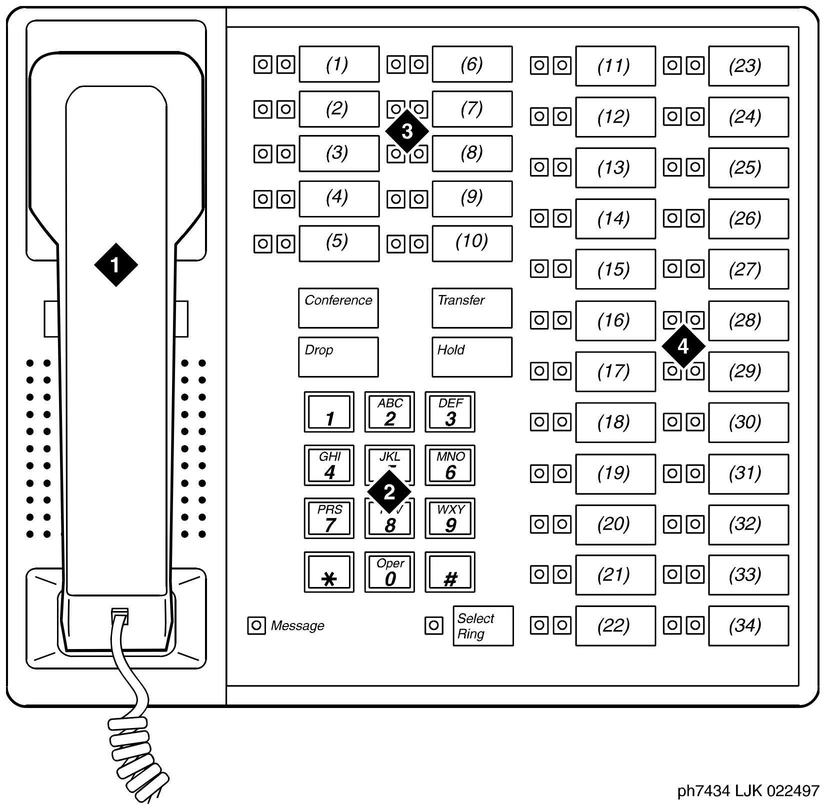


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons (11 to 34)**

Figure 243: 7434D telephone with optional call coverage module

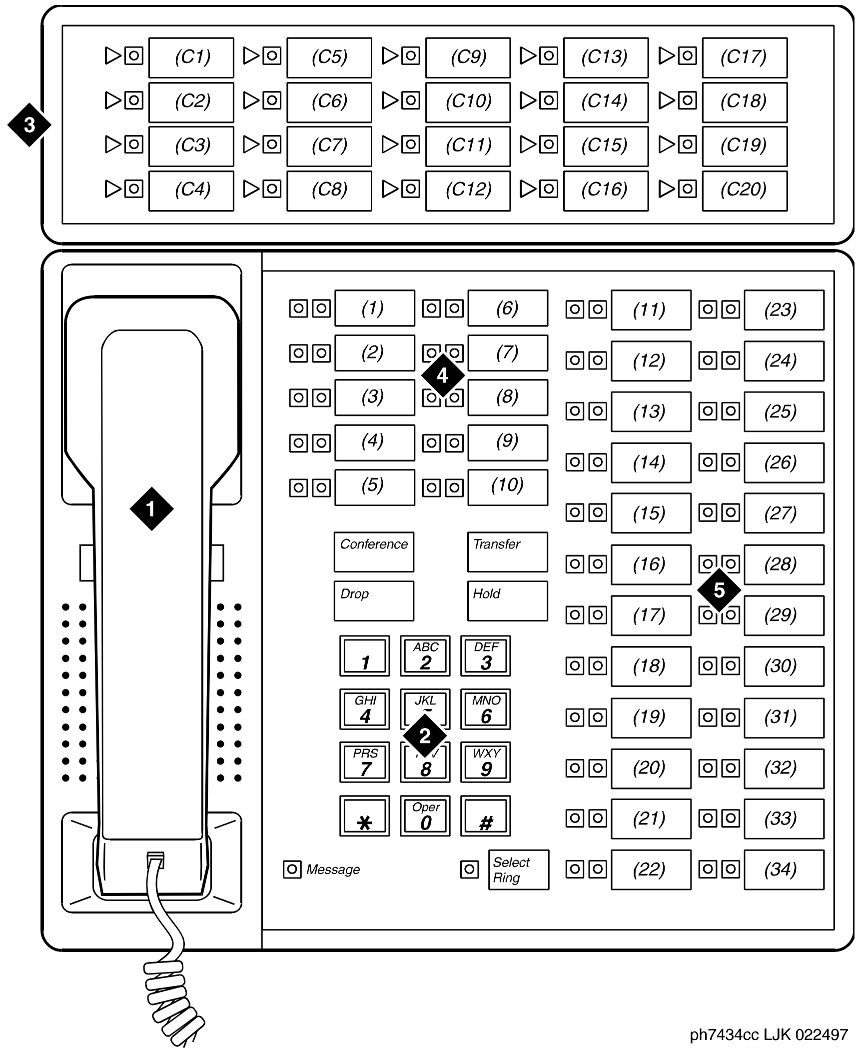


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Call coverage module with 20 coverage module buttons and status lamps**
4. **10 programmable buttons**
5. **24 feature buttons (11 to 34)**

Figure 244: 7434D telephone with optional digital display module

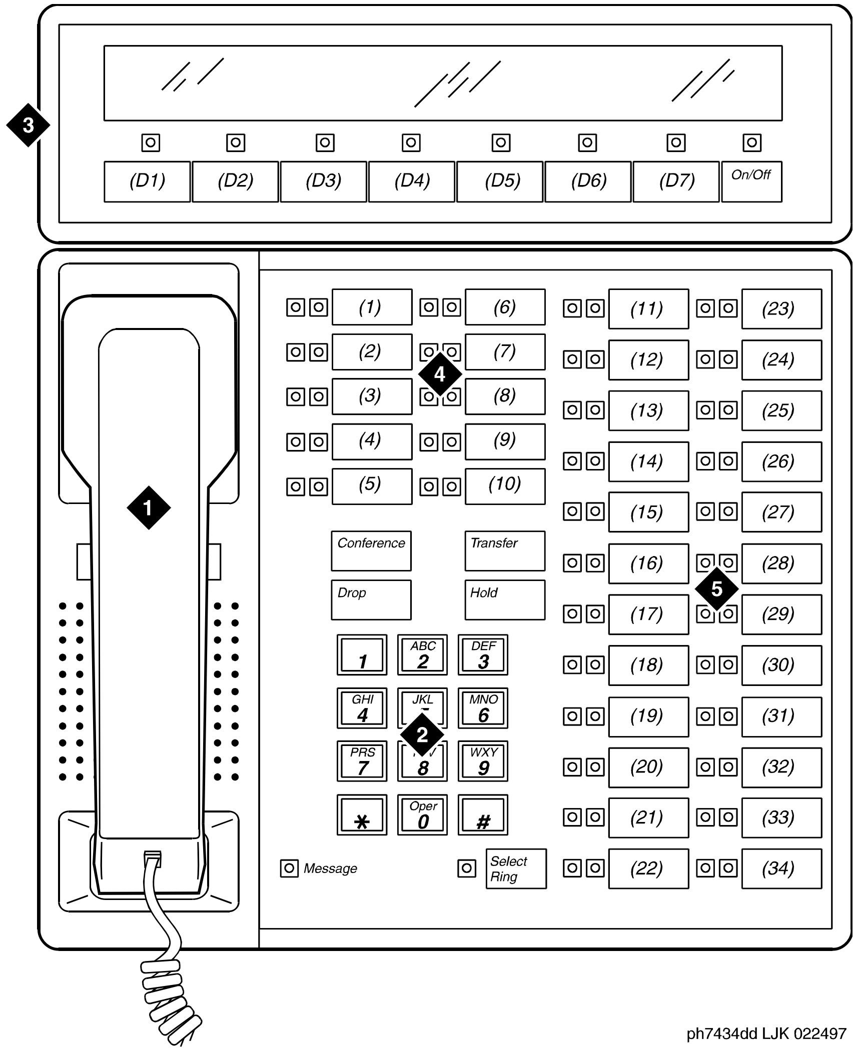


Figure notes:

1. **Handset**
2. **Dial pad**
3. **Digital display module with 7 display buttons**
4. **10 programmable buttons**
5. **24 feature buttons (11 to 34)**

ISDN telephones (7500s & 8500s)

The Integrated Services Digital Network (ISDN) telephones include both the 7500-series and the 8500-series telephones. Each of these telephones uses the ISDN communications through a 4-wire "T"-interface.

National ISDN (NI-1 and NI-2) BRI telephones

Telephones that comply with National ISDN (NI-1 and NI-2) BRI standards can serve as telephones on a Communication Manager system if they are programmed in Call Appearance Call Handling - Electronic Key Telephone Service (CACH-EKTS) mode. Examples of NI-BRI telephones that are supported are:

- Nortel 5317T
- L3 STE
- 6210 and 6220 Tone Commander

NI-BRI telephones with Communication Manager offer multiple call appearances, the **Conference**, **Transfer**, **Hold**, and **Drop** feature buttons, and a Message Waiting Indicator. NI-BRI telephone users must access all other features of Communication Manager using feature access codes. Additional feature buttons on an NI-BRI telephone can be assigned only as call appearances, and the number of call appearance buttons administered in Communication Manager must match the number of call appearances administered on the telephone. Bridged call appearances are not supported.

NI-BRI telephones also have access to all features of the Multi-level Precedence and Preemption (MLPP) capability if MLPP has been enabled.

7505D ISDN-BRI telephone

The 7505D telephone serves as a telephone if it is equipped with a Voice Only Module (VOM). As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7505 can be equipped with an optional Asynchronous Data Module (ADM) that provides the user with simultaneous voice and data capabilities. The 7505 equipped with the ADM offers the same voice capabilities as the 7505 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

7506D ISDN-BRI telephone

The 7506D serves as a telephone if it is equipped with a VOM. As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7506 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7506 equipped with the ADM offers the same voice capabilities as the 7506 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive asynchronous data through the digital network.

7507D ISDN-BRI telephone

The 7507D serves as a telephone if it is equipped with a VOM. As a telephone it offers programmable buttons, fixed feature buttons, Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7507 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7507 equipped with the ADM offers the same voice capabilities as the 7507 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

8503D ISDN-BRI telephones

The 8503T telephone offers: four standard fixed feature buttons, a Message light, three call appearance/flexible feature buttons, each with a red and green light, 12 programmable memory-dialing locations on the dial pad keys, a PROGRAM button for storing numbers at the memory-dialing locations and a MEMORY button for dialing these programmed numbers or codes, the Redial feature, the Mute feature.

8510T ISDN-BRI telephone

The 8510T voice/data telephone offers: 10 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. You can administer the softkey buttons. Four softkeys and display control buttons below the a 2-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

8520T ISDN-BRI telephones

The 8520T voice/data telephone offers the following: 20 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. Ten softkeys and four display control buttons located on either side of the 7-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

Figure 245: 8503D telephone

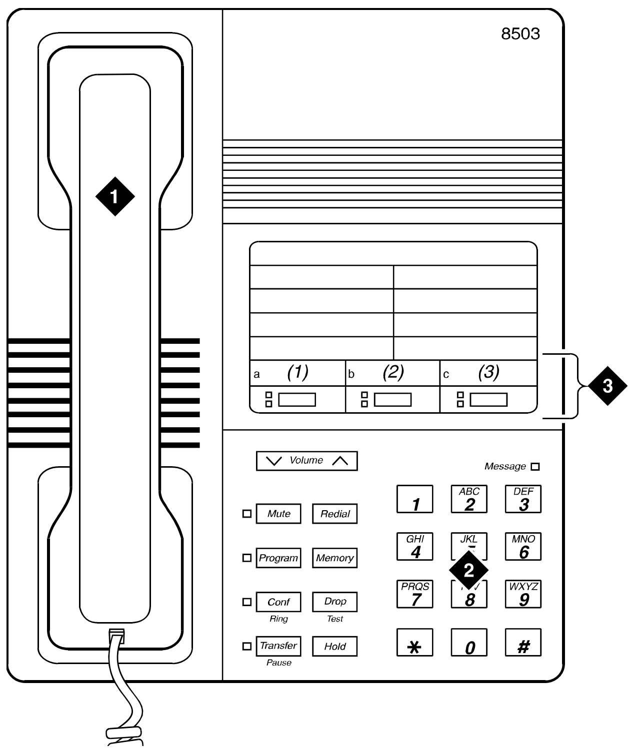
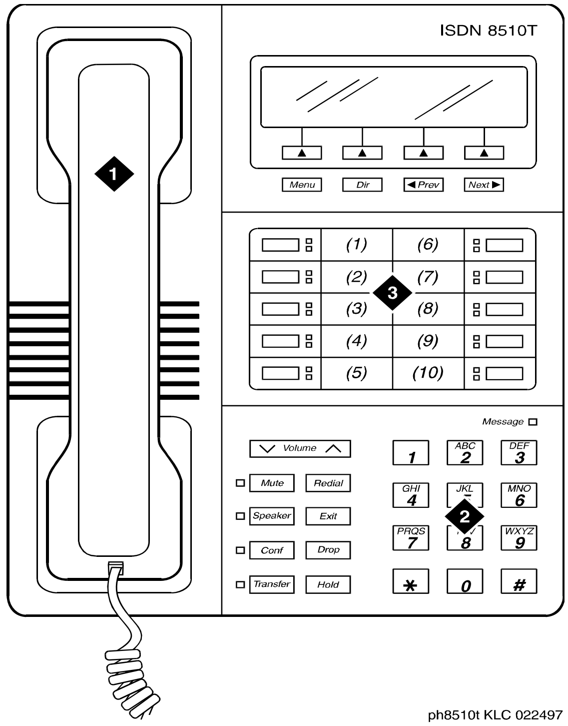


Figure notes:

1. **Handset**
2. **Dial pad**
3. **3 programmable buttons**

Figure 246: 8510T telephone



ph8510t KLC 022497

Figure notes:

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 247: 8520T telephone

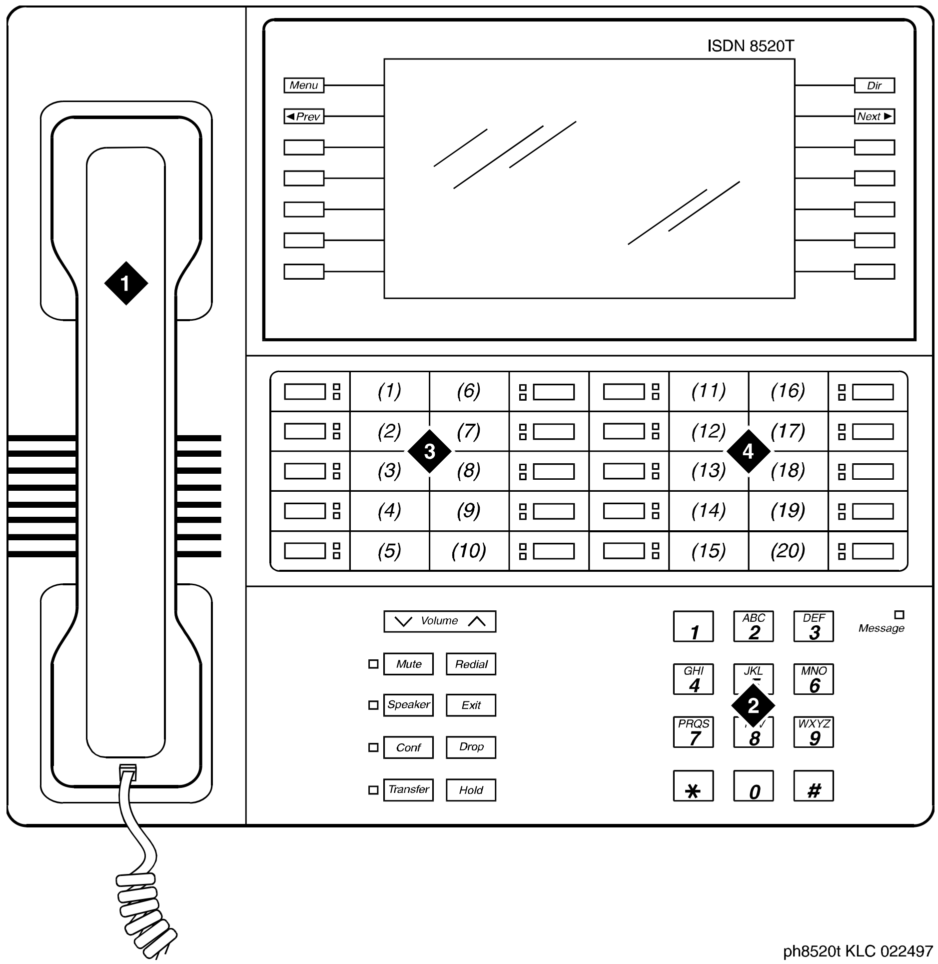


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **10 programmable buttons (11 to 20)**

Note:

The 8520T telephone supports 20 call appearances. The system maximum of 10 call appearance buttons still applies. You can administer the buttons that are not used as call appearance buttons as bridged appearances.

8110 telephones

The basic 8110 (8110A01A, 8110A01B, and 811A01C) and the modified 8110M (8110A01D) telephones are single-line analog telephones. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons with PROGRAM and PAUSE buttons, automatic redial, a flashing red Message light, and a Hold button. They also have built-in speakerphones with Mute capability and the Automatic Answer feature.

8400-series telephones

8403B telephones

The 8403 telephone is a multi-appearance digital telephone with three call appearance buttons, Conference, Transfer, Drop, and Hold buttons, a TEST button, a blue FEATURE button which allows you to access 12 system features assigned by the System Manager and to choose from among eight different ringing patterns, a MUTE button, a SPEAKER button which accesses a 1-way, listen-only speaker, a red Message light, and a Volume control button.

The 8403 can be used in either a 4-wire or 2-wire environment.

8405B telephone

There are four varieties of the 8405 telephone: the 8405B and 8405B+, the 8405D and 8405D+. All four varieties are multi-appearance digital telephones with five call appearance/feature buttons.The 8405 telephones also have four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button, a TEST button, and a Volume control button. The 8405D and 8405D+ allow you to administer 12 softkey feature buttons in addition to the call appearance and feature buttons.

The four 8405 variations have the following differences:

- The 8405B has a 1-way, listen-only speaker, with NO display.
- The 8405B+ has a 2-way speakerphone, without a display
- The 8405D has a 1-way, listen-only speaker and a 2-line by 24-character display.
- The 8405D+ has a 2-way speakerphone and a 2-line by 24-character display.

The 8405 telephones work in 4-wire or 2-wire environments.

8410B telephone

The 8410 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

- The 8410B is the basic set, without a display.
- The 8410D (8410D03A) has a built-in 2-line by 24-character display. Those users who have an 8410D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used *in addition to* the features on the call appearance/ feature buttons.

The 8410 telephone can work in both 4-wire and 2-wire environments.

8411B and 8411D telephones

The 8411 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a blue SHIFT button, a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

The rear of the 8411 telephone has two jacks: The Analog Adjunct jack can be used for connecting answering machines, fax machines, PC or laptop data/fax modem cards, data sets or modems, audio teleconferencing equipment, and TTY machines commonly used by the hearing impaired. The RS-232-D Jack can be used for connecting the telephone to a COM port on an IBM®-compatible personal computer on which you can load PassageWay Solution software.

There are two varieties of the 8411 telephone: the 8411B (8411D02A) is the basic set, without a display; the 8411D (8411D01A) has a built-in 2-line by 24-character display. Those users who have an 8411D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features on the call appearance/ feature buttons.

The 8411 telephone can work in both 4-wire and 2-wire environments.

8434D telephone

The basic 8434 (8434D01A) and the enhanced 8434DX (8434D02A) telephones are multi-appearance digital telephones which offer 34 call appearance/feature buttons, each with a red light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which accesses either a 2-way speakerphone or a 1-way listen-only speaker, a TEST button, a SHIFT button (some 8434DX telephones will have a RING button instead), a red Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character VFD display. The 8434 and 8434DX also have five softkeys and four display control buttons which allow the user to access 15 features. These softkey features can be used *in addition to* the features on the call appearance/feature buttons.

The 8434 and 8434DX telephones can be used in both a 4-wire and a 2-wire environment.

Note:

In order to use the display on the 8434 or 8434DX telephone and to use an 801A expansion module connected to the 8434DX, you must connect an auxiliary power supply to the telephone.

You can connect an 801A Expansion Module to the 8434DX telephone to provide 24 additional call appearance/feature buttons.

Figure 248: 8403B telephone

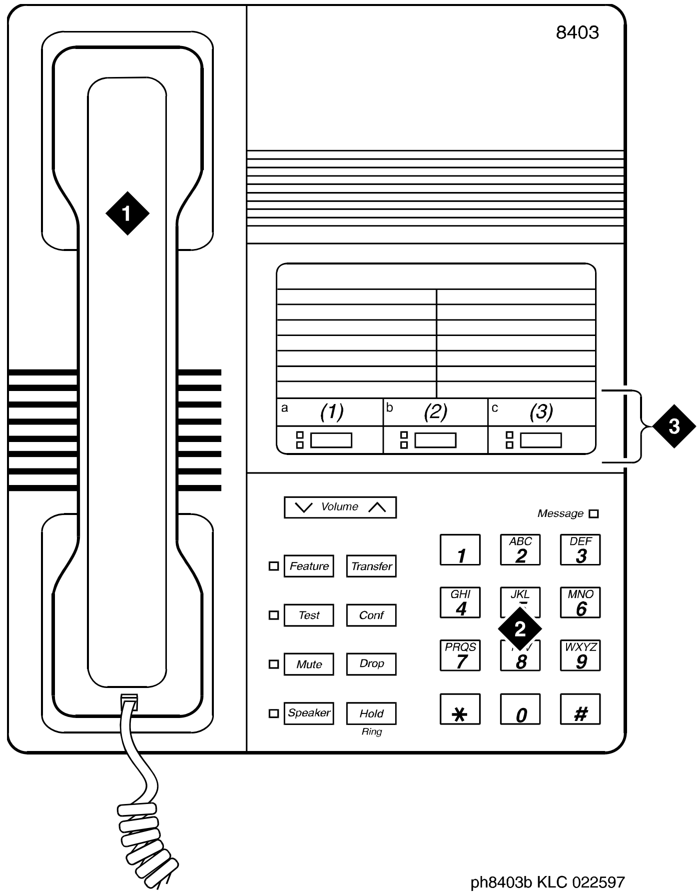


Figure notes:

1. Handset
2. Dial pad
3. 3 programmable buttons

Figure 249: 8410B telephone

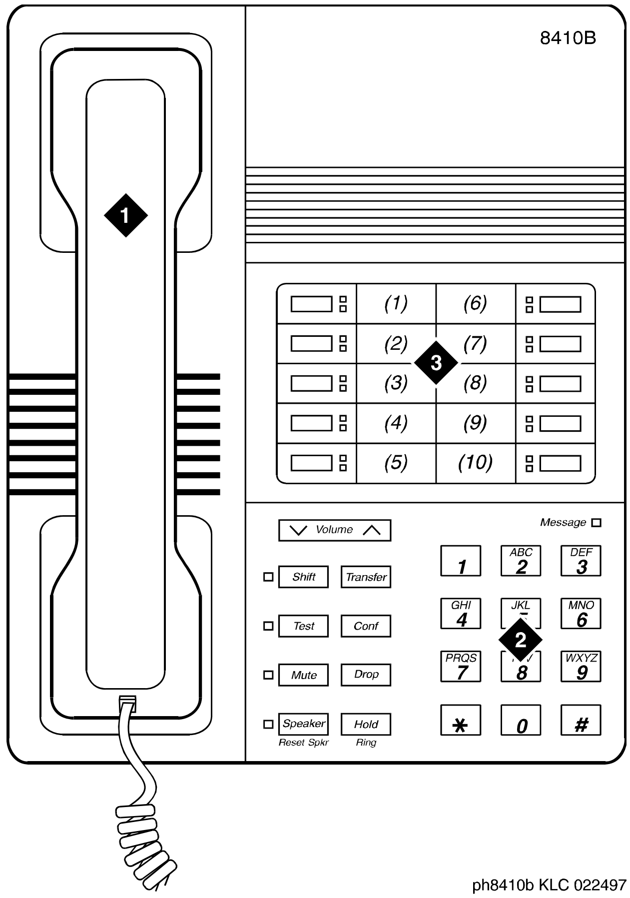


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Note:

The 8405B and 8405B+ look like the 8410B with the exception that the 8405 series do not have the second column of line appearances.

Figure 250: 8410D telephone

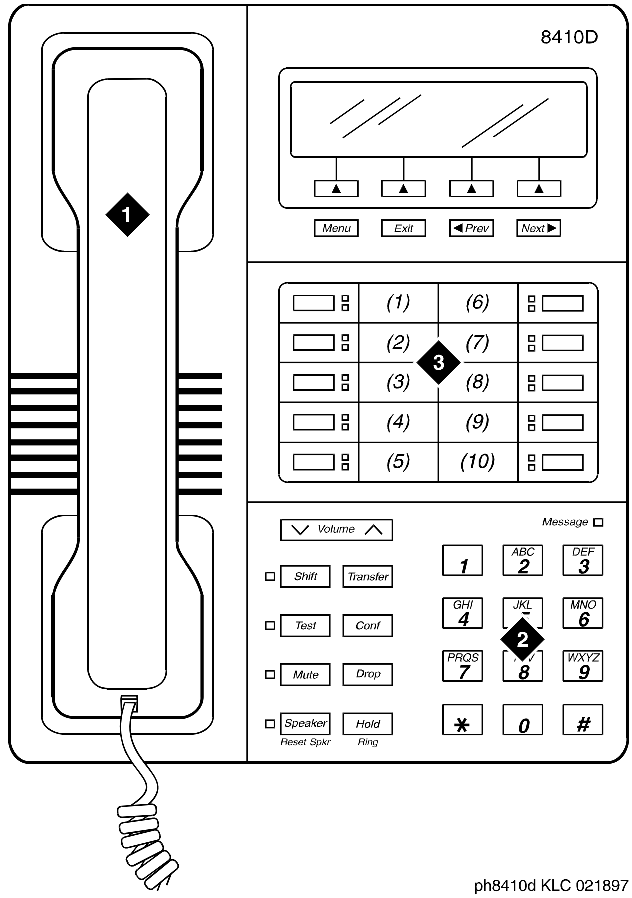


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**

Note:

The 8405D and 8405D+ look like the 8410D with the exception that the 8405 series do not have the second column of line appearances.

Figure 251: 8434D telephone

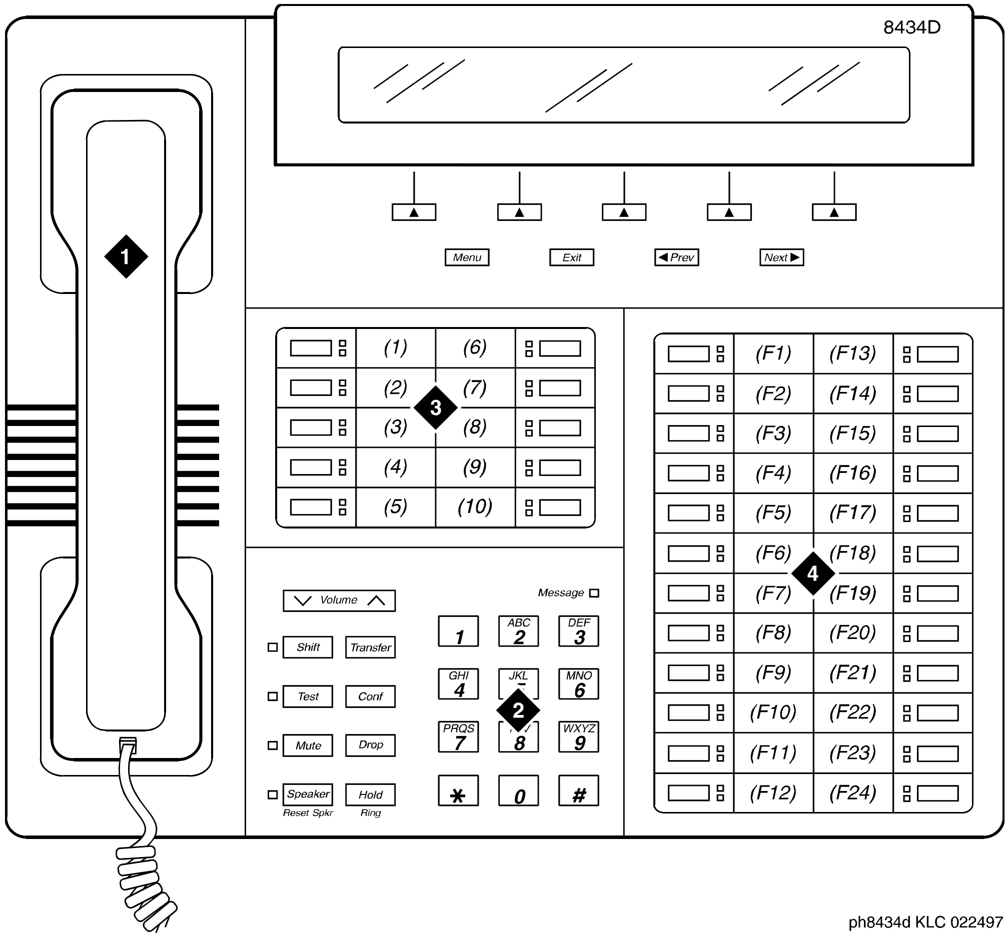


Figure notes:

1. **Handset**
2. **Dial pad**
3. **10 programmable buttons**
4. **24 feature buttons**

8110 telephones

The basic 8110 (8110A01A, 8110A01B, and 811A01C) and the modified 8110M (8110A01D) telephones are single-line analog telephones. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons with PROGRAM and PAUSE buttons, automatic redial, a flashing red Message light, and a Hold button. They also have built-in speakerphones with Mute capability and the Automatic Answer feature.

9600-series IP telephones

The 9600 Series IP telephones are similar to the 4610/4620 line of H.323 IP telephones in how they communicate with Communication Manager, but have a different look and feel. These telephones use Internet Protocol (IP) technology with Ethernet line interfaces and support the H.323 protocol only. The 9600 Series IP Telephones provide support for DHCP, HTTP, and HTTPS over IPv4/UDP, which enhance the administration and servicing of the telephones. These telephones use DHCP to obtain dynamic IP Addresses, and HTTPS or HTTP to download new versions of software or customized settings for the telephones. The 9610 cannot be used by Call Center Agents; hence, it cannot/should not be translated for use with IP Agent.

9610 IP telephone

Communication Manager supports the 9610 IP telephone like the 4606 IP telephone as detailed below, with exceptions as noted:

- One fixed call appearance is included in the button information download to the phone, as well as support for four fixed Communication Manager features: Directory, Next, Call Display, and Normal (also known as Exit) – these are included in the button information download to the phone, but with blank labels
- 40 character display message support
- Support for Communication Manager call processing features
- Support of ID Request/Response – with support for a new set ID number of hex code D4
- Support for update of the Message Waiting Indicator status
- Support for Communication Manager call processing features when invoked by Feature Access Codes, including the Auto-Call Back feature
- No support for a button expansion module
- No speaker or speakerphone functionality (note that the “none” setting is downloaded to the phone)
- No support for enabling or disabling the mute feature (note that the “n” setting is downloaded to the phone)

Note:

The 9610 supports only one call appearance and four fixed Communication Manager features, which are presented via the context-sensitive softkeys.The 9610 does not support any expansion modules. Note that if used on a Communication Manager release prior to Communication Manager 4.0, you must administer the 9610 as a 4606 for full functionality.

9620 IP telephone

Communication Manager supports the 9620 IP telephone like the 4610 IP telephone as detailed below, with exceptions as noted:

- Support for the conference, transfer, hold, exit, and drop features
- 12 administrable call appearance/feature buttons (with a maximum of 10 call appearances or a maximum of 12 bridged appearances)
- Speakerphone functionality (none, one-way, and two-way)
- 40 character display message support
- Support for Communication Manager call processing features (except for those features specifically blocked via administration)
- Support of ID Request/Response – with support for a new set ID number of hex code D0
- Support for a new Voice Mail Retrieval function (via the phone Message button)
- Support for update of the Message Waiting Indicator status
- No support for a button expansion module

Note:

The 9620 does not have fixed conference, transfer, hold, or drop buttons, but shows these features as context-sensitive softkeys. From the Communication Manager perspective, however, the feature behavior is the same as that of the 4610 telephone. While the 4610 allows 24 buttons, the 9620 only supports 12 buttons. The Communication Manager Voice Mail Retrieval capability is supported on the 9620 station. Like the 4610, the 9620 does not support any expansion modules.

9630/9640 IP telephone

Communication Manager supports the 9630 and 9640 IP telephones like the 4620 IP telephone as detailed below, with exceptions as noted:

- 24 administrable call appearance/feature buttons per main telephone
- Support for up to three button expansion modules (SBM24). Each SBM24 allows an additional 24 call appearance/feature buttons per module
- Support for a maximum of 96 call appearance/feature buttons between the phone and three SBM24s. (with a maximum of 10 call appearances or a maximum of 96 bridged appearances)
- Speakerphone functionality (none, one-way, and two-way)
- 40 character display message support

- Support for Communication Manager call processing features (except for those features specifically blocked via administration)
- Support of ID Request/Response – with support for a new set ID number of hex code D1for the 9630 and D5 for the 9640
- Support for a new Voice Mail Retrieval function (via the phone Message button)
- Support for update of the Message Waiting Indicator status

Note:

The 9630 and 9640 IP telephones do not have fixed conference, transfer, hold, or drop buttons, but show these features as context-sensitive softkeys. From the Communication Manager perspective, however, the feature behavior is the same as that of the 4620 telephone. While the 4620 allows one feature expansion module for a total of up to 48 buttons, the 9630/9640 supports up to three button expansion modules for a maximum of 96 buttons. Like the 4620, the number of call appearances is 10, but the number of bridged appearances can grow to 96. The Communication Manager Voice Mail Retrieval capability is also supported on the 9630/9640 stations.

9650 IP telephone

Communication Manager supports the 9650 IP telephone like the 4620 IP telephone as detailed below, with exceptions as noted:

- Support for the conference, transfer, hold, exit, and drop features
- 24 administrable call appearance/feature buttons per main telephone (16 of the 24 buttons are be displayed on the 9650's auxiliary buttons)
- Support for up to three button expansion modules (SBM24). Each SBM24 allows an additional 24 call appearance/feature buttons per module
- Support for a maximum 96 call appearance/feature buttons between the phone and three SBM24s. (maximum of 10 call appearances or a maximum 96 bridged appearances)
- Speakerphone functionality (none, one-way, and two-way)
- 40 character display message support
- Support for Communication Manager call processing features (except for those features specifically blocked via administration)
- Support of ID Request/Response – with support for a new set ID number of hex code D2
- Support for a new Voice Mail Retrieval function (via the phone Message button)
- Support for update of the Message Waiting Indicator status

Note:

The 9650 does not have fixed conference, transfer, hold, or drop buttons, but show these features as context-sensitive softkeys. From the Communication Manager perspective, however, the feature behavior is the same as that of the 4620 telephone. While the 4620 allows one feature expansion module for a total of up to 48 buttons, the 9650 supports up to three button expansion modules supporting a maximum of 96 buttons. Like the 4620, the number of call appearances is 10, but the number of bridged appearances can grow to 96. The Communication Manager Voice Mail Retrieval capability is also supported on the 9650 stations.

CALLMASTER telephones

There are several types of CALLMASTER telephones:

- 602A and 602D CALLMASTER

  The 602 CALLMASTER models have a display, a Message light, a Mute button, and four fixed feature buttons: Conference, Drop, Hold, and Transfer. You can administer its 10 call appearance/feature (2-lamp) buttons and its 17 feature-only (1-lamp) buttons.

- 603D (CALLMASTER II)

  The CALLMASTER II model has a display, a Message light, and the Mute, Select, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons.

- 603E (CALLMASTER III)

  The CALLMASTER III model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER III to either a standard 4-wire DCP or a 2-wire circuit pack.

- 603F (CALLMASTER IV)

  The CALLMASTER IV model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER IV to either a standard 4-wire DCP or a 2-wire circuit pack.

- 606A (CALLMASTER VI)

  The CALLMASTER VI model is a miniature, 8-button, 2-headset jack, digital telephone that is controlled by the user's personal computer (PC) through an RS-232 serial-port connection.

- 607A (CALLMASTER V)

  The CALLMASTER V model has a display, softkeys, and the display control buttons (Menu, Exit, Previous, and Next). This model does not have a standard handset, but you can connect a handset to one of its headset jacks. The CALLMASTER V has six fixed feature buttons: Speaker, Mute, Hold, Redial, Conference, and Transfer. You can administer its 16 call appearance/feature (2-lamp) buttons, however, one of these buttons must be administered as a Headset On/Off button and a second one must be administered as a Release button. You can also administer the 12 softkey buttons.

Figure 252: CALLMASTER II/CALLMASTER III digital telephone

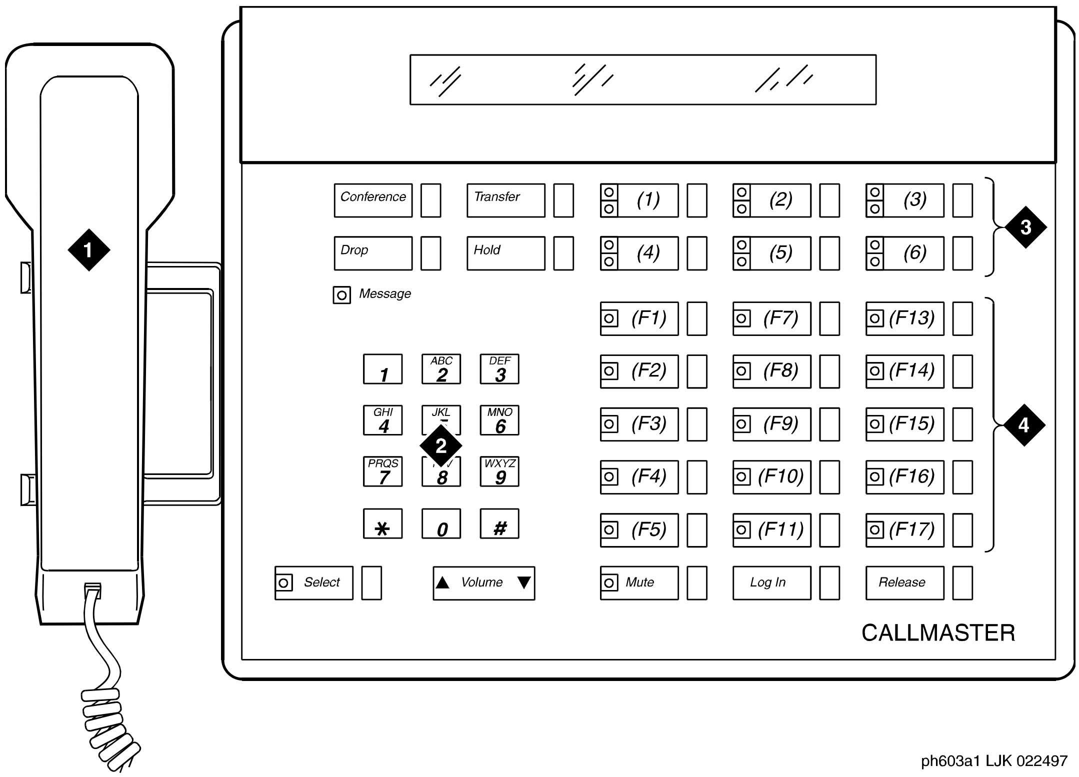


Figure notes:

1. Handset
2. Dial pad
3. 6 programmable buttons
4. 18 feature buttons

Figure 253: CALLMASTER IV digital telephone

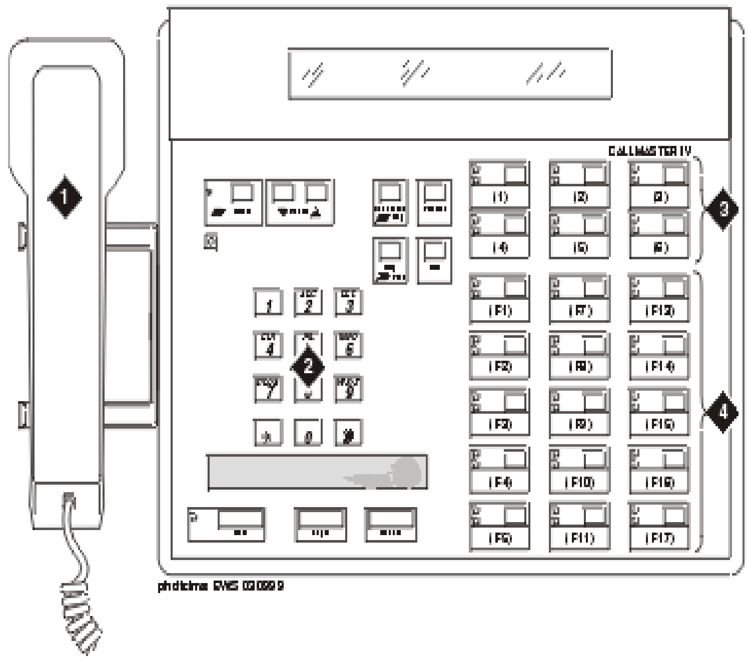


Figure notes:

1. Handset
2. Dial pad
3. 6 programmable buttons
4. 15 feature buttons

Figure 254: CALLMASTER V digital telephone

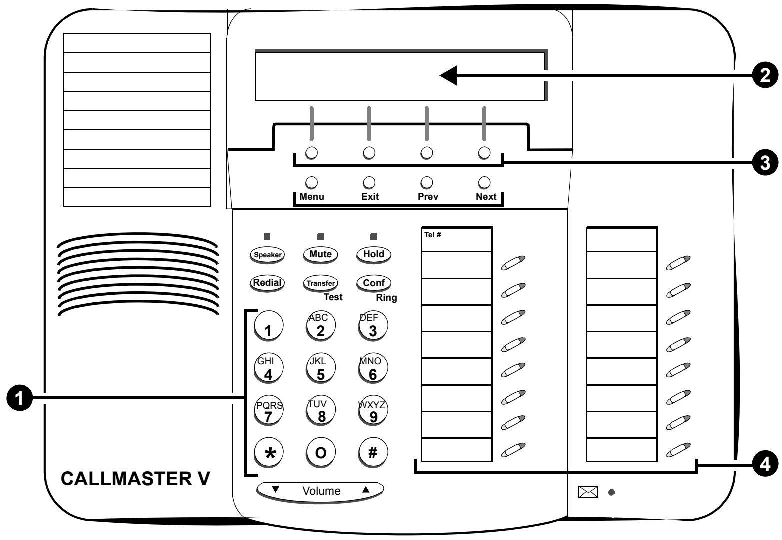


Figure notes:

1. Dial pad
2. Display
3. 4 softkey buttons
4. 16 call appearance/feature buttons

Cordless telephone

MDC9000 cordless telephone

The MDC 9000 Cordless Telephone has two basic parts, the handset and the charging base.

- The handset has line/programmable feature/intercom buttons, Conference, Drop, Hold, Transfer, Status, and Feature buttons, Headset On/Off and Handset On/Off buttons, a LCD display, an earpiece volume control switch, battery charging contacts, a directory card, and a headset jack.
- The charging base has a handset hook, ringer volume controls, battery charging contacts, a Talk indicator, a Charge indicator, a Message indicator, and a base ringer.

MDW9000 cordless telephone

The MDW 9000 Wireless Telephone is part of the TransTalk 9000 Digital Wireless System family of telephones. This wireless telephone has three basic parts, the handset the charging cradle, and the radio module.

- The handset has line/programmable feature/intercom buttons, Drop, Mute, Hold, Conference, and Transfer buttons, a Headset On/Off button and a Handset On/Off button, a LCD display, a Volume control switch, battery charging contacts, a flexible antenna, and a Headset jack.
- The charging cradle has a handset hook, a spare battery cover, a spare battery.
- The radio module has Power, Pass, and Radio indicator lights, a top hook, card edge, and snap lock which connect the radio module to the carrier assembly/backplane, an antenna, and power plug and line connectors.

Figure 255: MDC9000 and MDW9000 cordless telephones

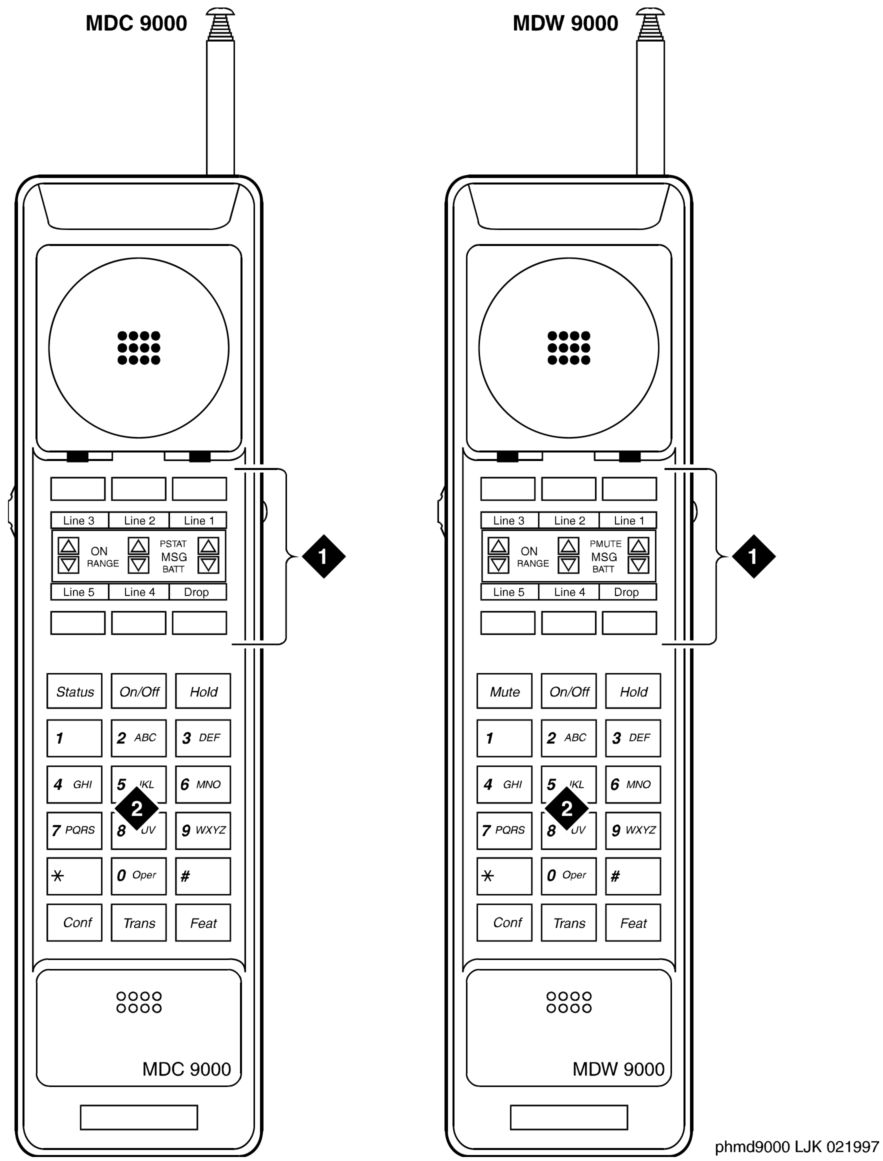


Figure notes:

1. 6 programmable buttons
2. Dial pad
3. 6 programmable buttons
4. Dial pad

Figure 256: MDC9000 and MDW9000 cordless telephones

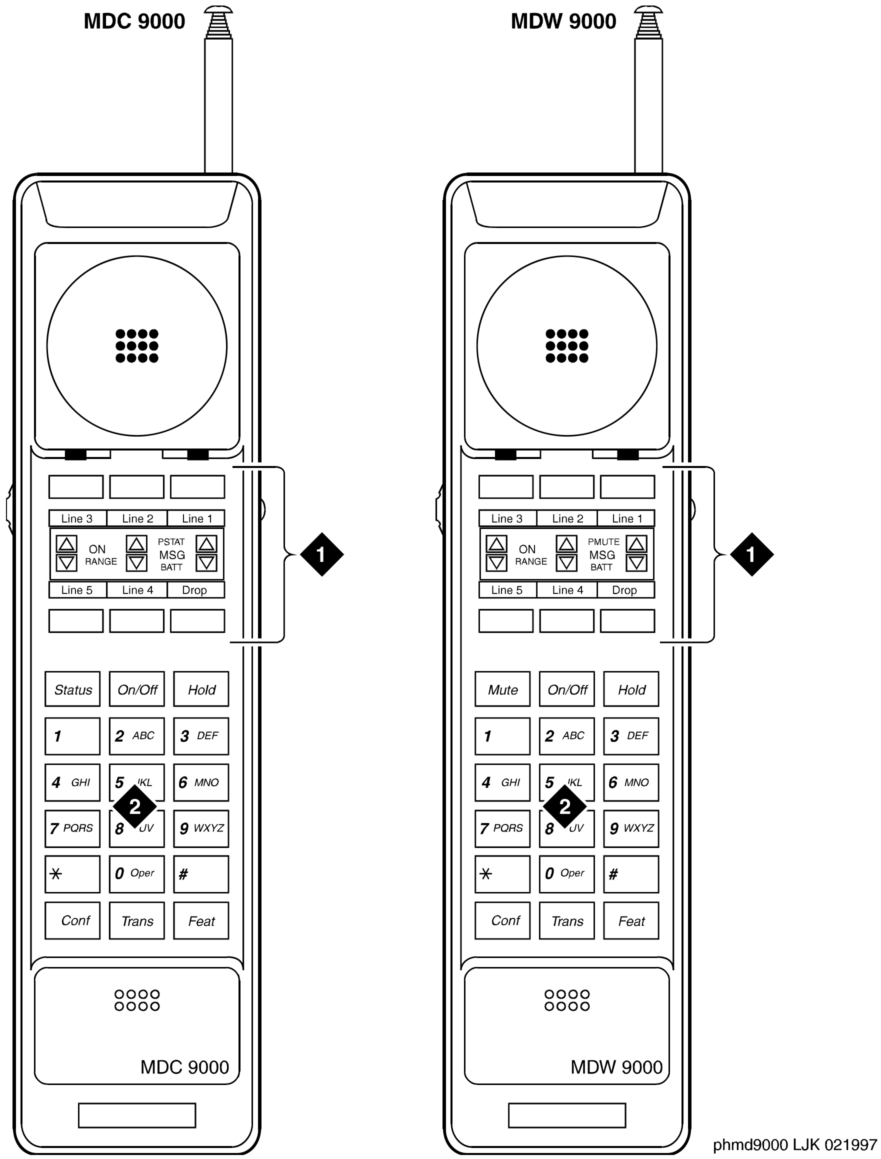


Figure notes:

1. 6 programmable buttons
2. Dial pad
3. 6 programmable buttons
4. Dial pad

Internet Protocol (IP) Softphones

IP Softphones can turn a PC or a laptop into an advanced telephone. Using the Internet Protocol (IP) network, you can place calls, take calls, and handle calls on your PC. With certain exceptions, every feature available for wired-endpoint voice calling is available for IP-based calling; it supports internetworking with conventional circuit-switched stations and trunks.

There are three softphone models that are fixed at the time of purchase and do not change afterward:

- Avaya IP Softphone (for additional information, go to http://www.avaya.com/support, click on **Telephones and End User Devices**, then scroll down to **IP Telephony**)
- Avaya IP Agent (for additional information, go to http://www.avaya.com/support, click on **Call Center/CRM**, then scroll down to **IP Agent**)
- Avaya Softconsole (for additional information, go to http://www.avaya.com/support, click on **Call Center/CRM**, then scroll down to **IP Telephony**)

The connection for the end-point is supported whether the customer is directly connected using a network inter-face card (NIC) or the customer has dialed into their network using a dial-up point-to-point (PPP) account. The connection is managed via a registration session with the Avaya server. The call server manages the voice connection directly to a circuit-switched telephone (Telecommuter configuration), directly to an IP Telephone (Control of IP Telephone configuration), directly to a DCP digital telephone (Control of DCP Telephone configuration), or over UDP/RTP to the IP Softphone's iClarity IP Audio H.323 audio end-point (Road Warrior or VoIP configuration).

At any time, an end user can put a softphone into one of the following modes by changing a software menu:

- Road-Warrior (Voice over IP (VoIP))
- Telecommuter
- Native H.323 (only available with Avaya IP Softphone R2)
- Control of IP Telephone (only available with IP Softphone R4 and later)
- Control of DCP Telephone (only available with IP Softphone R5 and later)

Note:

Beginning with the November 2003 release of Communication Manager, R1 and R2 IP Softphone and IP Agent, which use a dual connect (two extensions) architecture, are no longer supported. R3 and R4 IP Softphone and IP Agent, which use a single connect (one extension) architecture, continue to be supported. This applies to the RoadWarrior and the Telecommuter configurations for the IP Softphone. Native H.323 registrations for R1 and R2 Softphones continue to be supported.

Note:

Beginning with the January 2005 release of Communication Manager, the Message Flow Control feature is available. When this feature is enabled, previously existing IP soft clients will fail to register. The clients listed below are impacted by this condition.

- IP Softphone R3V1
- IP Softphone R3V2
- IP Softphone R3V2.1
- IP Softphone R4
- IP Softphone R5.0
- IP Softphone R5.1
- IP Agent V3
- IP Agent R4
- IP AgentR5
- Softconsole 1.0
- Softconsole 1.5

For assistance, contact your Avaya technical support representative.

Road-warrior mode

The road-warrior (VoIP) mode enables travelers to use Communication Manager features from temporary remote locations, such as a hotel room. The road-warrior configuration uses two separate software applications running on a user's PC that is connected to Communication Manager over an IP network. The single network connection carries two channels: one for call-control signaling and one for voice. IP Softphone software handles the call signaling and an H.323 V2-compliant audio application such as Avaya iClarity, which is installed with Avaya IP Softphone Release 3 and later (or a third-party application, like Microsoft NetMeeting) handles the voice communications. The user's PC provides the audio connection via a microphone and speakers.

Telecommuter mode

The telecommuter mode enables remote workers to use Communication Manager features from a remote location, such as while telecommuting from a home office. The telecommuter configuration uses two connections to Avaya Communication Manager:

- a connection to the PC over the IP network
- a connection to the telephone over the public-switched telephone network (PSTN)

The PC user places and takes calls with the IP Softphone interface and uses the telephone handset to speak and listen.

You can also use a variation of the telecommuter for call center agents: Avaya IP Agent. This mode uses the Avaya IP Agent interface instead of the IP Softphone interface to emulate a remote CallMaster telephone.

Stand-alone H.323

The stand-alone H.323 mode enables travelers to use some Avaya Communication Manager features from a remote location. This mode uses a PC running an H.323 v2-compliant audio application, such as Microsoft NetMeeting. The H.323 mode controls the call signaling and the voice path. However, since it does not use the IP Softphone interface, this configuration is capable of operating only as an analog or single-line telephone making one call at a time without any additional assigned features. You can provide stand-alone H.323 users only features that can they can activate with dial access codes.

Control of IP Telephone

This mode allows you to make and receive calls under the control of the IP Softphone - just like in the Telecommuter or Road Warrior mode. The big difference is that you have a true, feature-rich digital telephone under your control. In the Road Warrior mode, there is no telephone. In the Telecommuter mode, the telephone you are using (whether analog, digital, or IP telephone) is used only to carry audio and not for any features. In this mode (if you have an IP telephone), you get the best of both worlds. The terminals that are supported for this registration mode are shown in Supported Terminals for Control of Terminal/Station on page 719.

Control of DCP Telephone

This feature provides a registration endpoint configuration that will allow an IP softphone and a non-softphone telephone to be in service on the same extension at the same time. In this new configuration, the call control is done by both the softphone and the telephone endpoint. This feature supports speakerphones, handsets, and headsets.

Table 12: Supported Terminals for Control of Terminal/Station

Terminal Type	Control of the Terminal/Station			
	"Regular" Control		"Shared" Control	
	Road Warrior	Telecommuter	Via IP Telephone	Via the server
2402	Yes	Yes	N/A	Yes
2410	Yes	Yes	N/A	Yes
2420	Yes	Yes	N/A	Yes
2420 w/Expansion Module	Yes	Yes	N/A	Yes
4601	Yes	Yes	No	Yes
4602	Yes	Yes	No	N/A
4606	Yes	Yes	Yes	Yes
4610	Yes	Yes	Yes	Yes
4612	Yes	Yes	Yes	Yes
4620	Yes	Yes	Yes	Yes
4620 w/Expansion Module	Yes	Yes	Yes	N/A
4624	Yes	Yes	Yes	Yes
6402D	Yes	Yes	N/A	Yes
6408D/6408D+	Yes	Yes	N/A	Yes
6416D+	Yes	Yes	N/A	Yes
6416D+ w/Expansion Module	Yes	Yes	N/A	Yes
6424D+	Yes	Yes	N/A	Yes
				1 of 2

Table 12: Supported Terminals for Control of Terminal/Station (continued)

Terminal Type	Control of the Terminal/Station			
	"Regular" Control		"Shared" Control	
	Road Warrior	Telecommuter	Via IP Telephone	Via the server
6424D+ w/Expansion Module	Yes	Yes	N/A	Yes
8405D/8405D+	Yes	Yes	N/A	No
8410D	Yes	Yes	N/A	No
8411D	Yes	Yes	N/A	No
8434D	Yes	Yes	N/A	No
8434D w/Expansion Module	Yes	Yes	N/A	No
6402	Yes	Yes	N/A	N/A
6408/6408+	Yes	Yes	N/A	N/A
7403D	Yes	Yes	N/A	N/A
7404D	Yes	Yes	N/A	N/A
7405D	Yes	Yes	N/A	N/A
7406+/7406D	Yes	Yes	N/A	N/A
7407+/7407D	Yes	Yes	N/A	N/A
7410+/7410D	Yes	Yes	N/A	N/A
7434D	Yes	Yes	N/A	N/A
7444D	Yes	Yes	N/A	N/A
8403B	Yes	Yes	N/A	N/A
8405B/8405B+	Yes	Yes	N/A	N/A
8410B	Yes	Yes	N/A	N/A
8411B	Yes	Yes	N/A	N/A
				2 of 2

Related topics

For instructions on how to administer an IP Softphone on your system, see Adding an IP Softphone on page 104.

On the IP Softphone CD, see *IP Softphone Overview and Troubleshooting* and *IP Softphone Getting Started*.

Chapter 19: Screen Reference

This chapter contains descriptions of Communication Manager screens that are used in performing administrative tasks. These are most often screens that are invoked using commands such as **`add`**, **`change`**, and **`remove`**.

For maintenance-related screens that are invoked using commands such as **`list`**, **`display`**, and **`status`**, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

AAR and ARS Digit Analysis Table

Avaya Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Note:

Typing the command **`change aar analysis`** or **`change ars analysis`** displays an all-locations Digit Analysis screen. To access a per-location screen, type **`change aar analysis location`** ***`n`*** or **`change ars analysis location`** ***`n`***, where ***`n`*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 257: AAR Digit Analysis Table screen

```
change aar analysis n                                          Page 1 of X

                             AAR DIGIT ANALYSIS TABLE
                                   Location:All         Percent Full:

     Dialed          Total     Route     Call    Node   ANI
     String         Min Max   Pattern    Type    Num    Reqd
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
```

Figure 258: ARS Digit Analysis Table screen

```
change ars analysis                                          Page 1 of X

                             ARS DIGIT ANALYSIS TABLE
                                   Location: All         Percent Full:

     Dialed          Total     Route     Call    Node   ANI
     String         Min Max   Pattern    Type    Num    Reqd
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
_________________   __  __     _____     ____    ___     n
```

ANI Reqd

Valid entries	Usage
y/n	Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Multifrequency-Signaling-Related System Parameters** screen is **n**.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field on the **Feature-Related System Parameters** screen is **y**. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Call Type (for AAR only)

Enter the call type associated with each dialed string. Call types indicate numbering requirements on different trunk networks. ISDN Protocols are listed in the table below.

Valid entries	Usage
aar	Regular AAR calls
intl	The Route Index contains public network ISDN trunks that require international type of number encodings.
pubu	The Route Index contains public network ISDN trunks that require unknown type of number encodings.
lev0 to lev2	Specify ISDN Private Numbering Plan (PNP) number formats. (See Numbering — Private Format on page 1396 for more information.)
unku	The **unku** AAR Call Type makes it easier to set up an Implicit (Unknown) Numbering Plan, in which users dial each other by extension (optionally preceded by a node number), without an ARS or AAR Access Code (e.g., “9” or “8”).

ISDN Protocol

Call Type	Numbering Plan Identifier	Type of Numbering
aar	E.164(1)	national(2)
intl	E.164(1)	international(1)
pubu	E.164(1)	unknown(0)
lev0	PNP(9)	local(4)
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (for ARS only)

Valid entries	Usage	China # 1 Call Type
alrt	alerts attendant consoles or other digital telephones when an emergency call is placed	normal
emer	emergency call	normal
fnpa	10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit "1")	attendant
hnpa	7-digit NANP call	normal
intl	public-network international number	toll-auto
iop	international operator	attendant
locl	public-network local number	normal
lpvt	local private	normal
natl	non-NANP	normal
npvt	national private	normal
nsvc	national service	normal
op	operator	attendant
pubu	public-network number (E.164)-unknown	normal
svcl	national(2)	toll-auto
svct	national(2)	normal
svft	service call, first party control	local
svfl	service call, first party control	toll

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number. For example, if a user dials 297-1234 and the AAR or ARS Digit Analysis Table has dialed string entries of 297-1 and 297-123, the match is on the 297-123 entry.

An exact match is made on a user-dialed number and dialed string entries with wildcard characters and an equal number of digits. For example, if a user dials 424, and there is a 424 entry and an X24 entry, the match is on the 424 entry.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Location

This is a display-only field. Typing the command **`change aar analysis`** ***n*** or **`change ars analysis`** ***n*** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change aar analysis location`** ***n*** or **`change ars analysis location`** ***n***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Avaya Communication Manager that uses this AAR/ARS Digit Analysis Table. On the **System Parameters Customer-Options (Optional Features)** screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear. For ARS, the **ARS** field must also be set to **y** on the **System Parameters Customer-Options (Optional Features)** screen.
all	Indicates that this AAR/ARS Digit Analysis Table is the default for all port network (cabinet) locations. Appears only if the **Multiple Locations** field is **n** on the **System Parameters Customer-Options (Optional Features)** screen.

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Node Number

Valid entries	Usage
1 to **999** or blank	Enter the number of the destination node in a private network if you are using node number routing or DCS. If you complete this field, leave the **Route Index** field blank.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS.

Route Pattern

Enter the route number you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
p1 to **p2000**	Specifies the route index number established on the Partition Routing Table
1 to **640**	Specifies the route pattern used to route the call.
1 to **999**	Specifies the route pattern used to route the call. For S8300 Servers only.
r1 to **r32**	Specifies the remote home numbering plan area table. Complete this field if RHNPA translations are required for the corresponding dialed string.
node	Designates node number routing
deny	Blocks the call

AAR and ARS Digit Conversion Table

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits can be inserted or deleted from the dialed number. For instance, you can tell the server running Communication Manager to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

Note:

Typing the command **`change aar digit-conversion`** or **`change ars digit-conversion`** displays the all-locations Digit Conversion Table screen. To access a per-location screen, type **`change aar digit-conversion location`** ***n*** or **`change ars digit-conversion`** ***n***, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 259: AAR Digit Conversion Table screen

```
change aar digit-conversion                                    Page 1 of 2

                           AAR DIGIT CONVERSION TABLE
                                  Location:All                Percent Full:


  Matching Pattern      Min    Max    Del    Replacement String   Net   Conv ANI Req
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
  __________________    __     __     __     __________________   ___    _       _
```

Figure 260: ARS Digit Conversion Table screen

```
change ars digit-conversion                                      Page 1 of 2

                       ARS DIGIT CONVERSION TABLE
                             Location:All                  Percent Full:

  Matching Pattern     Min   Max   Del   Replacement String   Net Conv ANI Req
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
  __________________   __    __    __    __________________   ___   _       _
```

Note:

When you access the screen with the **`display`** or **`change`** command, the entries are sorted in the order of the matching pattern. Digits appear before characters.

ANI Req

This field applies only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Multifrequency-Signaling-Related System Parameters** screen is **n**.

Valid entries	Usage
y/n	Enter **y** to require ANI on incoming R2-MFC or Russian MF ANI calls. Must be **y** to enable EC500 origination features.
r	Allowed only if the **Allow ANI Restriction on AAR/ARS** field is **y** on the **Feature-Related System Parameters** screen. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion.

Del

Valid entries	Usage
0 to Min	Number of digits you want the system to delete from the beginning of the dialed string.

Location

This is a display-only field. Typing the command **`change aar digit-conversion`** ***`n`*** or **`change ars digit-conversion`** ***`n`*** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change aar digit-conversion location`** ***`n`*** or **`change ars digit-conversion location`** ***`n`***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Avaya Communication Manager for this AAR/ARS Digit Conversion Table. On the **System Parameters Customer-Options (Optional Features)** screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear. For ARS, the **ARS** field must also be set to **y** on the **System Parameters Customer-Options (Optional Features)** screen.
all	Indicates that this AAR/ARS Digit Conversion Table is the default for all port network (cabinet) locations.

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the number you want the server running Avaya Communication Manager to match to dialed numbers. If a Prefix Digit 1 is required for 10-digit direct distance dialing (DDD) numbers, be sure the matching pattern begins with a 1.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this Matching Pattern.

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to this Matching Pattern.

Net

Enter the call-processing server network used to analyze the converted number.

Valid entries	Usage
ext, aar, ars	Analyze the converted digit-string as an extension number, an AAR address, or an ARS address.

Percent Full

Displays the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Replacement String

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
*	
#	Use # to indicate end-of-dialing. It must be at the end of the digit-string.
blank	

Abbreviated Dialing List

This screen establishes system-wide or personal lists for speed dialing.

Enhanced List

The Enhanced Abbreviated Dialing List can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Note:

Dialing must be enabled in your license file before you can program an Enhanced List. When the feature is enabled, the **Abbreviated Dialing Enhanced List** field on the **System Parameters Customer-Options (Optional Features)** screen displays **y**.

You can define two Enhanced Abbreviated Dialing Lists in the system. Before you assign numbers to a list, you must define whether you want a 3-digit or 4-digit enhanced list on the **Feature-Related System Parameters** screen. If you select 3-digit enhanced list, the list can be up to 10 separate screens numbered from 0 to 9 that allow you to define up to 1000 numbers. If you select a 4-digit enhanced list, a list can include up to 100 separate screens numbered 0 to 99 that allow you to assign up to 10,000 numbers on each list. The two Enhanced Abbreviated Dialing Lists together can support up to 20,000 entries.

If you want your attendants to use abbreviated dialing, you must also administer the **Console Parameters** screen.

Figure 261: Abbreviated Dialing Enhanced List screen

```
display abbreviated-dialing enhanced                                 Page 1 of x
                                 ABBREVIATED DIALING LIST
                                     Enhanced List
                     Size (multiple of 5): 5                    Privileged? n
    DIAL CODE
      100: ________________________
      101: ________________________
      102: ________________________
      103: ________________________
      104: ________________________
      105: ________________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
* (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Privileged

Indicates whether users of this list can dial any number in the list, regardless of the COR of the station from which they dial.

Valid entries	Usage
y/n	Set this field to **n** if you want the system to verify that this station is allowed to dial this number.

Size (multiple of 5)

The number of dial code list entries you want in this list.

Valid entries	Usage
5 to **100**, in multiples of 5	Up to 100 entries per screen

Group List

This screen implements the Abbreviated Dialing Group List. The Group Lists are controlled by the System Administrator. Up to 100 numbers can be entered per group list that can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 262: Abbreviated Dialing Group List screen

```
change abbreviated-dialing group                                    Page 1 of X
                            ABBREVIATED DIALING LIST
                               Group List: ____
       Size (multiple of 5):   5  Program Ext: ________   Privileged? n
    DIAL CODE
         01: ________________
         02: ________________
         03: ________________
         04: ________________
         05: ________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Group List

This is a display-only field when the screen is accessed using an administration command such as **`add`** or **`change`**.

Valid entries	Usage
Display-only field	Enter a group number when completing a paper screen.

Privileged

Valid entries	Usage
y	If **y** is entered, the calling telephone's class of restriction (COR) is never checked and any number in the group list will be dialed.
n	If **n** is entered, the calling telephone's COR is checked to determine if the number can be dialed.

Program Ext

Enter the extension that you want to give permission to program the Group List.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Personal List

This screen establishes a personal dialing list for telephone/data module users. The personal list must first be assigned to the telephone by the System Administrator before the telephone user can add entries in the list. The lists can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 263: Abbreviated Dialing Personal List screen

```
change abbreviated-dialing personal                           Page 1 of x
                         ABBREVIATED DIALING LIST

                  Personal List: ________ List Number: ___
                  Size (multiple of 5): 5
DIAL CODE
     01: ________________
     02: ________________
     03: ________________
     04: ________________
     05: ________________
     06: ________________
     07: ________________
     08: ________________
     09: ________________
     00: ________________
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than **5** in the **Size** field, the system increases the number of dial codes to the number you specified.

Note:

Although the **Abbreviated Dialing Personal List** screen shows dial codes with a leading zero (i.e., 01, 02, 03), the user enters only the digit following the zero and not the zero itself to successfully access the extension administered on that dial code.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds.

List Number

A display-only field indicating which of the three personal lists is defined for the telephone.

Personal List

A display-only field indicating the extension of the telephone that will use this list.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

System List

This screen implements a system abbreviated-dialing list. Only one system list can be assigned and is administered by the System Administrator. The list can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Figure 264: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                                  Page  1 of x
                         ABBREVIATED DIALING LIST

                                 SYSTEM LIST
Size (multiple of 5): 100      Privileged? n      Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  11:                                  11:*************
  12:                                  12:*************
  13:                                  13:*************
  14:                                  14:*************
  15:                                  15:*************
  16:                                  16:*************
  17:                                  17:*************
  18:                                  18:*************
  19:                                  19:*************
  20:                                  20:*************
  21:                                  21:*************
  22:                                  22:*************
  23:                                  23:*************
  24:                                  24:*************
  25:                                  25:*************
```

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the **Size** field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
* (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds.

Label Language

This field provides administration of personalized labels on the 2420/4620 telephone sets. If this field is changed to another language, all administered labels in the original language are saved and the labels for the new language are read in and displayed.

Valid entries	Usage
English **Italian** **French** **Spanish** **user-defined** **Unicode**	Enter the appropriate language for the 2420/4620 labels. **Note:** Unicode display is only available for Unicode-supported telephones. Currently, the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones support Unicode display. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

LABELS FOR 2420/4620 STATIONS

This field provides the administrative capability to actually customize the labels for the system-wide Abbreviated Dial buttons on the 2420/4620 telephone sets.

Valid entries	Usage
A-Z, a-z, 0-9, and ! & * ? ; ’ ^ ( ) , . : -	Up to 15 alphanumeric characters

Privileged

Valid entries	Usage
y	Enter **y** if the originating party’s class of restriction (COR) is never checked and any number in the list can be dialed.
n	Enter **n** if the COR is to be checked to determine if the number can be dialed.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

The Figure 265 shows the last page of the **Abbreviated Dialing System** screen when, on the **System Parameters Customer-Options (Optional Features)** screen, the **A/D Grp/Sys List Dialing Start at 01** field is **n**.

Figure 265: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                               Page   7 of  x
                         ABBREVIATED DIALING LIST

                                 SYSTEM LIST
                                        Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  01:                                  01:*************
  02:                                  02:*************
  03:                                  03:*************
  04:                                  04:*************
  05:                                  05:*************
  06:                                  06:*************
  07:                                  07:*************
  08:                                  08:*************
  09:                                  09:*************
  10:                                  10:*************
```

Figure 266 shows the last page of the **Abbreviated Dialing System** screen when, on the **System Parameters Customer-Options (Optional Features)** screen, the **A/D Grp/Sys List Dialing Start at 01** field is **y**.

Figure 266: Abbreviated Dialing System List screen

```
add abbreviated-dialing system                               Page   7 of  x
                         ABBREVIATED DIALING LIST

                                 SYSTEM LIST
                                        Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  91:                                  91:*************
  92:                                  92:*************
  93:                                  93:*************
  94:                                  94:*************
  95:                                  95:*************
  96:                                  96:*************
  97:                                  97:*************
  98:                                  98:*************
  99:                                  99:*************
  00:                                  00:*************
```

7103A Button List

This screen assigns abbreviated dialing numbers to the 7103A telephone buttons. The entries can then be accessed by 7103A telephone users to place local, long-distance, and international calls; activate/deactivate features; or to access remote computer equipment. This screen applies only to 7103A fixed feature telephones. Only one 7103A abbreviated dialing list can be implemented in the system and it applies to all 7103A fixed feature telephones in the system. This list is controlled by the System Administrator.

Figure 267: Abbreviated Dialing List — 7103A Button List screen

```
display abbreviated-dialing 7103A-buttons                    Page 1 of x
                             ABBREVIATED DIALING LIST
                                7103A Button List

      DIAL CODE (FOR THE 7103A STATION BUTTONS)
       1: ________________________      5. ________________________
       2: ________________________      6. ________________________
       3: ________________________      7. ________________________
       4: ________________________      8. ________________________
```

DIAL CODE

Enter the number you want to assign to each dial code (button). Any additions or changes apply to all 7103A fixed feature telephones. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability might be impaired.

Vector Directory Number extension can also be assigned.

Valid entries	Usage
Digits 0 to 9	Up to 24 characters
* (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Mark

Valid entries	Usage
~s	Start suppressing display of the digits being outpulsed.
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds. Not available for S8300 Servers.

Access Endpoint

This screen administers Access Endpoints and Wideband Access endpoints.

Note:

You can administer Wideband Access Endpoints only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Wideband Switching** field is **y**.

An Access Endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. Access Endpoints eliminate the need to dedicate an entire trunk group for the access of a single trunk by providing the capability to assign an extension number to a single trunk.

An Access Endpoint can be specified as the Originator or Destination endpoint of an administered connection.

A Wideband Access Endpoint (WAE) is an endpoint application connected to line-side non-ISDN T1 or E1 facilities and, like Access Endpoints, have no signaling interface with the system.

The WAE is defined by a starting port (DS0) and a width specifying the number of adjacent nonsignaling DS0s (positioned within a DS1 facility) that make up the endpoint. This width can be between 2 and 31 adjacent DS0s.

Note:

Access Endpoints and Wideband Access Endpoints consume the same resources that trunks use. Thus, the sum of Access Endpoints and trunks cannot exceed the maximum number of trunks available in your system configuration.

Field descriptions for page 1

Figure 268: Access Endpoint screen

```
add access-endpoint next                                          Page   1 of   x
                              ACCESS ENDPOINT

            Extension: 30001      (Starting) Port:_______
   Communication Type: voice-grade-data      Name:_____________

                  COR: 1                      COS: 1
                   TN: 1                      ITC: restricted
```

Communication Type

Valid entries	Usage
voice-grade-data	For an analog tie trunk access endpoint.
56k-data	For a DS1 access endpoint enter as appropriate (**64k-data** is not allowed for robbed-bit trunks).
64k-data	
wideband	For a Wideband access endpoint

COR

The COR is administered so that only an administered connection (AC) endpoint can be connected to another AC endpoint.

Valid entries	Usage
0 to **995**	Enter the appropriate class of restriction (COR) number.

COS

The COS is administered (see Class of Service on page 852) so that the use of the Call Forwarding All Calls feature for access endpoints is prohibited.

Valid entries	Usage
0 to **15**	Enter the appropriate COS number.

Extension

A display-only field showing the extension number as specified in the command line, or shows the next available extension number if **next** was entered on the command line. This is the extension number assigned to the nonsignaling trunk and used to access the trunk endpoint.

ITC (Information Transfer Capability)

This field is used to determine the type of transmission facilities to be used for ISDN calls originating from this endpoint. Displays when the **Communication Type** field is **56k-data**, **64k-data**, or **Wideband**.

When adding an access endpoint with the ITC administered as unrestricted, its associated port has to be a channel of a DS1 circuit pack with the **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

An unrestricted access endpoint can only be from B8ZS DS1 circuit pack.

When adding an access endpoint with the ITC administered as restricted, its associated port can be a channel from a DS1 circuit pack with the **Zero Code Suppression** field administered as ZCS or B8ZS.

For an existing access endpoint, ITC can only be changed from restricted to unrestricted if its associated port is a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its **Zero Code Suppression** field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

```
An unrestricted access endpoint can use only B8ZS DS1 circuit pack
```

Without this end validation, a user could administer an access endpoint as unrestricted when in fact it is restricted, that is, its associated port is a member of a DS1 circuit pack that uses ZCS data transmission.

Valid entries	Usage
unrestricted	When **unrestricted**, only unrestricted transmission facilities (b8zs) will be used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is). For Wideband Access Endpoints, enter **unrestricted**.
restricted	When **restricted**, either restricted (zcs-ami) or unrestricted transmission facilities is used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of eight digital zeros is converted to a sequence of seven zeros and a digital one) via zcs coding on DS1 circuit pack.

Name

Enter an name for the endpoint.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

(Starting) Port

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number.
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway

Valid entries	Usage
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Note:
For Wideband Access Endpoints, analog tie trunks cannot be used and the DS1 Interface circuit pack, Version C or later, must be used.

The DS1 circuit number corresponds to the channel that will carry the data traffic. Channels 1 through 31 (DS1 Interface only) or channels 1 through 24 (DS1 Tie Trunk, DS1 Interface, or DS1 Interface (32) circuit packs) can be used when the **DS1 Signaling Type** field is **robbed-bit** or **isdn-ext**. For Common Channel or ISDN-PRI signaling, channel use is limited to channels 1 through 30 (DS1 Interface circuit pack only) or channels 1 through 23 (DS1 Interface (32) or DS1 Interface). A channel can be administered as an access endpoint regardless of the DS1 signaling type.

TN

Valid entries	Usage
1 to 100	Enter the Tenant Partition number.

Width

Appears if the **Communication Type** field is **wideband**. This field cannot be blank.

Valid entries	Usage
2 to **31**	Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the WAE.
6	A width of **6** defines a 384 Kbps WAE.

Administered Connection

This screen assigns an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The AC is established automatically by the system whenever the system restarts or the AC is due to be active. See "Administered Connections" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, and Access Endpoint on page 745 for additional information.

Field descriptions for page 1

Figure 269: Administered Connection screen

```
change administered-connection                                Page 1 of x
                         ADMINISTERED CONNECTION
  Connection Number:  1                          Enable? y
         Originator:  ________
        Destination:  __________________________________
               Name:  __________________

AUTHORIZED TIME OF DAY

         Continuous? n
                     Sun? n Mon? n Tue? n Wed? n Thu? n  Fri? n  Sat? n
         Start Time:  00:00
           Duration:  000:00

  MISCELLANEOUS PARAMETERS

                 Alarm Type:  warning  Alarm Threshold: 5
                                        Retry Interval: 2
                   Priority:  5       Auto Restoration? y
```

Connection Number

This is a display-only field showing an unassigned AC number when the screen is accessed using an administration command such as **`change`** or **`display`**.

Destination

Used to route the AC to a desired endpoint. Enter the address of the destination access or data endpoint. This endpoint is the terminating party of the AC and need not be local to the server on which the AC is assigned. The entry must be consistent with the local Communication Manager server's dial plan (that is, the first digits are assigned as an extension, feature access code, or trunk access code, or DDD Number). If a local extension is entered, it must be assigned to either an access or data endpoint. Abbreviated Dialing entries can be used in this field.

Valid entries	Usage
Extension/string	Enter the assigned access endpoint/data module extension or valid dialed string.

Enable

Provides the administered connection.

Valid entries	Usage
y	Indicates an attempt will be made to establish the AC when the AC is due to be active.
n	The AC will not be made or if it is up, it will drop.

Name

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (S8300 Server, S87XX IP-PNC only)	Enter a short identification of the AC. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Originator

Enter the assigned access endpoint/data module extension.

Data Line circuit pack

- Asynchronous EIA 232C compatible equipment

Digital Line circuit pack connections, including:

- MPDM (700D), MTDM (700B, 700C, 700E), 7400D data module
- 7400A, 7400B, 7400C HSL, 8400B data module
- 7401D telephone with 7400B or 8400B data module
- 7403D/7405D/7407D/7410D/7434D telephone with DTDM or 7400B or 8400B data module
- 7404D or 7406D telephone
- 510D personal terminal
- 515 BCT, 615 BCT, or 715 BCT terminal
- Connection between PC and the server running Avaya Communication Manager

ISDN-BRI Line circuit pack connections, including:

- 7500 data module
- 7505D/7506D/7507D telephone with ADM

Valid entries	Usage
Assigned access endpoint/ data module extension	The endpoint must be local to the server on which the AC is administered. Nonsignaling DS1 trunk or analog tie trunk.

Authorized Time of Day

Continuous

The connection will be up all the time or re-established if the connection goes down.

Valid entries	Usage
y	Indicates that the AC is continuous (that is, not scheduled to be active at a certain time). If **y** is entered, the seven **Start Days** and associated **Duration** fields do not appear.
n	Displays the **Start Days** fields.

Duration

Enter the period of time that the scheduled AC should remain active. This period is specified in two fields separated by a colon. The maximum duration is 167 hours and 59 minutes (that is, 1 minute less than 1 week). Only appears if the **Continuous** field is **n**.

Valid entries	Usage
000 through **167**	For the **hour** field.
00 through **59**	For the **minute** field.

Start Days (Sun through Sat)

These fields indicate only the days on which an attempt will be made to establish the AC and not necessarily the days it is active. A scheduled AC might be active over a number of days, and, in this situation, these fields should be used only to specify the days on which the AC starts and not other days on which the AC might be active. Only appears if the **Continuous** field is **n**.

Valid entries	Usage
y	Enter **y** in each of the required days of the week fields to indicate that an attempt will be made to establish the AC.
n	Displays the day fields.

Start Time

Only appears if the **Continuous** field is **n**.

Valid entries	Usage
00:00 through **23:59**	Enter the time of the day when an attempt should begin to establish a scheduled AC. The time is specified in two fields separated by a colon.

Miscellaneous Parameters

Alarm Threshold

Only appears if an entry in the **Alarm Type** field is other than **none**. Enter the number of times an attempt to establish or reestablish an AC must fail consecutively before an AC alarm generates. (An alarm will be generated after the fourth retry has failed; thus, with the retry interval of 2 minutes, an alarm will be generated approximately 8 minutes after the first failure occurs.)

Valid entries	Usage
1 through **10**	An alarm generates on the first failure if this field is **1**.

Alarm Type

Enter the type of alarm to be generated if the AC cannot be initially established, or fails and cannot be reestablished, and the number of consecutive failures equals the alarm threshold. All AC alarms and the errors that caused the alarms are recorded in the system's alarm and error log. In addition, a status lamp associated with an attendant console or telephone feature button (**ac-alarm**) can be used to indicate the AC alarm.

Valid entries	Usage
major	Failures that cause critical degradation of service and require immediate attention.
minor	Failures that cause some degradation of service, but do not render a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems might be impairing service to a few trunks or stations or interfering with one feature across the entire system.
warning	Failures that cause no significant degradation of service or failures in equipment external to the system. Warning alarms are not reported to the attendant console or INADS.
none	The alarm notification is disabled for this AC.

Auto Restoration

Valid entries	Usage
y	Enter **y** to indicate an attempt is to be made to reestablish an AC that failed. Auto restoration is available only for an AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group. A **y** in this field is ignored in all other situations.

Priority

Enter a number that is to be used to determine the order in which ACs are to be established.

Valid entries	Usage
1 to **8**	1 is the highest and 8 the lowest priority.

Retry Interval

Valid entries	Usage
1 to **60**	Enter the number of minutes between attempts to establish or reestablish the AC.

Agent LoginID

Use this screen in an Expert Agent Selection (EAS) environment to add or change agent login IDs and skill assignments. If you add or change skills on the Avaya S8XXX Server, the agent must log out and then log in again before the changes take effect. Note that in non-EAS (basic Automatic Call Distribution) environments, this screen does not appear at all, and agents are assigned directly on the **Hunt Group** screen. The agent's properties are assigned to the physical telephone extension. For more information, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Field descriptions for page 1

Figure 270: Agent LoginID screen

```
add agent-loginID 9011                                              Page 1 of X
                                   AGENT LOGINID

            Login ID: 9011_                                          AAS? _
                Name:_________________________                     AUDIX? _
                  TN: 1__                                  LWC Reception: spe
                 COR: 1                        AUDIX Name for Messaging: ______
       Coverage Path: ____          Messaging Server Name for Messaging: ______
       Security Code: ____                     LoginID for ISDN Display? n
                                                                Password: ______
                                                  Password (enter again): ______
                                                             Auto Answer: ______
                                                       MIA Across Skills: ______
                                               ACW Agent Considered Idle: ______
                                               AUX Work Reason Code Type: ______
                                                 Logout Reason Code Type: ______
                          Maximum time agent in ACW before logout (sec): ______
                                                 Forced Agent Logout Time: ___:__



            WARNING:  Agent must log in again before skill changes take effect
```

AAS

Enter **y** if this extension is used as a port for an Auto Available Split/Skill. Default is **n**.

Entering **y** in the **AAS** field clears the password and requires execution of the **`remove agent-loginid`** command. To set AAS to **n**, remove this logical agent and add it again. This option is intended for switch adjunct equipment ports only, not human agents.

ACW Agent Considered Idle

Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue. Valid entries are **system** (default), **n**, and **y**. The **system** value indicates that settings assigned on the **Feature-Related System Parameters** screen apply.

Audix

Enter **y** if this extension is used as a port for AUDIX. Default is **n**. The **AAS** and **AUDIX** fields cannot both be **y**.

Audix Name for Messaging

Do one of the following actions:

- Enter the name of the messaging system used for LWC Reception, or
- Enter the name of the messaging system that provides coverage for this Agent LoginID.

Auto Answer

When using EAS, the agent's auto answer setting applies to the station where the agent logs in. If the auto answer setting for that station is different, the agent's setting overrides the station's setting. The following entries are valid:

- **all** - immediately sends all ACD and non ACD calls to the agent. The station is also given a single ring while a non-ACD call is connected. The **ringer-off** button can be used to prevent the ring when, on the **Feature-Related System Parameters** screen, the **Allow Ringer-off with Auto-Answer** field is set to **y**.
- **acd** - only ACD split /skill calls and direct agent calls go to auto answer. If this field is **acd**, non ACD calls terminated to the agent ring audibly.
- **none** - all calls terminated to this agent receive an audible ringing treatment. This is the default.
- **station** - auto answer for the agent is controlled by the auto answer field on the **Station** screen.

Aux Work Reason Code Type

Valid entries	Usage
system	Settings assigned on the **Feature-Related System Parameters** screen apply. This is the default.
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.

COR

Enter the Class of Restriction for the agent. Valid entries are **0** to **995**. Default is **1**.

Coverage Path

Enter the number of the coverage path used by calls to the LoginID. Valid entries are a path number between **1** and **999**, time of day table **t1** to **t999**, or blank (default). This is used when the agent is logged out, does not answer, or is busy to personal calls when logged in.

Direct Agents Calls First

This field replaces the **Service Objective** field when percent-allocation is entered in the **Call Handling Preference** field. Enter **y** if you want direct agent calls to override the percent-allocation call selection method and be delivered before other ACD calls. Enter **n** if you want direct agent calls to be treated like other ACD calls. For more information, see the *Avaya Business Advocate User Guide*, 07-300653.

Forced Agent Logout Time

This field enables the Forced Agent Logout by Clock Time feature by administering a time of day to automatically log out agents using an hour and minute field. Valid entries for the hour field are **01-23**. Valid entries for the minute field are **00**, **15**, **30**, and **45**. The default is blank (not administered). Examples: 15:00, 18:15, 20:30, 23:45.

Login ID

Display-only field. Contains the identifier for the Logical Agent as entered on the command line.

LoginID for ISDN Display

Enter **y** if the Agent LoginID CPN (Calling Party Number) and Name field is to be included in ISDN messaging over network facilities. If set to **n** (the default), the physical station extension CPN and Name is sent. The **Send Name** field on the ISDN Trunk Group screen prevents sending out the calling party name and number if set to **n**, and may prevent sending it if set to **r** (restricted).

Logout Reason Code Type

Valid entries	Usage
system	Settings assigned on the **Feature-Related System Parameters** screen apply.This is the default.
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.

LWC Reception

Enter the name of the messaging system where Leave Word Calling messages for this Agent Login ID will be stored. Valid entries are **audix**, **msa**, **spe** (default), and **none**.

Maximum time agent in ACW before logout (sec)

This field is used for setting a maximum time the agent can be in ACW on a per agent basis. Valid entries are:

- **system** - Settings assigned on the **Feature-Related System Parameters** screen apply. This is the default.
- **none** - ACW timeout does not apply to this agent.

- **30-9999** sec - Indicates a specific timeout period. This setting will take precedence over the system setting for maximum time in ACW.

Messaging Server Name for Messaging

Do one of the following actions:

- Enter the name of the Messaging Server used for LWC Reception.
- Enter the name of the Messaging Server that provides coverage for this Agent LoginID.
- Leave blank (default).

MIA Across Skills

Enter **y** to remove an agent from the MIA queues for all the splits or skills that the agent is available in when the agent answers a call from any of the assigned splits or skills. Enter **n** to exclude ACW agents for the queue.Valid entries are **system** (default), **n**, and **y**. The **system** value indicates that settings assigned on the **Feature-Related System Parameters** screen apply.

Name

Enter up to a 27-character string naming the agent. Any alpha-numeric character is valid. Default is blank.

Note:

The **Name** field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones. For more information on Unicode language display, see Administering Unicode display on page 203. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Password

Appears only if both the **AAS** and **AUDIX** fields are **n**. Enter up to nine digits as the password the Agent must enter upon login. Valid entries are the digits **0** through **9**. Enter the minimum number of digits in this field specified by the **Minimum Agent-LoginID Password Length** field on the **Feature-Related System Parameters** screen. Default is blank. Values entered in this field are not displayed on the screen.

Password (enter again)

Appears only if both the **AAS** and **AUDIX** fields are n. Reenter the same password exactly as it was entered in the **Password** field. Default is blank. Values entered in this field are not displayed on the screen.

Port Extension

Appears only if either the **AAS** or **AUDIX** field is **y**. Enter the assigned extension for the AAS or AUDIX port. This extension cannot be a VDN or an Agent LoginID. Default is blank.

Security Code

Enter the 4-digit security code (password) for the Demand Print messages feature. This field can be blank (default).

TN

Enter the partition number for tenant partitioning. Valid entries are **1** to **20**. Default is **1**.

Field descriptions for page 2

The second page of the **Agent LoginID** screen contains agent skill information.

Figure 271: Agent Login ID screen

```
add agent-loginID 9011                                           Page 2 of X
                                   AGENT LOGINID
      Direct Agent Skill:                                   Service Objective?
Call Handling Preference:                               Local Call Preference?


    SN RL SL PA          SN RL SL PA          SN RL SL PA          SN RL SL PA
 1: __ _  __ ___    16:  __ _  __ ___    31: __ _  __ ___    46: __ _  __ ___
 2: __ _  __ ___    17:  __ _  __ ___    32: __ _  __ ___    47: __ _  __ ___
 3: __ _  __ ___    18:  __ _  __ ___    33: __ _  __ ___    48: __ _  __ ___
 4: __ _  __ ___    19:  __ _  __ ___    34: __ _  __ ___    49: __ _  __ ___
 5. __ _  __ ___    20:  __ _  __ ___    35: __ _  __ ___    50: __ _  __ ___
 6: __ _  __ ___    21:  __ _  __ ___    36: __ _  __ ___    51: __ _  __ ___
 7: __ _  __ ___    22:  __ _  __ ___    37: __ _  __ ___    52: __ _  __ ___
 8: __ _  __ ___    23:  __ _  __ ___    38: __ _  __ ___    53: __ _  __ ___
 9: __ _  __ ___    24:  __ _  __ ___    39: __ _  __ ___    54: __ _  __ ___
10. __ _  __ ___    25:  __ _  __ ___    40: __ _  __ ___    55: __ _  __ ___
11: __ _  __ ___    26:  __ _  __ ___    41: __ _  __ ___    56: __ _  __ ___
12: __ _  __ ___    27:  __ _  __ ___    42: __ _  __ ___    57: __ _  __ ___
13: __ _  __ ___    28:  __ _  __ ___    43: __ _  __ ___    58: __ _  __ ___
14: __ _  __ ___    29:  __ _  __ ___    44: __ _  __ ___    59: __ _  __ ___
15. __ _  __ ___    30:  __ _  __ ___    45: __ _  __ ___    60: __ _  __ ___
```

Call Handling Preference

When calls are in queue and an agent becomes available, the **skill-level** setting delivers the highest priority, oldest call waiting for the agent's highest level skill. Other choices are **greatest-need** and **percent-allocation**. **Greatest-need** delivers the oldest, highest priority call waiting for any of the agent's skills. **Percent allocation** delivers a call from the skill that will otherwise deviate most from its administered allocation. **Percent-allocation** is available only with Avaya Business Advocate software. For more information, see the *Avaya Business Advocate User Guide*, 07-300653.

Direct Agent Skill

Enter the number of the skill used to handle Direct Agent calls. Valid entries are **1** to **99**, or blank (default).

Local Call Preference

Enter **y** to indicate that for calls queued in more than one skill for a multi-skilled EAS agent, the system should give preference to matching the trunk location number of the queued call to the location number of the previously-busy agent. Valid settings are **n** (default) or **y**. You can only set this field to **y** if the **Call Center Release** field on the **Feature-Related System Parameters** screen is **3.0** or later, and the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**.

PA

Percent Allocation. If the call handling preference is **percent-allocation**, you must enter a percentage for each of the agent's skills. Enter a number between 1 and 100 for each skill. Your entries for all of the agent's skills together must total 100%. Do not use target allocations for reserve skills. Percent Allocation is available as part of the Advocate software.

RL

Reserve Level. Enter any reserve levels assigned to this agent with the Service Level Supervisor feature. You can assign a reserve level of 1 or 2. When this skill reaches the corresponding EWT threshold set on the **Hunt Group** form, this agent will automatically be logged into the skill and will take calls until the skill's EWT drops below the preassigned overload threshold. Service Level Supervisor is available as part of the Advocate software.

Service Objective

Appears only when **Call Handling Preference** is **greatest-need** or **skill-level**. Valid entries are **y** or **n**. Service Objective is administered on the **Hunt Group** screen and the **Agent LoginID** screen. The server selects calls for agents according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and the administered service objective for the skill. Service Objective is a feature that is part of the Advocate software.

SL

Skill Level. Enter a skill level for each of an agent's assigned skills. If EAS-PHD is not optioned, 2priority levels are available. If EAS-PHD is optioned, 16 priority levels are available. In releases prior to R3V5, level 1 was the primary skill and level 2 was the secondary skill.

SN

Skill Number. Enter the Skill Hunt Group(s) that this agent handles. The same skill cannot be entered twice. Consider the following options:

- If EAS-PHD is not optioned, enter up to four skills.
- If EAS-PHD is optioned, enter up to 20 or 60 skills depending on the platform.
- Assigning a large number of skills to agents can potentially impact system performance. Review system designs with the ATAC when a significant number of agents have greater than 20 skills per agent.

Alias Station

This screen allows you to configure the system so that you can administer new telephone types that are not supported by your system software. This screen maps new telephone models to a supported telephone model. This mapping does not guarantee compatibility, but allows unsupported models to be administered and tracked by their own names.

Some administrators also use this screen to "name" non-telephone devices. For example, you know that you can add a modem to your system by simply administering the extension as the standard analog type 2500. But, if you listed your stations, how would you know which extensions are modems? Instead, you could use the **Alias** screen to create a ‘modem’ alias to type 2500 and enter modem in the **Type** field for every modem you add to your system. For more information, see Adding a fax or modem on page 103.

Tip:

When you upgrade a system that uses an alias set type to a new release, the system determines if the aliased type is supported in the new release (is now a native set type). When you review the **Alias Station** screen, you might see alias types that have become native. If the type is now native, the last character of the aliased set type becomes a "#."

Field descriptions for page 1

Figure 272: Alias Station screen

```
change alias station                                                 Page 1 of x
                                    ALIAS STATION
                        Alias Set Type    Supported Set Type

                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                             ______              ______
                '#' indicates previously aliased set type is now native
```

Alias Set Type

Enter up to a 5-character name for the non-supported telephone type that you want to alias to a similar supported telephone type. Do not use blank characters.

Supported Set Type

Enter a supported telephone type that you want to map (or alias) to the alias set type. Valid supported telephone types are listed in Telephones on page 1539.

Note:

Data Communication Protocol (DCP) telephone types must be aliased to DCP telephone types, hybrid types to hybrid types, and analog to analog types.

Alphanumeric Dialing Table

This screen associates alpha-names to dialed digit strings. This allows telephone users to place a data call by simply typing the alpha-name. Users need only remember far-end alpha-names instead of the actual digit strings.

The screen consists of paired **Alpha-name/Mapped String** fields. Entries can be made in any order on the screen. However, before the screen is displayed for changing or reviewing, the entries in the table are sorted alphanumerically by the alpha-name. All entries will be moved to the beginning of the table, leaving all blank entries at the end.

Field descriptions for page 1

Figure 273: Alphanumeric Dial Table screen

```
change alphanumeric-dial-table                                  Page 1 of x

                        ALPHANUMERIC DIALING TABLE
  XXX of XXX administered
  Alpha-name          Mapped String         |  Alpha-name          Mapped String
                                            |
  ____________  ______________________      |  _____________ ____________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ___________________
  ____________  ______________________      |  _____________  ______________________
```

Alpha-name

All alpha-names in the table must be unique and cannot be referenced in their own "Mapped String." The alpha-names can be used in any other "Mapped String" and multiple times in a "Mapped String."

Valid entries	Usage
From 1 to 8 alphanumeric characters	The entry must start with an alphabetic character and cannot have blank spaces between characters.

Mapped String

Enter from 1 to 24 characters that might contain alphanumeric, readability, delimiters, and/or special characters. The entry is used to generate the final dialing string and can include Facility Access Codes.

Note:

A Mapped String cannot contain an Alpha-Name whose Mapped String also contains an Alpha-Name.

Valid entries	Usage
Digits 0 to 9	Numeric
A through **Z**, **a** through **z**	Alpha (note uppercase entries are mapped to lowercase)
(	Readability character
)	Readability character
/	Readability character
-	Readability character
+	Wait for dial tone
%	Rest of digits are for end to end signaling
","	Pause for 1.5 seconds
space	Readability character
#	DTMF digit pound
*	DTMF digit asterisk
^	Readability character

Announcements/Audio Sources

Use this screen to assign announcements to circuit packs and port locations.

Field descriptions for page 1

Figure 274: Announcements/Audio Sources screen

```
add announcement 26451                                              Page 1 of X
                          ANNOUNCEMENTS/AUDIO SOURCES

 Extension: 26451                                  COR: 1
 Annnc Name: collect_some_digits                     TN: 1
 Annc Type: integrated                           Queue? y
Group/Port:                               Queue Length?
 Protected? n                                    Rate: 64
```

Annc Name

Valid entries	Usage
up to 27-character alpha-numeric filename (no ., /, :, *, ?, <, >, \, .wav, or blanks in this field for VAL circuit packs only)	Enter the name of the announcement you are associating with the specified extension. For VAL announcements, this field is required. The value in this field becomes the filename of the announcement. The **.wav** file extension, which is part of the filename stored on the circuit pack, does not appear. Do not enter **.wav** as part of the filename. Names on a single VAL circuit pack must be unique. The system checks for duplicate filenames on the same VAL circuit pack. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Annc Type

Enter the type of announcement you want to assign to this extension number.

If you enter **integrated** or **integ-rep**, complete the **Queue**, **Protected**, **Rate**, and **Port** fields. If you enter **analog**, **ds1-fd**, **ds1-sa**, **ds1-ops**, or **aux-trunk**, complete **Queue Length** (if **Q** is **y**) and **Port**.

Valid entries	Usage
analog	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Avaya Communication Manager through an analog port. Ringing starts playback.
analog-m	Use for continuous playing music or audio source from an external announcement device.
analog-fd	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to the server running Avaya Communication Manager through an analog port. Ringing starts playback. Sends forward disconnect signal to stop playback.
aux-trunk	Auxiliary trunk. Use with an external announcement device with a 4-wire "aux" interface.
aux-trk-m	Auxiliary trunk. Use with continuously playing music or audio sources that do not indicate playback is active.
ds1-fd	Assigned to DS1 ports on circuit packs. Callers do not hear a click when the device hangs up. Provides a disconnect to stop playback when the announcement is done.
ds1-ops	Callers do not hear a click when the device hangs up.
ds1-sa	Provides a disconnect to stop playback when the announcement is done. Callers do not hear a click when the device hangs up.
integrated	Stored internally on the Avaya DEFINITY or Avaya S8XXX Server on a special integrated announcement circuit pack. Use for general announcements and VDN of Origin Announcements.
integ-mus	Integrated music source.
integ-rep	Integrated repeating

COR

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) you want associated with this announcement.

Extension

Valid entries	Usage
1 to 7 digits	The extension number associated with the announcement being added/ displayed/changed/removed. This field is display-only. It is auto-populated based on the extension entered in the command line.

CAUTION:

When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Group/Port

Use this field to enter the announcement board location or the Audio Group number. If **Annc Type** is **integrated**, then this field displays as **Group/Board**. Also, when **Annc Type** is integrated, the **Queue Length** field does not appear. If **Annc Type** is *not* **integrated**, then the **Group/Port** field displays as **Port**.

Type the group number in one of the following ways:

- **Gnn** where ***nn*** represents a one or two-digit audio group number.
- The location of the VAL or the TN750 announcement circuit pack. Enter the necessary characters in the **aaxss** format (where ***aa*** = the cabinet number, ***x*** = the carrier, and ***ss*** = the slot number).
- **gggv9** for media gateway vVAL, where ***ggg*** is the gateway number of the media gateway (up to 250).

Note:

To administer DID Intercept announcements in a multi-location system where each location or city needs a different announcement, enter an audio group in this field instead of a VAL port.

Protected

Use this field to set the protection mode for an integrated announcement/music extension. When you set this field to **y**, the recording is protected and cannot be deleted or changed via a telephone session or FTP (via SAT or VAL Manager). When you set this field to **n**, the recording can be changed or deleted by users with console permissions to delete or change the recording. Changing or deleting using the telephone recording session requires the **console permissions** class of service (COS). When the **Type** is **analog**, **ds1** or **aux-trunk**, **N/A** appears in this field.

Valid entries	Usage
y	Enter **y** to protect the integrated announcement from being deleted or changed by any user. For VAL, after an announcement file resides on the circuit pack (recorded or FTP transfer), you can set this field to **y** to protect the file (read-only).
n	Enter **n** to allow telephone session users with console permission and/or FTP to change or delete an announcement. Use this value when you initially administer an announcement or subsequently need to change or delete it.

Queue

Valid entries	Usage
y(es)	Enter **y** to queue calls for the announcement if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. The caller is always connected to the beginning of the announcement. Enter **y** for ACD and vectoring delay announcements. Call centers should always use this option. This is the default.
n(o)	No queue and no barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is not available.
b(argein)	Enter **b** to set up barge-in if the **Type** field is **integrated**, **integ-rep** or **aux-trunk**. When **Type** is **integ-mus**, this field defaults to **b**. Callers are connected to the announcement at any time while it is playing. **Note:** The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated circuit pack. The initial request to play an announcement selects an available port on the board on which the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue (**Q** field must be **y**). Otherwise, the request to play is denied, and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform "calls connected" limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

Queue Length

The queue length is the number of calls that can queue for this announcement. The maximum number of queues allowed depends on your system configuration.

The **Queue Length** field applies if the **Queue** field is **y** and the **Type** field is **analog**, ds1 or **aux-trunk**. When the **Type** field is **integrated** or **integ-rep**, **N/A** appears in this field. Integrated announcements have a pre-set queue length

.

Valid entries	Usage
The maximum number your system allows	Number of calls that can be queued for this announcement

Rate

Enter the recording rate speed (in 1000 bits/second) for TN750 or ISSPA integrated announcements. A different recording speed can be used for each integrated announcement. With VAL type sources, the default is **64** and cannot be changed. When the **Type** field is **analog**, **ds1** or **aux-trunk**, **N/A** appears in this field.

Valid entries	Usage
16	16 kbps (8 minutes and 32 seconds of announcement time per circuit pack or 1 hour and 24 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 240 minutes of storage time). This rate does not provide a high-quality recording. Avaya does not recommend this for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps (4 minutes and 16 seconds of total announcement time for the TN750; for the ISSPA, there are 120 minutes of storage time).
64	64 kbps (for 2 minutes and 8 seconds of announcement time per circuit pack or 42 minutes for 10 circuit packs for the TN750; for the ISSPA, there are 60 minutes of storage time). This is the default for VAL.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number, if any.

ARS Toll Table

This screen assigns ARS Toll Tables used by Subnet Trunking. Use it to specify whether calls to CO codes listed on the table are toll or non-toll calls. You specify non-toll calls based on the last 2 digits of the distant-end of the trunk group.

Field descriptions for page 1

Figure 275: ARS Toll Table screen

```
change ars toll
                             ARS TOLL TABLE: __               Page 1 of x

                            OFFICE CODES: x00-x99
   00: y   10: y   20: y   30: y  40: y  50: y  60: y  70: y  80: y  90: y
   01: y   11: y   21: y   31: y  41: y  51: y  61: y  71: y  81: y  91: y
   02: y   12: y   22: y   32: y  42: y  52: y  62: y  72: y  82: y  92: y
   03: y   13: y   23: y   33: y  43: y  53: y  63: y  73: y  83: y  93: y
   04: y   14: y   24: y   34: y  44: y  54: y  64: y  74: y  84: y  94: y
   05: y   15: y   25: y   35: y  45: y  55: y  65: y  75: y  85: y  95: y
   06: y   16: y   26: y   36: y  46: y  56: y  66: y  76: y  86: y  96: y
   07: y   17: y   27: y   37: y  47: y  57: y  67: y  77: y  87: y  97: y
   08: y   18: y   28: y   38: y  48: y  58: y  68: y  78: y  88: y  98: y
   09: y   19: y   29: y   39: y  49: y  59: y  69: y  79: y  89: y  99: y
```

00: through 99:

These fields represent the last 2 digits of the codes within the 100-block of numbers. Designate each as a number toll or non-toll call.

Valid entries	Usage
y/n	Enter **n** to designate a non-toll CO code.

ARS TOLL TABLE

Valid entries	Usage
2 through **9**	Identify the number of the ARS Toll Table.

OFFICE CODES

Valid entries	Usage
200 to **299** through **900** to **999**	Identify the block of numbers on this screen.

Attendant Console

This screen assigns an Attendant Console to the system.

Field descriptions for page 1

Figure 276: Attendant Console screen

```
add attendant n                                           Page   1 of  x
                             ATTENDANT CONSOLE 1

         Type: console          Name: 27 character attd cons name
    Extension: 1000            Group: 1                Auto Answer: none
 Console Type: principal          TN: 1                Data Module? y
         Port: 01C1106           COR: 1          Disp Client Redir? n
Security Code:                   COS: 1           Display Language: english
                                                 H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote            Local Remote             Local Remote
 1: 9                   5:                       9:
 2: 82                  6:                      10:
 3:                     7:                      11:
 4:                     8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

Attendant Console x

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Auto Answer

Valid entries	Usage
all	Entering **all** indicates an incoming call to an idle attendant will be answered automatically without any action (no button presses required) by the attendant.
acd	Entering **acd** indicates only ACD split/skill calls and direct agent calls can auto answer. Non-ACD calls terminated to an attendant console with Auto Answer set to **acd** ring audibly.
none	Entering **none** causes all calls terminated to this attendant console to receive some sort of audible ringing treatment.

Console Type

Enter this console's intended use. There can only be one **night-only** or one **day/night** console in the system unless Tenant Partitioning is administered. Night Service is activated from the principal console or from the one station set per-system that has a **nite-serv** button.

Valid entries	Usage
principal	Puts the attendant console into night service.
day-only	Will not get night service calls.
night-only	Handles only night service calls.
day/night	Handles day or night service calls.

COR

Valid entries	Usage
0 through **95**	Enter the class of restriction that reflects the desired restriction.

COS

Valid entries	Usage
0 through **15**	Enter the class of service (COS) for this attendant console.

Data Module

Valid entries	Usage
y/n	Enter **y** if the console is to be connected to a data terminal via 7400B or 8400 Data Module. If **y** is entered, complete the **Data Module** screen (page 4).

Disp Client Redir

This field is administrable only if the Hospitality feature has been enabled on the **System Parameters Customer-Options (Optional Features)** screen. This field affects the station's display on calls originated from a station with Client Room Class of Service.

Valid entries	Usage
y	When the field is **y**, the redirection information for a call originating from a Client Room and terminating to this station displays. **Note:** For stations with an **audix** station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, this field must be **y**.
n	When the field is **n**, then for all calls originating from a Client Room (even redirected calls) that terminate to this station, this station's display will not show the redirection information. Only the client name and extension (or room, depending on what is administered on the **Hospitality** screen) displays.

Display Language

Enter the language in which you want console messages displayed.

Valid entries	Usage
English	Enter the language in which you want messages to be displayed.
French	

Valid entries	Usage
Italian	
Spanish	
user-defined	
Unicode	Unicode display is only available for Unicode-supported telephones. Currently, the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones support Unicode display. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.

Extension (Optional)

Enter the extension for the individual attendant console. Individual attendant extensions allow attendants to use features that an attendant group cannot use. For example, extensions can be members of a DDC or UCD group. An individual attendant extension can have its own Class of Restriction and Class of Service.

If you give your attendants an individual extension, users can call the attendant by dialing the extension or you can assign them an abbreviated-dialing button for fast access to the attendant.

Valid entries	Usage
An unassigned extension or blank	If an extension is not assigned, the attendant can only be addressed as a member of the attendant group. If the attendant has a data module, the **Extension** field cannot be blank.

Group

Valid entries	Usage
1 to **128**	Enter the Attendant Group number.

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can handle only a limited number of conversion calls, you might need to limit the number of telephones with H.320 conversion.

Valid entries	Usage
y/n	Enter **y** for H.320 compliant calls.

Name

Enter the name of this console.

Valid entries	Usage
Up to 27 alphanumeric characters	Any entry is accepted. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Port

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number.
1 through **80** (DEFINITY CSI) or **1** through **250** (S87XX/S8300 Servers)	Gateway
V1 through **V9**	Module
	1 of 2

Valid entries	Usage
01 through **31**	Circuit
ip	SoftConsole IP attendant. You also must have the **Type** field as **302B** and enter a security code. ip is allowed only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **IP Attendant Consoles** field is **y**.
x	Indicates that there is no hardware associated with the port assignment. An individual attendant extension must be assigned in the **Extension** field.
	2 of 2

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Each attendant console requires a port on a digital line circuit pack. For reliability, the attendant consoles should not be assigned to ports on the same digital line circuit pack. For example, if three attendant consoles are to be provided, assign each console to a port on three different digital line circuit pack, if possible. However, if required, all attendant consoles can be assigned to ports on the same digital line circuit pack.

Security Code

Does not apply to S87XX Series IP-PNC. Enter the security code required by the SoftConsole IP attendant. The required security code length is determined by **Minimum Security Code Length** on the **Feature-Related System Parameters** screen.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Type

Valid entries	Usage
console	Indicates the type of attendant console being administered.
302	Use for 302B/C/D or SoftConsole IP attendant.

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)

Enter the trunk access codes (TACs) for local and remote servers. (There are fields for one local TAC and one remote TAC per button labeled **Local** and **Remote**.) The local TAC (1 to 4 digits) refers to a trunk group or Loudspeaker Paging zone on this server. Remote TACs are only useful in a private network (including DCS) network. The remote TAC (1 to 3 digits) refers to a trunk group on the remote server. If a remote TAC is given, then the local TAC must see a trunk group that connects directly to the remote server running Avaya Communication Manager and is also limited to 1 to 3 digits.

Avaya recommends a DCS trunk be specified as the local TAC between the local and remote servers. If the TAC specified as local between the local and remote servers is not a DCS trunk, the remote trunk cannot be monitored by the local server running Avaya Communication Manager.

Valid entries	Usage
1 to 4 digit number	Enter the trunk access codes (TACs) for local and remote servers.
***** or **#**	Can be used as first digit

HUNDREDS SELECT BUTTON ASSIGNMENTS

Enter in the appropriate field (1 through 20), the hundreds group to be associated with a **Hundreds Group Select** button located on an optional selector console.

Valid entries	Usage
1 to 5 digit hundreds group (plus prefix, if needed)	Fields 1 through 8 are used when the selector console is a 24A-type console and fields 1 through 20 are used for a 26A-type console. Enter a hundreds group number that represents all but the last two digits of an extension number (for example, the **Hundreds Select** Button — on the selector console — for extension 3822 would be "38").

Field descriptions for page 2

Figure 277: Attendant Console screen (page 2)

```
change attendant 1                                             Page   2 of   4
                               ATTENDANT CONSOLE

VIS FEATURE OPTIONS

              Auto Start? y
      Echo Digits Dialed? y
```

VIS FEATURE OPTIONS

Use these fields to administer Visually Impaired Service option.

Auto Start

Valid entries	Usage
y/n	Enter **y** to allow an attendant to press any key on the keypad to start a call without the need to first press the **Start** button.

Echo Digits Dialed

Valid entries	Usage
y/n	Enter **y** to provide voiced confirmation of dialed digits.

Field descriptions for page 2 (SoftConsole IP Attendant)

Figure 278: Attendant Console Data Module screen (page 2)

```
change attendant n                                              Page 2 of x
                                   ATTENDANT
IP FEATURE OPTIONS


Remote Softphone Emergency Calls: as-on-local  Direct IP-IP Audio Connections?y
Emergency Location Ext: 1001      Always use? n          IP Audio Hairpinning?n
```

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: ● The **Remote Softphone Emergency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used. The softphone's user-entered settings are ignored. ● If an IP telephone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used.
n	For more information, see the description for the **Emergency Location Extension** field on the **Station** screen. This is the default.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension is the starting point for identifying the street address or nearby location when an emergency call is made. The entry in this field is manipulated by CAMA Numbering Format before being sent over CAMA trunks; or similarly by Numbering — Public/Unknown Format before being sent over ISDN trunks. For more information about this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field on the next page.

Valid entries	Usage
1 to 8 digits	Enter the Emergency Location Extension for the SoftConsole IP Attendant.

IP Audio Hairpinning

Allows IP endpoints to be connected through the server's IP circuit pack.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack for Avaya Communication Manager in IP format, without going through the Communication Manager TDM bus. Default is **n**.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when the **IP Softphone** field is set to **y** on the **Station** screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya representative if you have questions about emergency calls from IP telephones.

Valid entries	Usage
as-on-local	Type **as-on-local** to achieve the following results: ● If the administrator populates the **IP Address Mapping** screen with emergency numbers, the value **as-on-local** functions as follows: ● If the **Emergency Location Extension** field in the **Attendant Console** screen is the same as the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP). ● If the **Emergency Location Extension** field in the **Attendant Console** screen is different from the **Emergency Location Extension** field in the **IP Address Mapping** screen, the value **as-on-local** sends the extension in the **IP Address Mapping** screen to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the Avaya S8XXX Server than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the Avaya S8XXX Server that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the Avaya S8XXX Server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the Avaya S8XXX Server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.

Field descriptions for Attendant Console Data Module screen

This page displays as page 3 if the **Data Module** field on Page 1 is **y**.

Figure 279: Attendant Console Data Module screen

```
change attendant n                                                  Page 3 of x
ATTENDANT
DATA MODULE
     Data Extension: ____         Name: _______________   BCC: 2
                                   COS: 1_
                                   COR: 1_
                ITC: restricted     TN: 1_

ABBREVIATED DIALING
 List1: ________

SPECIAL DIALING OPTION:


ASSIGNED MEMBER (Station with a data extension button for this data module)
        Ext      Name
     1:
```

DATA MODULE

BCC

A display-only field that appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Note:

The **BCC** value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

COR

Valid entries	Usage
0 to **995**	Enter the desired class of restriction (COR) number.

COS

Valid entries	Usage
0 to **15**	Enter the desired (COS) number to designate allowed features. See Class of Service on page 852 for additional information on the allowed features.

Data Extension

Enter the extension number assigned to the data module.

Valid entries	Usage
1 to 5-digit number	Must agree with the system's dial plan

ITC

Valid entries	Usage
restricted	
unrestricted	

Name

Enter the name of the user associated with the data module. The name is optional; it can be left blank.

Note:
Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partitioning number.

ABBREVIATED DIALING

List1

Valid entries	Usage
s	System
g	Group. If **g** is entered, a group number is also required.
p	Personal. If **p** is entered, a personal list number also is required.
e	Enhanced

SPECIAL DIALING OPTION

Valid entries	Usage
hot-line	Enter one of the dialing options that are available. This identifies the destination of all calls when this data module originates calls.
default	

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Only displays when the **Special Dialing Option** field is **hot-line** or **default** (S87XX Series IP-PNC only). The associated AD number is dialed when the user goes off-hook on a Data Hot Line call.

Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

DEFAULT DIALING Abbreviated Dialing Dial Code

The associated AD number is dialed when the user goes off-hook and enters a carriage return following the "DIAL" prompt. The data call originator also can perform data terminal dialing by specifying a dial string that might or might not contain alphanumeric names. Only displays when the **Special Dialing Option** field is **default**.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

Ext

This is the extension number of a previously administered user who has an associated **Data Extension** button and who will share the use of the module.

Name

Contains the name assigned to the above extension number.

Field descriptions for page 3

Figure 280: Attendant Console screen

```
change attendant n                                                    Page 3 of x
                                ATTENDANT CONSOLE
FEATURE BUTTON ASSIGNMENTS
  1: split_____                          13: __________
  2: __________                          14: __________
  3: __________                          15: __________
  4: __________                          16: __________
  5: __________                          17: __________
  6: hold _____  *                       18: __________
  7: __________                          19: forced-rel
  8: aux-work      RC:    Grp:           20: __________
  9: __________                          21: __________
 10: __________                          22: __________
 11: __________                          23: night-serv *
 12: __________                          24: pos-busy__ *
```

If this is a non-IP attendant console this is page 3 of the **Attendant Console** screen.

FEATURE BUTTON ASSIGNMENTS

Enter the feature buttons from Attendant Console Feature Buttons on page 186 that you want to assign to the attendant console. The fixed buttons that cannot be changed (that is, split and forced release) are shown on the screen. The **hold**, **night-serv**, and **pos-busy** buttons are shown in the system default locations. These buttons can be administered elsewhere on the screen. The following table provides descriptions of feature buttons that are unique to the attendant console. See the Attendant Console Feature Buttons and the Telephone Feature Buttons Table sections for more information about the buttons.

Valid entries	Usage
Audible Tones On/Off	
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).
	1 of 4

Valid entries	Usage
alt-frl	Alternate FRL. Alternate facility restriction level; allows the attendant to activate or deactivate the AFRL feature. When activated, this allows the originating device (lines or trunks) to use an alternate set of the facility restriction levels to originate a call (1 per console).
Attendant Control of Trunk Group Access	
act-tr-grp	Activate trunk group access; allows the attendant to control a trunk group. All calls going to the trunks are routed to the attendant (1 per console).
deact-tr-g	Deactivate trunk group access; allows the attendant to release control of a trunk group (1 per console).
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (1 per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there are one or more calls on the emergency attendant queue (1 per console).
hold	Hold. When the **Hold** button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the **call type** button associated with the loop is lit (1 per console).
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the "Pos Avail" light is turned off, and the light associated with the **pos-busy** button is lit. Pushing the **pos-busy** button a second time takes the console out of "position busy" mode, turns on the "Pos Avail" light and turns off the light associated with the **pos-busy** button. If the **pos-busy** button is administered on a 2-LED button, the top LED flashes when the last attendant goes into "Position Busy" mode. Otherwise, if the button has only one LED, the single LED associated with the **pos-busy** button flashes (1 per console).
serial-cal	Serial Call. This button allows the attendant-extended calls to return to the same attendant if the trunk remains off-hook (1 per console).
override	Attendant Override. This button enables the attendant to override diversion features such as, Call Forwarding, Call Coverage, and so on (1 per console).
intrusion	Call Offer. Depression of this button allows the attendant to extend a call when the called party is active on another call (1 per console).
dont-split	Don’t Split. This button allows the attendant to not split away a call when dialing (1 per console).
	2 of 4

Valid entries	Usage
vis	Visually Impaired Attendant Service (**vis**) — This button activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: ● "con-stat" repeats the console status. ● "display" calls out display contents. ● "dtgs-stat" calls out the DTGS status. ● "last-mess" repeats the last message. ● "last-op" calls out the last operation.
Trunk Group Select — In addition to the 12 Direct Trunk Group Selection (DTGS) Button Assignments on Field descriptions for page 1, up to 12 single lamp **DTGS** buttons can be administered on this page. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using **Attendant Control of Trunk Group Select** buttons. The single lamp **DTGS** buttons can be administered as follows:	
local-tgs	Local trunk group select; allows the attendant to access trunk groups on the local server running Avaya Communication Manager (combination of 12 local-tgs/remote-tgs per console).
remote-tgs	Remote trunk group select; allows the attendant to access trunk groups on a remote server running Avaya Communication Manager (combination of 12 local-tgs/remote-tgs per console).
hundrd-sel	Hundreds group select; in addition to the fixed **HGS** buttons on Field descriptions for page 1, a user can administer hundreds group select feature buttons on this page. When a feature button is administered as "hundrd-sel," a subfield appears that must then be administered in the same manner as the fixed **HGS** button fields (a 1 to 3 digit hundreds group plus prefix, if needed). Administered **hundrd-sel** feature buttons operate in the same manner as fixed **HGS** buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Thus, if all 20 fixed **HGS** buttons have been administered, no **hundrd-sel** feature buttons can be administered. **Note:** If no fixed **HGS** buttons are administered, 19 **hundrd-sel** feature buttons are available. This is because 5 of the 24 feature buttons must be used for required feature buttons (**hold**, **pos-busy**, **night-serv**, **forced-rel**, and **split**)
group-disp	Group Display. Allows the attendant to see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. Allows the attendant to select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.
	3 of 4

Valid entries	Usage
Attendant Room Status	
occ-rooms	Occupied rooms; allows the attendant to see which rooms are occupied.
maid-stat	Maid status; allows the attendant to see which rooms are in one of six specified states.
vu-display	VuStats (**vu-display**) — This button allows users with display telephones and attendants to turn on the VuStats display. The limit to the number of VuStats feature buttons you can administer depends on how many feature buttons are available on the attendant console you are administering. The system is designed to allow you to set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.
	4 of 4

- If 12 HGS buttons are assigned on field descriptions for page 2, Avaya recommends that the **night**, **pos-busy**, and **hold** buttons be reassigned to locations 20, 21, and 3, respectively. The **HGS** buttons should then be assigned to the right-most three columns, as required.

Field descriptions for page 4

Figure 281: Attendant Console screen

```
change attendant n                                                Page 4 of x
                                  ATTENDANT CONSOLE
DISPLAY BUTTON ASSIGNMENTS
  1: normal____                             5: delete-msg
  2: inspect___                             6: call-disp_
  3: cov-msg-rt                             7: date-time_
  4: next______                             8: timer_____
```

DISPLAY MODULE BUTTON ASSIGNMENTS

Display-type buttons obtain display functions on the associated alphanumeric display. These buttons are noted as [display button] in the **Feature** or **Function** column on the table. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is not operational. These buttons are noted as [status lamp]. If a Call Cover Msg Rt (**cov-msg-rt**) button is assigned, a Leave Word Calling Delete Msg (**delete-msg**) button and a Next (**next**) button must also be assigned.

Audio Group

Use the **Audio Group** screen to add, change, or display a specified audio group. An audio group is a collection of recorded audio sources that have been placed in a group to facilitate their selection. The three pages of this screen provide for administering up to 260 audio source locations for an audio group.

Field descriptions for page 1

Figure 282: Audio Group screen

```
add audio-group next                                          Page 1 of x
                              Audio Group 2

                            Group Name:

AUDIO SOURCE LOCATION
     1:         16:         31:         46:         61:         76:
     2:         17:         32:         47:         62:         77:
     3:         18:         33:         48:         63:         78:
     4:         19:         34:         49:         64:         79:
     5:         29:         35:         50:         65:         80:
     6:         21:         36:         51:         66:         81:
     7:         22:         37:         52:         67:         82:
     8:         23:         38:         53:         68:         83:
     9:         24:         39:         54:         69:         84:
    10:         25:         40:         55:         70:         85:
    11:         26:         41:         56:         71:         86:
    12:         27:         42:         57:         72:         87:
    13:         28:         43:         58:         73:         88:
    14:         29:         44:         59:         74:         89:
    15:         30:         45:         60:         75:         90:
```

Audio Source Location

Enter the board location for this audio group: cabinet(1-64):carrier(A-E):slot(1-20):OR gateway(1-250):module(V1-V9).

Group Name

Enter an alpha-numeric name of the audio group for identification.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Audix-MSA Node Names

Field descriptions for page 1

Figure 283: Audix-MSA Node Names screen

```
change node-names audix-msa                                   Page 1 of x

                         AUDIX-MSA NODE NAMES

 Audix Name         IP Address     MSA Names       IP Address
 audixA_            __.__.__.__    _______         __.__.__.__
 audixB_            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
 _______            __.__.__.__    _______         __.__.__.__
```

Audix Names

Identifies the name of the AUDIX node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The node names must be unique on each server running Avaya Communication Manager.

IP Address

The IP address associated with the node name.

MSA Names

Identifies the name of the MSA node.

Valid entries	Usage
1 to 7 character string	Used as a label for the associated IP address. The MSA names must be unique on each server running Avaya Communication Manager.

Authorization Code — COR Mapping

You use this screen to assign authorization codes and the class of restriction (COR) that is associated with a given authorization code. See "Authorization Codes" and "Class of Restriction" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*,* for more information on how Authorization Codes work with COR.

To maximize the security of your system:

- Administer authorization codes to the maximum length allowed by the system
- Create random (nonconsecutive) authorization codes
- Change authorization codes at least quarterly
- Deactivate authorization codes immediately if a user leaves the company or changes assignments
- Assign each authorization code the minimum level of calling permissions required

Field descriptions for page 1

Figure 284: Authorization Code - COR Mapping screen

```
change authorization-code                                          Page 1 of x
                       Authorization Code - COR Mapping
     Note:    XX codes administered.  Use "list" to display all codes.
     AC     COR    AC     COR    AC     COR    AC     COR    AC      COR    AC     COR
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
```

AC

Valid entries	Usage
Any combination of between 4 and 13 digits	The number of digits must agree with the number assigned to the **Authorization Code Length** field on the **Feature-Related System Parameters** screen. To enhance system security, choose Authorization Codes of 13 random digits.

COR

Valid entries	Usage
0 to **995**	When a user dials the associated authorization code, this is the COR that the telephone or other facility will assume for that call.

Number of Codes Administered

Displays the number of Authorization Codes already administered using the **Authorization Codes** screen. There is a maximum number of authorization codes that you can use. To find out what this number is for your system, type **`display capacity`**, and page down to find the authorization code information. For details on the **System Capacity** screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Best Service Routing

This screen administers the routing numbers for each location associated with a particular application. This allows the Avaya DEFINITY server or Avaya S8XXX Server to compare specified skills, identify the skill that will provide the best service to a call, and deliver the call to that resource.

For information on setting up Best Service Routing (BSR), see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Field descriptions for page 1

Figure 285: Best Service Routing screen

```
change best-service-routing n                                  Page 1 of x

                            BEST SERVICE ROUTING

Number: 1     Name: ARS              Maximum Suppression Time: 30     Lock? n

Num   Location Name  Switch Node     Status Poll VDN   Interflow VDN  Net Redir?
1     st10     auto                  95022011          3035552121         y
4     st10     auto                  95022014          3035551110         n
```

Interflow VDN

Valid entries	Usage
0 to **9**, *, #, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) blank (DEFINITY CSI)	When a given remote Avaya server is the best available, the origin Avaya server interflows the call to this vector on the remote server. Each remote Avaya server in a given application has to have a dedicated interflow server.

Location Name

Indicates the location.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY CSI)	Enter a name for the location.

Lock

Indicates whether this application is locked.

Valid entries	Usage
y/n (DEFINITY CSI)	Set to **y** to prevent this application from being sent to Call Management System (CMS).

Maximum Suppression Time

Prevents callers from connecting to a VDN within a certain time period after receiving a busy signal.

Valid entries	Usage
0 to **60** (DEFINITY CSI)	Enter time in seconds.

Name

Contains the name assigned to the BSR number.

Valid entries	Usage
Up to 15 alphanumeric characters. (DEFINITY CSI)	Assign a descriptive name for the physical location. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Net Redir

Valid entries	Usage
y/n (DEFINITY CSI)	Set to **y** for each to location to which calls are to be redirected using Network Call Redirection.

Num

This field corresponds to the "consider location x" step from the **Call Vector** screen.

Valid entries	Usage
1 to **255** (DEFINITY CSI)	Enter the number.

Number

This display-only field corresponds to the **BSR Application** field on the **Vector Directory Number** screen.

Status Poll VDN

This field specifies the AAR or ARS pattern that routes over an IP trunk. The status poll vector on the remote Avaya server compares resources on that server and replies to the origin server with information on the best of these. Each remote Avaya server in a given application has to have a dedicated status poll vector.

Valid entries	Usage
0 to **9**, *****, **#**, **~p** (pause) **~w/~W** (wait) **~m** (mark) **~s** (suppress) or blank (DEFINITY CSI)	Specify the AAR or ARS pattern that routes over an IP trunk

Switch Node

Enter a number unique to the switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank (DEFINITY CSI)	This number is an important part of the UCID tag and must be unique to the server running Avaya Communication Manager.

Bulletin Board

Use the bulletin board to post and receive information. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11 to 19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

You can enter up to 40 characters of text per line. You can also enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The TAB key moves the cursor to the next line.

Field descriptions for page 1

Figure 286: Bulletin Board screen

```
change bulletin-board                                           Page 1 of x

              Message (* indicates high-priority)                      Date
          *Avaya is in the process of                                03/02/93
          *investigating your trunk lockup problem.                  03/02/93
          *The Bulletin Board will be updated as                     03/02/93
          *we find information.                                      03/02/93
          *  We have identified the problem.                         03/04/93
          *The trunk you added does not provide                      03/04/93
          *disconnect supervision. However, the                      03/04/93
          *trunk group was administered as such.                     03/04/93
          *Please call Pat J. for details.                           03/04/93
          We recently added a new trunk group (14)                   03/02/93
          and have had many of the members getting                   03/02/93
          locked up.                                                 03/02/93
          We see the error - thanks for checking.                    03/05/93
```

Date

This display-only field contains the date the information was entered or last changed.

Lines 1 through 10

These lines are reserved for high priority messages and are noted with an asterisk (*) in the first column on the left. If you have an *init* or *inads* login you can enter high-priority information to trigger the high-priority message at login time.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\’˜;:’,"<.>/?	Enter any information.

Lines 11 through 19

These lines can be used by anyone with access.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'˜;:’,"<.>/?	Enter any information.

Field descriptions for pages 2 and 3

Date

This display only field contains the date the information was entered or last changed.

Lines 1 through 20

These lines can be used by anyone with access.

Valid entries	Usage
A to **Z** **a** to **z** Blank **0** to **9** !@#$%^&*()_+-=[]{}|\'˜;:’,"<.>/?	Enter any information.

Button Type Customization Restrictions

Use this screen to restrict button label customization of up to 50 specified button types for users who are not considered to be VIP users. This helps you to manage the usage of your system's allocation of customized button labels to ensure that VIP users have the button label customization resource available to them.

Field descriptions for page 1

Figure 287: Button Type Customization Restrictions screen

```
 change button-restriction                                    Page 1 of x

                        Button Type Customization Restrictions

Restrict Customization Of Button Types? y

Restrict Customization Of Labels For The Following Button Types:

   abr-spchar   ~p ____________   ____________   ____________   ____________
   abrv-dial    s  ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________
   ____________    ____________   ____________   ____________   ____________

Caution: Adding new button types to this list will cause all button labels
         associated with that button type to be lost
```

Restrict Customization Of Button Types

Use this field to enable/disable restriction of feature button label customization.

Valid entries	Usage
y	Enter **y** to enable restriction of feature button label customization. This is the default.
n	When you enter **n** in this field, users have the capability to customize labels for all buttons on their telephones.

Restrict Customization Of Labels For The Following Button Types

This field appears when **Restrict Customization of Button Types** is **y**.

Valid entries	Usage
valid button type from the list of entries	Enter the button type you want to restrict from label customization. **Note:** When you enter the special button types **abr-spchar** or **abrv-dial**, an additional field appears to the right of the button type as shown in Figure 287. Use this special field to specify the special character associated with the **abr-spchar** button type or the Abbreviated Dialing List associated with the **abrv-dial** button type.

Call Type Digit Analysis Table

Use the **Call Type Digit Analysis Table** `(change calltype analysis)` to tell Communication Manager how to modify telephone numbers dialed from a telephone's call log from internal contacts, or a corporate directory. There must be at least one entry in the **Call Type Digit Analysis Table** for Call Type Digit Analysis to take place. Call Type Digit Analysis allows users to automatically place outgoing calls based on the telephone number information in the phone's call log, without the user having to modify the telephone number.

Field descriptions for page 1

Figure 288: Call Type Digit Analysis Table screen

```
change calltype analysis                                           Page 1 of x
                         CALL TYPE DIGIT ANALYSIS TABLE
                                      Location:   all
    Dialed String               Delete Insert     Type   Delete Insert       Type
Match: _____________________ 1: ___   __________       : ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
Match: _____________________ 1: ___   __________      2: ___   ___________
Length:Min    Max       __   3: ___    _________      4: ___    ____________
```

Location

This field is display-only. Its value is copied from the location specified in the command line, or if no location is entered, displays **all**.

Valid entries	Usage
numeric value **all**	Phones dialing from this location use the entries on this form. If there are matching entries in the telephone's location, those entries are used. If there are no matching entries in the phone's location, Communication Manager tries the entries in location **all**.

Dialed String Match

Valid entries	Usage
numeric value **x** **X** **Blank**	Communication Manager compares this digit string to the original digit string, looking for a match to complete analysis and routing. **x**, **X**= wildcard digits. Use to match anything, as in ARS analysis administration. Blank=default. Cannot be blank if other fields on the row pair contain data.

Dialed String length (Min, Max)

Valid entries	Usage
numeric value	Communication Manager compares digit strings of this length to the original digit string, looking for a match to complete analysis and routing.

Delete

Valid entries	Usage
numeric value	Communication Manager deletes this number of digits in the original digit string, from the left-hand side of the original digit string, to complete analysis and routing.

Insert

Valid entries	Usage
numeric value	Communication Manager inserts these digits into the left-hand side of the original digit string to complete analysis and routing.

Type

Valid entries	Usage
aar **ars** **ext** **udp**	Administered call type for this dialed string. Communication Manager tests the modified digit string against the administered call type. **aar** = Automatic Alternate Routing, digit analysis algorithm commonly used for private network calls. **ars** = Automatic Route Selection, digit analysis algorithm commonly used for public network calls. **ext** = extension entries in the dialplan analysis tables of type **ext**. **udp** = extension entries in the uniform-dialplan tables.

Call Vector

This screen programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN). See *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for additional information.

Note:

If the **Call Center Release** field is set to **4.0** or later, the **Call Vector** screen includes additional pages to support up to 99 vector steps.

Field descriptions for page 1

Figure 289: Call Vector screen

```
change vector nnnn                                            Page 1 of x
                               CALL VECTOR
  Number: nnnn           Name: _______________

Multimedia? n     Attendant Vectoring? n     Meet-me Conf? y           Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n      Holidays? n

 01 ____________
 02 ____________
 03 ____________
 04 ____________
 05 ____________
 06 ____________
 07 ____________
 08 ____________
 09 ____________
 10 ____________
 11 ____________

                   Press Esc f6 for Vector Editing
```

01 through XX

Enter vector commands as required (up to the maximum allowed in your configuration). For more information, see *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Valid entries	Usage
adjunct routing	Causes a message to be sent to an adjunct requesting routing instructions based on the CTI link number.
announcement	Provides the caller with a recorded announcement.
busy	Gives the caller a busy signal and causes termination of vector processing.
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).
collect	Allows the user to enter up to 16 digits from a touch-tone telephone, or allows the vector to retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the server running Avaya Communication Manager. Also allows the optional assignment of an announcement that will play immediately before the disconnect.
goto	Allows conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	Allows the caller to leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it "returns" best data for its location to the primary vector on the origin server.
return	Returns vector processing to the step following the **goto** command after a subroutine call has processed.
	1 of 2

Valid entries	Usage
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number).
set	Performs arithmetic and string operations and assigns values to a vector variable or to the digits buffer during vector processing.
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command's syntax. Also provides feedback (in the form of silence, ringback, or music) to the caller while the call advances in queue.
	2 of 2

ANI/II-Digits

A display-only field indicating whether you can use ANI and II-Digits Vector Routing Commands. ANI/II-Digits Routing requires that the **G3V4 Enhanced** field be **y**.

ASAI Routing

A display-only field indicating whether, on the **System Parameters Customer-Options (Optional Features)** screen, the **CallVisor Adjunct/Switch Applications Interface (ASAI Link Core Capabilities)** field is **y**.

Attendant Vectoring

This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Attendant Vectoring** field is **y**. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. Use this field to indicate attendant vectoring. If **Basic Vectoring** and **Vector Prompting** are both set to **n**, then the **Attendant Vectoring** field defaults to **y** and no changes are allowed to the field. When attendant vectoring is indicated for VDNs and vectors, all call center-associated fields (such as Skills and BSR) are removed.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector.
n	Default.

Basic

A display-only field indicating whether, on the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring (Basic)** field is **y**.

BSR

A display-only field indicating that on the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring (Best Service Routing)** field is **y**. Thus, you can use BSR commands and command elements in your vectors. An **n** indicates that the BSR option is not enabled.

CINFO

A display-only field indicating whether, on the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring (CINFO)** field is **y**.

EAS

A display-only field indicating whether, on the **System Parameters Customer-Options (Optional Features)** screen, the **Expert Agent Selection (EAS)** field is **y**.

Note:

When **Expert Agent Selection (EAS)** field is **y**, the help messages and error messages associated with this screen will reflect a terminology change from "Split" to "Skill." In addition, the vector commands entered also will be affected by this terminology change (for example, *check backup split* becomes *check backup skill* when EAS is enabled).

G3V4 Adv Route

A display-only field indicating whether you can use the G3V4 Advanced Vector Routing commands.

G3V4 Enhanced

A display-only field indicating whether you can use G3V4 Enhanced Vector Routing commands and features.

Holidays

A display-only field that appears when, on the screen, the **Vectoring (Holidays)** field is **y**.

LAI

A display-only field indicating whether **Look-Ahead Interflow** is enabled.

Lock

This field controls access to the vector from Avaya CentreVu products.

Note:
Always lock vectors that contain secure information (for example, access codes).

Valid entries	Usage
y	You do not want this vector to be accessible to these client programs. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**.
n	Gives CentreVu CMS and CentreVu Control Center users the ability to administer this vector from these client programs.

Meet-me Conf

This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Enhanced Conferencing** field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this vector. If **Meet-me Conference** is **y**, the **Lock** field also must be **y**. When the **Lock** field is **y**, the vector cannot be changed by adjunct vectoring programs such as Visual Vectors. **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time.

Multimedia

Indicates whether the vector should receive early answer treatment for multimedia calls. This only applies if the **Multimedia Call Handling** field is **y**. This field does not appear for S87XX Series IP-PNC.

Valid entries	Usage
y/n	If you expect this vector to receive multimedia calls, set this field to **y**. If this value is **y**, the call is considered to be answered at the start of vector processing, and billing for the call starts at that time. See Managing Multimedia Calling on page 335 for more information.

Name

Represents the vector name.

Valid entries	Usage
Up to 27 alphanumeric characters. Up to 15 alphanumeric characters (for S8300, S8400, S87XX IP-PNC Servers only)	This is an optional field. If ~r can be used to activate Network Call Redirection if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN Network Call Redirection** field is **y**. **Note:** The **Name** field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones. For more information on Unicode language display, see Administering Unicode display on page 203. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Number

Represents the vector number. A display-only field when the screen is accessed using a **change** or **display** administration command.

Prompting

A display-only field indicating whether, on the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring (Prompting)** field is **y**.

CAMA Numbering Format

This screen administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

This screen provides the CESID format by extension number or number blocks. This allows for multiple CESID formats to be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

The default CESID defines the CESID for all extensions that are not defined in the **Ext Code** field.

There are 446 CESID entries over 15 pages. The first page contains the Default CESID and 26 extensions to CESID entries. The second through fifteenth pages each contain 30 extensions to CESID entries.

Field descriptions for page 1

Figure 290: CAMA Numbering Format screen

```
change cama-numbering                                          Page 1 of x
                          CAMA NUMBERING - E911 FORMAT

System CESID Default: _____________

Ext Ext                              Total
Len Code            CESID            Length
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
__  _______________ ________________ __
```

CESID

Valid entries	Usage
1 to 11 digits or blank 1 to 16 digits or blank (S8300, S87XX IP-PNC Servers)	Enter the number that will be used to identify the calling terminal within an emergency service system. This field can represent a prefix to an extension or the entire CESID.

Ext Code

Valid entries	Usage
Leading extension digits or blank	Enter the leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code will be interpreted as a block of digits. For example, if the extension length is 4 and the extension code is 11, the CESID will serve extensions 1100 through 1199. The Ext Code 11 is for a DID block. An Ext Code of 126 might point a non-DID block to a nearby DID extension 5241666.

Ext Len

Valid entries	Usage
1 to **13** or blank	Enter the number of digits in the extension.

System CESID Default

Valid entries	Usage
1 to 16 digits	Enter a default CESID. This number will be sent over the CAMA trunk if the **Ext Code** field does not have an entry.

Total Length

Valid entries	Usage
1 to **16** or blank	Enter the total number of digits to send.

Capacities

The **System Capacity** screen (command **`display capacity`**) is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431. Detailed system capacity information can be found in *System Capacities Table for Avaya Communication Manager on Avaya Media Servers*, 03-300511.

CDR System Parameters

Use the **Call Detail Recording (CDR) System Parameters** screen to set parameters for the types of calls you want to record and how to format the information. You can use CDR records to determine call costs, diagnose problems, detect abuse, and optimize your network.

Field descriptions for page 1

Figure 291: CDR System Parameters screen

```
change system-parameters cdr                                    Page   1 of   x
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                     CDR Date Format: month/day
      Primary Output Format: printer       Primary Output Endpoint: CDR1
    Secondary Output Format:
           Use ISDN Layouts? n                   Enable CDR Storage on Disk? n
       Use Enhanced Formats? n      Condition Code 'T' for Redirected Calls? n
     Use Legacy CDR Formats? y                  Remove # From Called Number? n
Modified Circuit ID Display? n                             Intra-switch CDR? n
                  Record Outgoing Calls Only? y     Outg Trk Call Splitting? n
  Suppress CDR for Ineffective Call Attempts? y          Outg Attd Call Rec? y
      Disconnect Information in Place of FRL? n      Interworking Feat-flag? n
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
 Record Agent ID on Incoming? n      Record Agent ID on Outgoing? n
      Inc Trk Call Splitting? n
   Record Non-Call-Assoc TSC? n
       Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
    Privacy - Digits to Hide: 0             CDR Account Code Length: 4
```

Calls to Hunt Group — Record

Valid entries	Usage
member-ext	Enter **member-ext** to record the extension of the telephone or data terminal where the call terminated.
group-ext	Enter **group-ext** to record the extension that was dialed.

CDR Account Code Length

Valid entries	Usage
1 to **15**	Enter the number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

CDR Date Format

Use this field to select the format for the date stamp that begins each new day of call records.

Valid entries	Usage
month/day **day/month**	Choose the format that is most appropriate for your situation. If your company has many different sites, you might need to use the same format as the other locations.

Condition Code 'T' for Redirected Calls

You can elect to identify CDR records of calls that have been redirected automatically off the server running Avaya Communication Manager.

Valid entries	Usage
y	The Condition Code of both CDR records for the call will be 'T.'
n	The Condition Codes normally associated with the **Record Outgoing Calls Only** field are generated.

Digits to Record for Outgoing Calls

Valid entries	Usage
dialed	Use **dialed** to record the digits a user actually dials.
outpulsed	Use **outpulsed** to record the digits that Communication Manager actually sends out over the trunk, *including any additions or deletions that take place during routing*.

Disconnect Information in Place of FRL

Valid entries	Usage
y	Enter **y** to replace the **Facility Restriction Level (FRL)** field with information about why a call disconnects.
n	Enter **n** to record the call's FRL.

Enable CDR Storage on Disk

Valid entries	Usage
y/n	Enter **y** to enable the Survivable CDR feature. Default is **n**.

Force Entry of Acct Code for Calls Marked on Toll Analysis Form

Specifies whether an account code will be required when making a toll call. This will not necessarily be all chargeable calls and it might even include some non-chargeable calls. See Forcing Users to Enter Account Codes on page 641 for more information.

Valid entries	Usage
y	Enter **y** to deny all toll calls unless the user dials an account code. Forced Entry of Account Codes must be **y** on the **System Parameters Customer-Options (Optional Features)** screen.
n	Enter **y** to allow calls without an account code. *This does not override other calling restrictions.*

Inc Attd Call Record

Appears when **Inc Trk Call Splitting** is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Inc Trk Call Splitting

Appears when the **Record Outgoing Calls Only** field on the **System Parameters CDR** screen is **n**.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of incoming calls that are transferred or conferenced.

Interworking Feat-flag

See "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Valid entries	Usage
y	Enter **y** if you want the feature flag to indicate interworked outgoing ISDN calls. *An interworked call is one that passed through more than one ISDN node.*
n	Enter **n** if you want the feature flag to indicate no answer supervision for interworked calls.

Intra-Switch CDR

Valid entries	Usage
y/n	Enter **y** to record calls within Avaya Communication Manager. If you choose this option, you must complete the **Intra-Switch CDR** screen to indicate which extensions should be monitored.

Modified Circuit ID Display

This affects the "printer," "teleseer," and "59-character" output formats.

Valid entries	Usage
y	Enter **y** to display the circuit ID in its actual format (100’s, 10’s, units). For example, circuit ID 123 displays as 123. *You might need to verify that your output device can accept this format.*
n	Enter **n** to display the circuit ID in its default format (10’s, units, 100’s). For example, circuit ID 123 appears as 231.

Node Number (Local PBX ID)

A display-only field indicating the DCS switch node number in a network of switches.

Outg Attd Call Record

Only appears if **Outg Trk Call Splitting** is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Outg Trk Call Splitting

See "Call Splitting" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of outgoing calls that are transferred or conferenced.

Primary Output Endpoint

This field determines where the server running Avaya Communication Manager sends the CDR records, and is required if you specify a Primary Output Format.

Valid entries	Usage
eia	If you use the EIA port to connect the CDR device, enter **eia**.
Extension number	This is the extension of the data module (if used) that links the primary output device to the server running Avaya Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the **IP Services** screen.

Primary Output Format

Controls the format of the call records sent to the primary output device.

Valid entries	Usage
customized	Use this option if you have special call accounting needs that standard record formats do not accommodate. If you use a customized record format, you need to have call accounting software that is also customized to receive these records. Consult with your call accounting vendor before using this option.
printer	Use **printer** if you are sending the call detail records to a printer rather than to a record collection or call accounting system.
59-char **expanded** **lsu** **lsu-expand** **int-direct** **int-isdn** **int-process** **teleseer** **unformatted**	The remaining formats are standard record formats. The one you use must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Privacy — Digits to Hide

If you enable **CDR Privacy** on the **Station** screen for a given telephone, use this field to indicate how much of the dialed number to hide on the CDR record.

Valid entries	Usage
0 to **7**	Enter the number of digits to hide, counting from the end (right to left). For example, if you enter **4** in this field and the user dials 555-1234, only "555" would appear in the **Dialed Number** field of the CDR record.

Record Agent ID on Incoming

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. You cannot use both the **Called VDN** and the **Agent Login ID Instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Agent ID on Outgoing

Only displays if the **Expert Agent Selection (EAS)** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent's LoginID instead of the physical extension in the **Dialed Number** field of a CDR record.

Record Call-Assoc TSC

Valid entries	Usage
y/n	Enter **y** to create records for call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You might want to consider the capacity of your call collection device*.

Record Called Vector Directory Number Instead of Group or Member

If this option is enabled, the called VDN overrides the group or member information that normally appears in the **Dialed Number** field of the CDR record. If a call is directed through more than one VDN, the first VDN used for the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

You cannot use both the **Called VDN** and the **Agent Login ID instead of Group or Member**. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the Vector Directory Number (VDN) in the **Dialed Number** field of a CDR record.

Record Non-Call-Assoc TSC

A temporary signaling channel (TSC) is a virtual connection established within an ISDN D-channel.

Valid entries	Usage
y/n	Enter **y** to create records for non-call-associated temporary signaling connections. *If you have a lot of data connections this could increase the number of records. You might want to consider the capacity of your record collection device*.

Record Outgoing Calls Only

Valid entries	Usage
y	Enter **y** to record only outgoing calls. *This can save space if you are only concerned with charges for outbound calls.*
n	Enter **n** to record both outgoing and incoming calls.

Remove # From Called Number

Valid entries	Usage
y	Enter **y** to have the "#" (or "E") symbol removed from the **Dialed Number** field of the call detail record. *You might need to verify that your output device can accept this format.*
n	Enter **n** to have the trailing "#" (or "E") symbol appear in the **Dialed Number** field whenever inter-digit time out occurs or users dial # to indicate the end of dialing.

Secondary Output Endpoint

Appears when the **Secondary Output Format** field is administered.

Valid entries	Usage
eia	Use this if the secondary output device is connected to the eia port.
Extension number	This is the extension of the data module (if used) that links the secondary output device to the server running Avaya Communication Manager.
CDR1, CDR2	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the **IP Services** screen.

Secondary Output Format

Controls the format of the call records sent to the secondary output device.

CAUTION:

Only qualified (Avaya) service personnel should administer a secondary output device. This option might cause loss of data when the buffer contains large amounts of data.

Valid entries	Usage
customized **int-direct** **int-process** **lsu** **unformatted**	These are the only formats you can use for a secondary output device. The format must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are calls that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-by-Call Service Selection trunks, incoming calls rejected by Communication Manager due to NSF mismatch, and ISDN calls that did not complete at the far end, if a cause value was provided. These calls appear on the CDR record with a condition code "E."

Valid entries	Usage
y	Enter **y** to ignore ineffective call attempts. *Use this if you have limited storage space for CDR records and records often overrun the buffer.*
n	Enter **n** to report ineffective call attempts. *This can tell you if your users are often unable to place outgoing calls, or if a large number of incoming calls are not completed.* You can also use this if you need to have records of attempts to contact a client, and are using ISDN trunks. Using this option requires more space for records.

Use Enhanced Formats

Enhanced formats provide additional information about time in queue and ISDN call charges, where available. This affects the "expanded," "teleseer," "lsu," "printer," and "unformatted" output formats.

Valid entries	Usage
y/n	Enter **y** to enable the use of the Enhanced version of the specified primary output format. You cannot use Enhanced formats and ISDN formats at the same time.

Use ISDN Layouts

ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects "lsu" and "printer" output formats, as well as any format with ISDN layouts, such as "teleseer."

Valid entries	Usage
y/n	Enter **y** to enable the use of the ISDN version of the specified primary output format. You cannot use ISDN formats and Enhanced formats at the same time.

Use Legacy CDR Formats

Use this field to specify the use of pre-Communication Manager 4.0 (“legacy”) Call Detail Recording (CDR) formats in the CDR records the system produces, instead of the formats used in Communication Manager 4.0 and later. Listed below are the CDR formats that are impacted by the **Use Legacy CDR Formats** field. All other CDR formats remain unchanged.

CDR Format	Communication Manager 3.1 and earlier length	Communication Manager 4.0 and later length
ISDN Teleseer	80	82
Enhanced Teleseer	81	83
ISDN Printer	84	86
Enhanced Printer	85	87
ISDN LSU	59	61
Enhanced LSU	59	61
Expanded	135	139

CDR Format	Communication Manager 3.1 and earlier length	Communication Manager 4.0 and later length
Enhanced Expanded	151	155
Unformatted	105	109
Enhanced Unformatted	119	123
Int-ISDN	136	140

Valid entries	Usage
y	Enter **y** to use pre-Communication Manager 4.0 (“legacy”) CDR formats for CDR records. Default is **y**.
n	Enter **n** to use CDR formats for Communication Manager 4.0 and later. When this field is set to **n**, the **INS** field in the CDR records is increased from three to five characters, and the **Attendant Console** field is increased from two to four characters.

Field descriptions for page 2

This page appears only if the **Primary Output Format** field is **customized**.

Figure 292: CDR System Parameters screen

```
change system-parameters cdr                                   Page 2 of x
                              CDR SYSTEM PARAMETERS
        Data Item - Length          Data Item - Length       Data Item - Length
 1: time____________ - 4_  17: ________________ - __  33: ______________ - __
 2: space___________ - 1_  18: ________________ - __  34: ______________ - __
 3: duration________ - 4_  19: ________________ - __  35: ______________ - __
 4: return__________ - 1_  20: ________________ - __  36: ______________ - __
 5: line-feed_______ - 1_  21: ________________ - __  37: ______________ - __
 6: ________________ - __  22: ________________ - __  38: ______________ - __
 7: ________________ - __  23: ________________ - __  39: ______________ - __
 8: ________________ - __  24: ________________ - __  40: ______________ - __
 9: ________________ - __  25: ________________ - __  41: ______________ - __
10: ________________ - __  26: ________________ - __  42: ______________ - __
11: ________________ - __  27: ________________ - __  43: ______________ - __
12: ________________ - __  28: ________________ - __  44: ______________ - __
13: ________________ - __  29: ________________ - __  45: ______________ - __
14: ________________ - __  30: ________________ - __  46: ______________ - __
15: ________________ - __  31: ________________ - __  47: ______________ - __
16: ________________ - __  32: ________________ - __  48: ______________ - __
                                Record length = 11
```

Data Item

Enter the data items in the order they should appear on the customized record. Only use this screen if you have arranged with your vendor to customize your call accounting system to receive these records.

You must include at least one field in order to have a record. See the table below for valid entries. The last two data items in a the record must be **line-feed** and **return**, in that order.

For more information, see "Call Detail Recording" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205*.*

Data Item	Length	Data Item	Length
acct-code	15	ixc-code	4
attd-console	2	line-feed	1
auth-code	7	location-from	3
bandwidth	2	location-to	3
bcc	1	in-trk-code	4
calling-num	15	ma-uui	1
			1 of 2

Data Item	Length	Data Item	Length
clg-pty-cat	2	node-num	2
clg-num/in-tac	10	null	1
code-dial	4	out-crt-id	3
code-used	4	ppm	5
cond-code	1	res-flag	1
country-from	3	return	1
country-to	3	sec-dur	5
dialed-num	23	space	1
duration	4	time	4
feat-flag	1	timezone-from	3
frl	1	timezone-to	6
in-crt-id	3	tsc_ct	4
ins	3	tsc_flag	1
isdn-cc	11	vdn	5
			2 of 2

Length

Enter the length of each data item, if different from the default.

Valid entries	Usage
The maximum record length depends on the call accounting system you use. Check with your vendor.	The date field should be six-digits to ensure proper output. Certain fields default to the required length.

Record Length

Display-only field indicating the accumulated total length of the customized record, updated each time the length of a data item changes.

Change Station Extension

This screen allows an administrator to change extensions on the switch from one extension to another all at once. When the screen is filled out and submitted, all administration that was associated with the current extension will now be associated with the new extension. Any administration references of the extension being changed, such as references used in a vector, coverage, etc., will now reference the new extension. Once the extension has been changed, all references to the previous extension will be removed from the switch.

If an extension is changed that is also administered on an adjunct (such as voice mail or an ASAI link), the extension on the adjunct must also be changed to ensure proper functionality.

Note:

A forwarded extension administered as a button will not be handled by the **`change extension-station`** ***`xxxxxxx`*** command. It is recommended that the administrator use the **`list usage`** command prior to changing any extensions.

Field descriptions for page 1

Figure 293: Change Station Extension screen

```
change extension-station xxxxxxx                                Page 1 of x

                     CHANGE STATION EXTENSION

           Station Name: xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx     Port: xxxxxx

                                 FROM EXTENSION                  TO EXTENSION
                                 --------------                  -------------
                        Station: xxx-xxxx                         xxx-xxxx
                   Message Lamp: xxx-xxxx                         xxx-xxxx
         Emergency Location Ext.: xxx-xxxx                         xxx-xxxx
IP Parameter Emergency Location: xxx-xxxx                  See IP-Network Map Form
```

Note:

You cannot use the **`change extension-station`** command to change the extension of a station if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- You cannot change the extension of station A using the **`change extension-station`** command unless you first change station B to assign a different emergency location extension.

- You can change the extension of station B. If you do, the **Change Station Extension** screen displays station A's extension in the **Emergency Location Ext.** field under the **From Extension** header.

Emergency Location Extension

The Emergency Location Extension from the **Station** screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type a new extension for the **Emergency Location Ext**. field that will appear on the **Station** screen, up to seven numbers that make up a valid extension number for your dial plan.

IP Parameter Emergency Location

The Emergency Location Extension from the **IP Address Mapping** screen associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
n/a	The words "**See Ip-network Map Form**" display. The administrator can only change this field on the **IP Address Mapping** screen (using the `ip-network-map` command).

Message Lamp

The Message Lamp Extension associated with the current extension is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type a new extension for the **Message Lamp Ext.** field, up to seven numbers that make up a valid extension number for your dial plan.

Port

This field is read only, and displays the port of the existing extension.

Station

The current extension that is being changed (the extension that was typed in the **`change extension-station`** ***`xxxxxxx`*** command) is displayed under **From Extension**.

Valid entries for "To Extension"	Usage
0 to **9**	Type the new extension that you want the current extension changed to, up to seven numbers that make up a valid extension number for your dial plan.

Station Name

This field is read only, and displays the name of the existing extension (the extension that was typed in the **`change extension-station`** ***`xxxxxxx`*** command).

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Circuit Packs

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Class of Restriction

Use this screen to establish classes of restriction (COR). Classes of restriction control call origination and termination. Your system might use only one COR or as many as necessary to control calling privileges. You can assign up to 995 different CORs.

Consider the following to enhance your system security:

1. Assign a separate COR to incoming and outgoing trunk groups, then restrict calling between the two groups.
2. Limit the calling permissions as much as possible by setting appropriate Calling Party Restrictions and Facility Restriction Levels (FRLs).

Field descriptions for page 1

Figure 294: Class of Restriction screen (page 1)

```
change cor n                                                Page 1 of x
                             CLASS OF RESTRICTION
               COR Number: n
          COR Description: supervisor

                      FRL: 0                                   APLT? y
  Can Be Service Observed? n              Calling Party Restriction: none
Can Be A Service Observer? y               Called Party Restriction: none
 Partitioned Group Number: 1          Forced Entry of Account Codes? n
         Priority Queuing? n                    Direct Agent Calling? y
     Restriction Override: none           Facility Access Trunk Test? n
     Restricted Call List? n                     Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y               Fully Restricted Service? n
Group II Category For MFC: 7               Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_               Add/Remove Agent Skills? y
            MF ANI Prefix: ______          Automatic Charge Display? n
Hear System Music on Hold? y     PASTE(Display PBX Data on telephone)? n
                             Can Be Picked Up By Directed Call Pickup? n
                                         Can Use Directed Call Pickup? n
                                         Group Controlled Restriction: inactive
```

Access to MCT?

This field refers to Malicious Call Trace.

Valid entries	Usage
y	Enter **y** to allow permissions to activate a request to trace a malicious call.
n	Entering **n** prohibits this user from requesting a malicious call trace, but does not prevent this extension from appearing in the MCT History report, should this extension be the subject of a malicious call trace.

Add/Remove Agent Skills

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to add and remove skills.

APLT

Valid entries	Usage
y/n	Enter **n** to allow access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities. If fully restricted service is enabled, set this field to **n**.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Not available in the U.S

.

Valid entries	Usage
y	Displays call charges during and at the end of the call.
n	Call charges can be seen if users press the **disp-chrg** button before the call drops.

Called Party Restriction

Valid entries	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls are allowed to go through to the called party as well as attendant-assisted calls if the **Restriction Override** field in the public restricted station's COR is **attd** or **all**.
Termination	Blocks the called party from receiving any calls at any time.
none	No called party restrictions.

Calling Party Restriction

This field determines the level of calling restriction associated with this COR.

Note:

To enhance system security, limit calling permissions as much as possible.

Valid entries	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A telephone with this COR can initiate Remote Access calls, if the COR of the barrier code allows it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same server running Avaya Communication Manager or within a private network. To enhance security, Avaya recommends that you use outward restrictions when practical.
All-toll	Blocks the calling party from making ARS and trunk access calls from a facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the **Toll Analysis** screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls from the facility assigned the COR to certain toll areas as defined in the **Dialed String** field on the **Toll Analysis** screen. The **Dialed String** field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose **Dialed String** field also matches the dialed number. See Toll Analysis on page 1659 for additional information.
none	No calling party restrictions.

Can Be a Service Observer

If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

SECURITY ALERT:

The use of Service Observing features might be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

Note:

You cannot enter **y** in the previous two fields unless **Service Observing (Basic)** is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** if users with this COR can service observe other users.

Can Be Picked Up By Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow this Station's or EAS agent's calls to be picked up by using the Directed Call Pickup Up feature. Before you can set this field to **y**, you must set **Directed Call Pickup** on the **Feature-Related System Parameters** screen to **y**.

Can Be Service Observed

Note that this field allows or denies service observing for not only physical extensions, but also for logical agent IDs and VDNs. If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table

.

Valid entries	Usage
y/n	Enter **y** if users with this COR can be service observed.

Can Change Coverage

Valid entries	Usage
y/n	Enter **y** to allow station users with this COR to select one of two previously administered coverage paths or to activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location.

Can Use Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow the station, attendant, or EAS agent to pick up calls using the Directed Call Pickup feature. Set **Directed Call Pickup** on the **Feature-Related System Parameters** screen to **y** to set this field to **y**.

COR Description

Valid entries	Usage
Up to 35 characters	Enter a description of the COR that indicates how you use it. If you make this as clear as possible (for example, Customer Service, Legal Department), it will be easier to remember which COR to assign when you add users.

COR Number

This is a display-only field when the screen is accessed via an administration command such as **`change`** or **`display`**. Displays the COR number.

Direct Agent Calling

Valid entries	Usage
y/n	If this is **y**, users can dial an ACD agent's extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent can receive calls directly.

Facility Access Trunk Test

An associated feature button (**trk-ac-alm**) status lamp lights when a successful test attempt occurs. Pressing one of the alarm buttons (ten maximum) when its associated status lamp is lit turns off all lamps on all buttons whether the access is still in progress or has completed

.

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to perform Facility Access Trunk Tests.

Forced Entry of Account Codes

FEAC must be enabled on the **System Parameters Customer-Options (Optional Features)** screen and on the **CDR System Parameters** screen.

See Forcing Users to Enter Account Codes on page 641 for more information.

Note:
If a COR requiring entry of account codes is assigned a VDN, the route to commands executed by the associated vector will not be successful.

Valid entries	Usage
y/n	Enter **y** to indicate that an account code must be dialed when making outgoing trunk calls. If this is **y**, any telephone assigned the associated COR must dial an account code before making an outgoing call. If you set this to y for a COR assigned to a trunk group, users must dial account codes before calling out over that trunk group. This might be useful for trunks used in international calls, and those that are more expensive. If a call is being routed by ARS, account code checking is not done on the COR.

FRL

Valid entries	Usage
0 to **7**	Enter an originating FRL number. AAR and/or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator (typically, a telephone user). An originating FRL of 0 has the least calling privileges. To enhance system security, assign the lowest possible FRL.

Fully Restricted Service

Note:
If this field is enabled, the **APLT** field must be **n**.

Valid entries	Usage
y/n	When **y** entered for a given COR, stations assigned that COR will not have access to the public network for either incoming or outgoing calls.

Group II Category For MFC

This field always controls categories for Russian signaling trunks. It can control categories for R2-MFC signaling trunks, depending on the value of the **Use COR for Calling Party Category** field on the **Multifrequency-Signaling-Related System Parameters** screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling

.

Valid entries	Usage
1 to **10**	Enter the value you want the server running Avaya Communication Manager to send as the Calling and/or Called Party Category for telephones or trunks that use this COR.

Group Controlled Restriction

A display-only field that determines if the current COR is under controlled restriction. This field can help troubleshoot problems by first checking its value.

.

Valid entries	Usage
active	indicates the COR is controlled restricted
inactive	indicates the COR is not controlled restricted

Hear System Music on Hold

Valid entries	Usage
y/n	Enter **y** to allow the Music on Hold feature to be activated by a telephone.

Hear VDN of Origin Announcement

Valid entries	Usage
y/n	Enter **y** if users with this COR can receive VDN of Origin messages.

MF ANI Prefix

Defines the prefix to apply to an extension number when ANI is sent to the CO. This overrides any ANI prefix administered on the **Multifrequency Signaling** screen. This does not apply when ANI is tandemed through the Communication Manager server on tandem calls. This field also applies to the ANI for the server when the originating side is a trunk and there was no ANI.

Valid entries	Usage
1 to 7 digits or blank	If you want the entire number to display on the receiving end, enter all digits except the extension number.

Partitioned Group Number

This field appears only if **AAR/ARS Partitioning** is **y** and **Time of Day Routing** is **n** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS partitioned group number associated with this COR.

PASTE (Display PBX Data on telephone)

Valid entries	Usage
y/n	Enter **y** to download all lists. Enter **n** to disallow the PASTE feature.

Priority Queuing

Valid entries	Usage
y	Enter **y** to allow the telephone user's calls to be placed ahead of non-priority calls in a hunt group queue
n	If you do not use Automatic Call Distribution (**ACD** is not enabled on the **System Parameters Customer-Options (Optional Features)** screen), this field must be **n**.

Restricted Call List

This list can be used whether the COR is toll restricted. The Restricted Call List (RCL) has priority over the Toll Analysis Unrestricted Call List (UCL). A call attempt from a facility assigned a COR (with **RCL** field set to **y**), whose dialed digit string is on the **Toll Analysis** screen and is marked as being associated with the RCL, will be denied.

Valid entries	Usage
y/n	Enter **y** to specify that this COR will have access to the system's Restricted Call List (see Toll Analysis on page 1659).

Restriction Override

Allows the specified users to bypass restriction on conference, transfer or call forwarding operations.

Valid entries	Usage
attendant	A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. Enter **attendant** to give your attendants the ability to override this restriction.
all	Enter **all** if you want all of the users with this COR to override inward restrictions.
none	Enter **none** if you do not want any users of this COR to bypass the restrictions.

Send ANI for MFE

Only applicable for Spain. Valid for 2/6 signaling, but not 2/5 signaling. The following field appears only if **Expert Agent Selection (EAS)** is enabled on the **Feature-Related System-Parameters** screen.

Valid entries	Usage
y	Enter **y** to enable Automatic Number Identification (ANI). When the value is **y**, Avaya Communication Manager sends the calling party's number to the public or IBERCOM network so that charges will be broken down by line.
n	If this value is **n**, charges are not itemized by line, and your company will receive a single bill for the total number of calls made (block charging).

Time of Day Chart

Appears only if **Time of Day** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen. See Setting up Time of Day Routing on page 328 for more information.

Valid entries	Usage
1 to **8**	Enter the AAR/ARS time-of-day-chart number associated with this COR.

Unrestricted Call List

Any entry on the **Toll Analysis** screen with an **X** in the **Toll List** column is restricted, meaning that the system blocks any attempt to complete a call containing the Dialed String. However, this field overrides that restriction.

For example, if the **Toll Analysis** screen shows a **Dialed String** entry of 538 and there is an **X** in the **Toll List** column, the 538 number is restricted. To override this restriction, in the **Toll Analysis** screen, enter **X** in the **5** column under the **Unrestricted Call List** heading. In the **Class of Restriction** screen, in this field, enter **5** to complete the restriction override.

Valid entries	Usage
1 to **10** or blank	Appears when **Calling Party Restriction** is **all-toll** or **tac-toll**. This field allows a user to complete a toll call with "restricted" dialed digits. This field is associated with the **Dialed String** field on the **Toll Analysis** screen. An Unrestricted Call List number is denoted on that screen.

Field descriptions for page 2

Figure 295: Class of Restriction screen (page 2)

```
change cor nn                                                     Page 2 of x
                         CLASS OF RESTRICTION

                     MF Incoming Call Trace? n
               Brazil Collect Call Blocking? n
                     Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                     Remote Logout of Agent? n

                           Station Lock COR: 10
  Outgoing Trunk Disconnect Timer (minutes):
       Line Load Control:
 Maximum Precedence Level:        Preemptable?
 MLPP Service Domain:
      Station-Button Display of UUI IE Data?
      Service Observing by Recording Device?
                         ERASE 24xx USER DATA UPON
        Dissociate or unmerge at this phone: none
          EMU login or logoff at this phone: none
           Mask CPN/NAME for Internal Calls:
```

Block Enhanced Conference/Transfer Display

Use this field to add display messages regarding conference and transfer features on digital telephones.

Valid entries	Usage
y/n	Enter **y** to block all the enhanced conference/transfer display messages except "Transfer Completed."

Block Transfer Display

Valid entries	Usage
y/n	Enter **y** to prevent users of DCP, Hybrid, ISDN-BRI, or wireless display telephones from receiving a confirmation message when they transfer a call.

Brazil Collect Call Blocking

For Brazil only.

Valid entries	Usage
y/n	Enter **y** to permit all Brazilian trunks calls that terminate to a station to send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call. Set **Country** on the **Trunk Group** screen to **23** and set this field to **y**.

Erase 24xx User Data Upon: Dissociate or unmerge this phone

Use this field to administer what local terminal data items are erased when the 24xx is dissociated or unmerged.

Valid entries	Usage
none	No local terminal data is erased. This is the default.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Erase 24xx User Data Upon: EMU login or logoff at this phone

Use this field to administer what local terminal data items are erased upon Enterprise Mobility User (EMU) login or logoff.

Valid entries	Usage
none	No local terminal data is erased. This is the default.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Line Load Control

Valid entries	Usage
1 to **4**	Enter the line load control level for this COR, where **1** has no restrictions, and **4** is most restrictive.

Mask CPN/Name for Internal Calls

Valid entries	Usage
y/n	Enter **y** to hide the display of calling/called party numbers and administered name on internal calls.

Maximum Precedence Level

Assign a maximum precedence level for extensions with this COR for use with the Multiple Level Precedence and Preemption feature.

Valid entries	Usage
fo	Flash Override
fl	Flash
im	Immediate
pr	Priority
ro	Routine (default)

MF Incoming Call Trace

Valid entries	Usage
y/n	Enter **y** to allow assignment of a Call Trace COR to a station. Avaya Communication Manager then generates an MFC backward signal (administered on the **System-Parameters Multifrequency-Signaling** screen) during call setup instead of the "free" signal. This triggers the central office to collect trace information before releasing the calling party, if the terminating station's COR has this feature set to **y**.

MLPP Service Domain

Valid entries	Usage
1 to **16777215**	Enter the service domain for users and trunks to which this particular COR is assigned.

Outgoing Trunk Disconnect Timer (minutes)

This feature provides the capability to disconnect an outgoing trunk automatically after an administrable amount of time. This field defaults to blank (outgoing trunk calls are only disconnected when dropped by one or all parties), or you can enter a timer value in number of minutes to apply to outgoing trunk calls if the initiating party belongs to this COR

.

Valid entries	Usage
2 to **999**	Enter a value of as many as 3 characters in number of minutes. A warning tone is given to all parties on the trunk call 1 minute before the administered value (i.e., after 1 to 998 minutes have elapsed) and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.

Preemptable

Valid entries	Usage
y/n	Enter **y** to make extensions with this COR preemptable for Multiple Level Precedence and Preemption calls.

Remote Logout of Agent

Use a feature access code to logout an idle ACD or EAS agent without being at the agent's telephone.

Valid entries	Usage
y/n	Enter **y** to allow remote logout of an idle ACD or EAS agent.

Service Observing by Recording Device

Valid entries	Usage
y/n	When set to **y**, the service observer associated with the COR is actually a remote service-observing connection made by an audio recording device such as the Witness product. Default is **n**.

Station-Button Display of UUI IE Data

This field can only be set to **y** if the Call Center release is 3.0 or later.

.

Valid entries	Usage
y/n	Enter **y** to allow a station user to push a **uui-info** station-button and see up to 32 bytes of ASAI-related User-User-Information Information Element (UUI-IE) data. Pressing the **uui-info** button displaces the incoming call/collected digits display. Pressing **callr-info** redisplays the collected digits. Default is **n**.

Station Lock COR

This field defaults to the current screen COR. Extensions that are assigned this COR can use Station Lock with the access code administered on the **FAC** screen

.

Valid entries	Usage
0 to **995**	This field defaults to current COR.

Field descriptions for page 3

Figure 296: Class of Restriction screen (page 3)

```
change cor nn                                                  Page 3 of x
                      CLASS OF RESTRICTION

            SAC/CF Override by Team Btn? n
SAC/CF Override Protection for Team Btn? n
```

SAC/CF Override by Team Btn

This feature allows the user of a station with a **Team** button administered, who is monitoring another station, to directly reach the monitored station by pushing the **Team** button. This overrides any currently active rerouting (e.g., Send All Calls, Call Forwarding) on the monitored station.

Valid entries	Usage
y/n	Enter **y** to allow override of active rerouting on a monitored station. Default is **n**.

SAC/CF Override Protection for Team Btn

Valid entries	Usage
y/n	Enter **y** to protect stations in this COR from SAC/CF Override rerouting. Default is **n**.

Field descriptions for page 4 to 13

Use pages 4 to 13 to assign up to 995 CORs.

Figure 297: Class of Restriction screen (page 4)

```
change cor nn                                                 Page 4 of x
                                CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     15? n     30? n     44? n     58? n     72? n    86? n
  1? n     16? n     31? n     45? n     59? n     73? n    87? n
  2? n     17? n     32? n     46? n     60? n     74? n    88? n
  3? n     18? n     33? n     47? n     61? n     75? n    89? n
  4? n     19? n     34? n     48? n     62? n     76? n    90? n
  5? n     20? n     35? n     49? n     63? n     77? n    91? n
  6? n     21? n     36? n     50? n     64? n     78? n    92? n
  7? n     22? n     37? n     51? n     65? n     79? n    93? n
  8? n     23? n     38? n     52? n     66? n     80? n    94? n
  9? n     24? n     39? n     53? n     67? n     81? n    95? n
 10? n     25? n     40? n     54? n     68? n     82? n    96? n
 11? n     26? n     41? n     55? n     69? n     83? n    97? n
 12? n     27? n     42? n     56? n     70? n     84? n    98? n
 13? n     28? n     43? n     57? n     71? n     85? n    99? n
 14? n     29? n
```

CALLING PERMISSION

Valid entries	Usage
y/n	A **y** means an originating facility assigned this COR can be used to call facilities assigned this COR. Enter **n** for each COR number (0 through 95) that cannot be called by the COR being implemented.

SERVICE OBSERVING PERMISSION

Valid entries	Usage
y/n	A **y** grants permission to observe specific CORs. Enter **n** for each COR number (0 through 95) that cannot be observed by the COR being implemented.

Class of Service

This screen administers access permissions for call processing features that require dial code or feature button access.

Note:

Class of Service (COS) does not apply to trunk groups except for the Remote Access feature.

A COS assignment defines whether or not a telephone user can access or use the following features and functions. Up to 16 different COS numbers can be administered (0 to 15). When the **Tenant Partitioning** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen, you can administer up to 100 COS groups, each with 16 Classes of Service. This can be useful in controlling service to the stations and attendant of different tenants.

Field descriptions for page 1

Figure 298: Class of Service screen

```
change cos-group 1                                                  Page 1 of x
CLASS OF SERVICE       COS Group: 1     COS Name: COS Group 1



                                    0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                   n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls              n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                    n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling                n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions             n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                  n n n n n n n n n n  n  n  n  n  n  n
    Client Room                     n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net       n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA            n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access (PSA)   n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All         n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA        n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations    n n n n n n n n n n  n  n  n  n  n  n
    Contact Closure Activation      n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion             n n n n n n n n n n  n  n  n  n  n  n
```

The screen lists the default values for each COS/feature combination. For a particular combination, **y** allows access to the feature and **n** denies access. Assign entries on the screen for each COS to be implemented. Default values are shown on the screen.

CAUTION:

Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Automatic Callback

Allows this user to request Automatic Callback. See "Automatic Callback" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Automatic Exclusion

Allows a user to activate automatically Exclusion when they go off hook on a station that has an assigned **Exclusion** button. If set to **n**, allows a user manual exclusion when they press the **Exclusion** button before dialing or during a call. Appears when, on the **Feature-Related System Parameters** screen, the **Automatic Exclusion by COS** field is **y**.

Call Forwarding All Calls

Allows this user to forward all calls to any extension. See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Call Forwarding Busy/DA

Allows this user to forward calls to any extension when the dialed extension is busy or does not answer. See "Call Forwarding" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Client Room

Allows users to access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. You can administer class of service for Client Room only when you have Hospitality Services and a Property Management System interface.

Console Permissions

Console Permissions allow multiappearance telephone users to control the same features that the attendant controls. You might assign this permission to front-desk personnel in a hotel or motel, or to a call center supervisor. With console permission, a user can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or group of extensions
- Activate and deactivate Do Not Disturb for another extension or group of extensions
- Activate Call Forwarding for another extension
- Add and remove agent skills
- Record integrated announcements

Contact Closure Activation

Allows a user to open and close a contact closure relay.

COS Group

This field appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **Tenant Partitioning** field is **y**. The Class of Service group corresponding to the value given in the command line (`cos-group number` [between 1 to 100]. You can administer up to 100 COS groups.

COS Name

This field appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **Tenant Partitioning** field is **y**. The identifying name for this COS group.

Extended Forwarding All

Allows a user to administer call forwarding (for all calls) from a remote location. You cannot change a COS to **y** if **Extended Cvg/Fwd Admin** on the **System Parameters Customer-Options (Optional Features)** screen is **n**.

Extended Forwarding B/DA

Allows this user to administer call forwarding (when the dialed extension is busy or does not answer) from a remote location. You cannot change this COS to **y** if **Extended Cvg/Fwd Admin** on the **System Parameters Customer-Options (Optional Features)** screen is **n**.

Off-Hook Alert

See "Emergency Access to the Attendant" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. To enable this option, either the **Hospitality (Basic)** or **Emergency Access to Attendant** field must be enabled in your license file. When enabled, these fields display as **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Personal Station Access (PSA)

Allows users to associate a telephone to their extension with their programmed services, using a feature access code. This field must be set to **n** for virtual telephones. This field must be set to **y** at a user's home station in order for that user to use the Enterprise Mobility User (EMU) feature at other stations. You cannot change this field to **y** if **Personal Station Access (PSA)** on the **System Parameters Customer-Options (Optional Features)** screen is **n**. See "Personal Station Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. For more information about Enterprise Mobility User, see Setting Up Enterprise Mobility User on page 171.

Priority Calling

Allows user to dial a feature access code to originate a priority call. Such calls ring differently and override send all calls, if active. See "Priority Calling" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

QSIG Call Offer Originations

Allows this user to invoke QSIG Call Offer services. See *Administration for Network Connectivity for Avaya Communication Manager,* 555-233-504, for more information.

Restrict Call Fwd-Off Net

This restricts users from forwarding calls to the public network. For security reasons, this should be enabled for all classes of service except the ones you use for very special circumstances. See "Call Forwarding Off-net" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Trk-to-Trk Restriction Override

Users with this COS override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS. See "Trunk-to-Trunk Transfer" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

SECURITY ALERT:
Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

Field descriptions for page 2

Figure 299: Class of Service screen

```
change cos                                                              Page 2 of x
                                CLASS OF SERVICE

                                    0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

VIP Caller                          y y y y y y y y y y  y  y  y  y  y  y

Masking CPN/Name Override           y y y y y y y y y y  y  y  y  y  y  y
Call Forwarding Enhanced            y y y y y y y y y y  y  y  y  y  y  y
Priority Ip Video                   y y y y y y y y y y  y  y  y  y  y  y
Ad hoc Video Conferencing           y y y y y y y y y y  y  y  y  y  y  y
```

Ad hoc Video Conferencing

Enables Ad-hoc Video Conferencing, so that up to six users can participate in a video conference call.

Call Forwarding Enhanced

Allows users to designate different preferred destinations for forwarding calls that originate from internal and external callers.

Masking CPN/Name Override

Allows users to override the MCSNIC capability (that is, masking the display of calling party information and replacing it with a “hard-coded,” system-wide text string, "Info Restricted").

Priority Ip Video

Allows priority video calling, where video calls have an increased likelihood of receiving bandwidth and can also be allocated a larger maximum bandwidth per call.

VIP Caller

Enables automatic priority calling when assigned to the originator of a call. A call from a VIP phone is always a priority call without the use of a feature button or FAC. Default is **n**. For more information on the VIP Caller feature, See "Priority Calling" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205.

Code Calling IDs

On systems with chime paging, use this screen to assign a unique series of chimes (a *chime code*) to extensions. The chime code assigned to an extension plays over the speakers whenever that extension is paged. You can assign chime codes to up to 125 extensions. Page 1 of this screen provides for the entry of ID Assignments 111-245. Page 2, IDs 251-435, and Page 3, IDs 451-555.

Field descriptions for page 1

Figure 300: Code Calling IDs screen

```
change paging code-calling-ids                                    Page 1 of 3

                                     CODE CALLING IDs
     ID ASSIGNMENTS
              Id   Ext           Id   Ext           Id   Ext
              111:               141:               221:
              112:               142:               222:
              113:               143:               223:
              114:               144:               224:
              115:               145:               225:
              121:               151:               231:
              122:               152:               232:
              123:               153:               233:
              124:               154:               234:
              125:               155:               235:
              131:               211:               241:
              132:               212:               242:
              133:               213:               243:
              134:               214:               244:
              135:               215:               245:
```

Ext

This field assigns extensions to chime codes. Only one extension can be assigned to each chime code.

Valid entries	Usage
An extension	Enter a physical extension, not a VDN, to assign that extension to a code. Otherwise, leave this field blank.

Related topics

See Setting up Chime Paging Over Loudspeakers on page 519 for instructions.

See "Loudpeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information for a description of the feature.

Command Permission Categories

Beginning with Communication Manager 4.0, there is no longer a **Command Permission Categories** screen. For details on screens used for login permissions, see *Maintenance Commands for Avaya Communication Manager*, 03-300431, and "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Communication Interface Processor Channels

See **Processor Channel Assignment**.

Configuration Set

This screen defines a number of call treatment options for Extension to Cellular cellular telephone calls. The Extension to Cellular feature allows the use of up to 99 Configuration Sets, which are already defined in the system using default values.

Field descriptions for page 1

Figure 301: Configuration Set screen

```
change off-PBX-telephone configuration-set 1

                            CONFIGURATION SET: 1

            Configuration Set Description: Standard
                     Calling Number Style: network
       CDR for Calls to EC500 Destination? y
                      CDR for Origination: phone-number
              Fast Connect on Origination? n
             Post Connect Dialing Options: dtmf
            Cellular Voice Mail Detection: none
                            Barge-in Tone? n
              Calling Number Verification? y
Call Appearance Selection for Origination: primary-first
                         Confirmed Answer? n
```

Barge-In Tone

The barge-in tone adds security to Extension to Cellular. If a user is on an active Extension to Cellular call and another person joins the call from the Extension to Cellular enabled office phone, all parties on the call hear the barge-in tone.

Valid fields	Usage
y/n	Default is **n**.

Call Appearance Selection for Origination

Use this field to specify how the system selects a Call Appearance for call origination.

Valid entries	Usage
first-available	If **Bridged Calls** on the **Stations With Off-PBX Telephone Integration** screen is **y**, the system searches for the first available regular or bridged Call Appearance.
primary-first	This is the default. ● If **Bridged Calls** on the **Stations With Off-PBX Telephone Integration** screen is **n**, only regular Call Appearances are used for call origination. If a regular call appearance is not available, the call is not allowed. ● If **Bridged Calls** on the **Stations With Off-PBX Telephone Integration** screen is **y**, the system first searches for a regular Call Appearance for call origination. If a regular Call Appearance is not available, a second search is made that includes both regular and bridged Call Appearances.

Calling Number Style

Determines the format of the caller ID for calls from a local Avaya Communication Manager extension to an Extension to Cellular telephone.

Valid entries	Usage
network	Provides a display of only 10-digit numbers. For internal calls, the ISDN numbering tables are used to create the calling number and DCS calls use the ISDN calling number if provided. The externally provided calling number is used when available for externally originated calls.
pbx	Provides a display of less than 10-digits. Extensions sent as the calling number for all internally- and DCS network-originated calls.

Calling Number Verification

You can restrict what types of calls can be made to an Extension to Cellular cell phone. Accepted calling numbers can be "network provided" or "user provided, verified, and passed." An incoming call will not be allowed to reach the cell phone if both of the following are true:

- the **Calling Number Verification** field is set to **y**

  and

- the incoming call is not "network provided" or "user provided verified and passed"

The default value of **y** has no effect on normal usage of the Extension to Cellular feature. You might change the field to **n** if the switch is part of a private network.

Valid fields	Usage
y/n	Default is **y**

CDR for Calls to EC500 Destination

Determines whether a call detail record is generated for any call to the cell telephone.

Note:
CDR reporting for Extension to Cellular calls relies on the **CDR Reports** field on the **Trunk Group** screen. If, on the **Trunk Group** screen, the **CDR Reports** field is **n**, no CDR is generated even if this field is **y**.

Valid entries	Usage
y	Treats calls to the XMOBILE station as trunk calls and generates a CDR.
n	Treats calls to the XMOBILE station as internal calls and does not generate a CDR.

CDR for Origination

You can generate CDR records for a call that originates from an Extension to Cellular cell phone. To generate this CDR, you must enable the Incoming Trunk CDR. The CDR report does not include dialed Feature Name Extensions (FNEs). The entries for this field determine the CDR report format.

Valid entries	Usage
phone-number	The calling party on the CDR report is the 10-digit cell phone number. This is the default.
extension	The calling party on the CDR report is the internal office phone extension associated with the Extension to Cellular cell phone
none	The system does not generate an originating CDR report.

Cellular Voice Mail Detection

Cellular Voice Mail Detection prevents cellular voice mail from answering an Extension to Cellular call. When you enable Cellular Voice Mail Detection, the call server detects when the cell phone is not the entity that answers the call and brings the call back to the server. Communication Manager treats the call as a normal call to the office phone and the call goes to corporate voice mail. You can also set a timer for cellular voice mail detection that sets a time before Cellular Voice Mail Detection investigates a call.

Valid fields	Usage
none	Default is **none**
timed	Amount of time from 1-9 seconds (default = 4 sec)
message	Detect carrier voice mail

Configuration Set Description

Describes the purpose of the configuration set.

Valid entries	Usage
Up to 20 alphanumeric characters or blank	For example, Extension to Cellular handsets.

Confirmed Answer

Use this field to require the user to input a digit to confirm receipt of a call sent to a cellular telephone by the Extension to Cellular feature. Upon answering the incoming call on the cellular telephone, the user hears a dial tone.The user must then press any one of the digits on the telephone keypad. Until the system receives a digit, the system does not treat the call as answered. The length of time to wait for the digit can be administered from 5-20 seconds, with a default of 10 seconds. The system plays a recall dial-tone to indicate that input is expected. During the response interval, the original call continues to alert at the desk set and any stations bridged to the call. If the user does not enter a digit before the timeout interval expires, the call is pulled back from the cell phone.

Valid entries	Usage
y/n	Enter **y** to enable Confirmed Answer on Extension to Cellular calls for this station. Default is **n**.

Fast Connect on Origination

Determines whether some additional processing occurs on the server running Avaya Communication Manager prior to connecting a call.

Valid entries	Usage
y/n	Enter **y** to send CONNECT messages.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are mapped into XMOBILE stations. These options come into effect after the call has entered the active state (Communication Manager has sent a CONNECT message back to the network).

Valid entries	Usage
dtmf	Expect digits from either in-band or out-of-band, but not simultaneously. The server allocates a DTMF receiver whenever it needs to collect digits. This option normally would be used for Extension to Cellular XMOBILE station calls.
out-of-band	Expect all digits to be delivered by out-of-band signaling only. The server running Avaya Communication Manager collects digits that it needs from the out-of-band channel (no touch-tone receiver). In addition, any digits received when the server is not collecting digits are converted to DTMF and broadcast to all parties on the call. This option is in force for DECT XMOBILE station calls.
both	Expect all subsequent digits to be delivered by simultaneous in-band and out-of-band signaling. Out-of-band signaling consists of digits embedded in ISDN INFO messages while the in-band signaling consists of DTMF in the voice path. The server running Communication Manager collects all digits that it needs from the out-of-band channel. No touch tone receive is allocated in order to prevent collecting double digits. End-to-end signaling occurs transparently to the server via in-band transmission of DTMF. This option is in force for PHS XMOBILE station calls.

Console Parameters

This screen administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays on this screen.

Field descriptions for page 1

Figure 302: Console Parameters — page 1

```
change console-parameters                                       Page   1 of x
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 0                                 COR: 0
       Calls in Queue Warning: 5                  Attendant Lockout? y
        Ext Alert Port (TAAS):
                         CAS: none
                                             Night Service Act. Ext.:
                 IAS (Branch)? n             IAS Tie Trunk Group No.:
         IAS Att. Access Code:                 Alternate FRL Station:
             Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

AAR/ARS Access Code

Appears if the **CAS** field is **QSIG-branch**. An optional field that contains an AAR/ARS access code to route to the main PBX, if needed.

Valid entries	Usage
0 to **9**, *****, **#** blank	Enter up to 4 digits.

Alternate FRL Station

This is a display-only field indicating the extension of the alternate facility restriction level (FRL) activation station.

Attendant Group Name

Valid entries	Usage
1 to **27** alphanumeric characters	Enter a name for the attendant group.

Attendant Lockout

Attendant Lockout prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.

Attendant Lockout provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Valid entries	Usage
y/n	Enter **y** to activate Privacy — Attendant Lockout. If **y** is entered, the attendant is prohibited from reentering a conference call that has been placed on hold unless recalled by a telephone user on the call.

Attendant Vectoring VDN

This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**.

Valid entries	Usage
Assigned VDN extension or blank	Enter an assigned Attendant VDN extension or blank.

Backup Alerting

Indicates whether or not system users can pick up alerting calls if the attendant queue has reached its warning state.

Calls In Queue Warning

Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The console's first Call Waiting lamp lights when any incoming calls are waiting to be answered. The second lamp lights when the number of calls waiting equals the value you entered in the **Calls in Queue Warning** field.

Valid entries	Usage
1 to attendant queue maximum	Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. For queue maximum, see *System Capacities Table for Avaya Communication Manager on Avaya Media Servers*, 03-300511.

CAS

The CAS Main or Branch features must be enabled on the **System Parameters Customer-Options (Optional Features)** screen for either of these features to be functional here.

Valid entries	Usage
main	Enables CAS Main capability.
branch	Enables CAS Branch capability.
none	No Centralized Attendant Service is enabled.
QSIG-main	Can be used if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**. Indicates all attendants are located on the main PBX.
QSIG-branch	Can be used if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**. Indicates there are no local attendants and routes to the main PBX.

CAS Back-Up Ext.

This field handles attendant-seeking calls if the RLT trunk group to the CAS Main server is out of service or if CAS Back-Up is activated. This field must be explicitly defined as an extension in the dial plan. Neither a prefixed extension nor a VDN extension is allowed. Appears only when **branch** is entered in the **CAS** field.

Valid entries	Usage
An extension number for a station	Enter an extension in the dial plan to use for CAS backup.
individual attendant console	
hunt group	
TEG	

COR

For more information about Class of Restriction (COR), see *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired features for the attendant. You can override this COR, by assigning a different COR on the individual **Attendant Console** screen.

COS

For more information about Class of Service (COS), see *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Valid entries	Usage
1 to **15**	Enter a class of service (COS) number that reflects the desired features for all your attendant consoles. You can override this COS, by assigning a different COS on the individual **Attendant Console** screen.

DID-LDN Only to LDN Night Ext.

Valid entries	Usage
y	Enter **y** to allow only listed directory number (LDN) calls to go to the listed directory night service extension.
n	Enter **n** if you want all attendant seeking calls to route to the LDN night service extension.

Ext Alert Port (TAAS)

Enter the seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

Note:

Type an **x** in this field to indicate that there is no hardware associated with this port assignment. If an **x** is used here, you must also fill in the **Ext Alert (TAAS) Extension** field.

Ext Alert (TAAS) Extension

Appears only when an **x** is entered in the **Ext Alert Port (TAAS)** field. This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned (either via TTI or by changing the **Ext Alert Port** field from the G3-MA or other manager terminal) the extension is automatically removed and treated as unassigned.

IAS Att. Access Code

Enter the extension number of the attendant group at the main server running Avaya Communication Manager. This entry is required when IAS Branch is **y**. Does not appear if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**.

IAS (Branch)

Enables or disables Inter-PBX Attendant Service (IAS) Branch feature. Does not appear if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**.

Note:

CAS and IAS cannot both be active at the same time.

IAS Tie Trunk Group No.

Note:
Enter the number of the tie trunk group to the main for the IAS (Branch). This entry is required when IAS Branch is **y**. Does not appear if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Centralized Attendant** field is **y**.

Valid entries	Usage
1 to 666 **1** to **2000**	For DEFINITY CSI. For S8300/S87XX Servers

Night Service Act. Ext.

This is a display-only field containing the extension of the current night service activation station, if any. Such a station is administered by assigning it a **night-serv** button.

QSIG CAS Number

Appears if the **CAS** field is **QSIG-branch**. Contains the complete number of the attendant group at the main server running Avaya Communication Manager, or a vector directory number (VDN) local to the branch server. This field cannot be left blank

Valid entries	Usage
0 to **9**	Enter up to 20 digits.

RLT Trunk Group No.

Appears only when **branch** is entered in the **CAS** field. Enter the trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service.

Field descriptions for page 2

Figure 303: Console Parameters — page 2

```
change console-parameters                                          Page   2 of x
                              CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 10           Return Call Timeout (sec): 10
  Time in Queue Warning (sec):

  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y

ABBREVIATED DIALING
     List1: group    1        List2:                   List3:
            SAC Notification? n
                           COMMON SHARED EXTENSIONS
            Starting Extension:                 Count:
            Busy Indicator for Call Parked on Analog Station Without Hardware?
```

TIMING

Return Call Timeout (sec)

Valid entries	Usage
10 to **1024** or blank	Enter the time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Valid entries	Usage
9 to **999** or blank	Enter the number of seconds a call can remain in the attendant queue before activating an alert.

Time Reminder on Hold (sec)

Valid entries	Usage
10 to **1024**	Enter the number of seconds a call can remain on Hold.

INCOMING CALL REMINDERS

Alerting (sec)

Enter the number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service

No Answer Timeout (sec)

Enter the number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Valid entries	Usage
10 to **1024** or blank	

Secondary Alert on Held Reminder Calls?

Valid entries	Usage
y	Enter **y** to begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Enter **n** to have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and then switch to secondary alerting when the No Answer Timeout expires.

ABBREVIATED DIALING

List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each attendant. However, you cannot assign a personal list to an attendant

.

Valid entries	Usage
enhanced	Allows the attendant to access the enhanced system abbreviated dialing list.
group	Allows the attendant to access the specified group abbreviated dialing list. You also must enter a group number.
system	Allows the attendant to access the system abbreviated dialing list.

SAC Notification

Valid entries	Usage
y/n	Enables or disables Enhanced Attendant Notification for Send All Calls.

COMMON SHARED EXTENSIONS

Busy Indicator for Call Parked on Analog Station Without Hardware?

Valid entries	Usage
y/n	Enter **y** to indicate that the Busy Indicator lamp will light for incoming calls parked on AWOH stations. Default is **n**.

Count

Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions. For example, if you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

The extensions should be assigned to the optional Attendant Selector Console in the 00 through 09 block (bottom row) in any hundreds group for easy identification by the attendant. The lamp associated with the number will identify "call parked" or "no call parked," instead of busy or idle status.

Valid entries	Usage
1 to **1182** or blank	Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions.

Starting Extension

These extension numbers can be used by the attendant to park calls.

Field descriptions for page 3

Figure 304: Console Parameters — page 3

```
change console-parameters                                       Page 3 of x
                                CONSOLE PARAMETERS
QUEUE PRIORITIES
            Emergency Access:1_
             Assistance Call:2_
                     CO Call:2_
            DID to Attendant:2_
                    Tie Call:2_
         Redirected DID Call:2_
             Redirected Call:2_
                 Return Call:2_
                 Serial Call:2_
 Individual Attendant Access:2_
             Interpositional:2_
    VIP Wakeup Reminder Call:2_
          Miscellaneous Call:2_

 Call-Type Ordering Within Priority Levels? n
```

Call-Type Ordering Within Priority Levels?

If you use call-type ordering, calls to the attendant are first grouped by the queue priority level, then by call type, and, finally, in the order received.

Valid entries	Usage
y	Enter **y** if you want to present calls by call type. You can assign a **type-disp** button on the Attendant Console screen so that the attendant can review the call type for the active call.
n	Enter **n** if you wish the calls to be queued in chronological order by queue priority level.

QUEUE PRIORITIES

Attendant Priority Queue allows attendants to answer calls by call category (for example, by trunk type). The Attendant Priority Queue handles incoming calls to an attendant when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

You can assign the same priority level to more than one call. Priority 1 is the highest priority and is the default for Emergency Access. Assign a priority level from **1** through **13** to each of the call types.

The attendant call categories are:

- Emergency Access — A call from a telephone user who dials the emergency access code (default is highest-priority level)
- Assistance Call— A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated
- CO Call — An incoming trunk call (CO/FX/WATS trunk) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- DID to Attendant — An incoming DID trunk call to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Tie Call — An incoming TIE trunk call (dial-repeating or direct types) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Redirected DID Call — A DID or ACD call that times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and reroutes to the attendant group.
- Redirected Call — A call assigned to one attendant, but redirected to the attendant group because the attendant is now busy

- Return Call — A call returned to the attendant after it times out. If the attendant is now busy, the call redirects to the attendant group.
- Serial Call — A call from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access — A call from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available.
- Interposition — A call from one attendant to the Individual Attendant Access (IAA) extension of another attendant
- VIP Wakeup Reminder Call — A VIP Wakeup reminder call.
- Miscellaneous Call — All other calls.

The call types, in descending order of priority, are:

- Type 1 call: outgoing public-network calls receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
- Type 2 call: incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires
- Type 3 call: all other calls (internal calls, conference calls, and tie-trunk calls of any type)

Note that external public-network calls have priority over all other calls including conference calls. And, answered public-network calls have priority over those calls not yet answered.

Field descriptions for page 4

Figure 305: Console Parameters — Queue Priorities screen

```
change console-parameters                                          Page 4 of x
                              CONSOLE PARAMETERS

QUEUE PRIORITIES

      MLPP PRECEDENCE CALL
       Flash Override: 2
                Flash: 3
            Immediate: 4
             Priority: 5
```

QUEUE PRIORITIES

Flash Override

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash Override precedence level calls.

Flash

Valid entries	Usage
1 to **17**	Enter the queue priority for Flash precedence level calls.

Immediate

Valid entries	Usage
1 to **17**	Enter the queue priority for Immediate precedence level calls.

Priority

Valid entries	Usage
1 to **17**	Enter the queue priority for Priority precedence level calls.

Field descriptions for page 5

Note:

If MLPP is not enabled, the MLPP Queues page does not appear, and the following page appears as page 4.

Figure 306: Console Parameters — page 5

```
change console-parameters                                      Page 5 of x
                              CONSOLE PARAMETERS

      Use the 'list attendant' command to see all administered attendants.
```

Coverage Answer Group

This screen establishes Call Coverage Answer Groups.

An answer group contains up to eight members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and that group number appears in the department's coverage path. All telephones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration (see *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207). The members of the group are identified by their extension number. Any telephone, including those administered without hardware (X-ported (but not attendants) can be assigned to a coverage answer group. Note that members whose extensions are X-ported will not be alerted).

Field descriptions for page 1

Figure 307: Coverage Answer Group screen

```
change coverage answer-group n                                  Page 1 of x
                        COVERAGE ANSWER GROUP

                      Group Number: 3___
                      Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ __________________________
2: ________ ___________________________ 6: ________ __________________________
3: ________ ___________________________ 7: ________ __________________________
4: ________ ___________________________ 8: ________ __________________________
```

Ext

Valid entries	Usage
An assigned extension for a station.	Enter the extension number (cannot be a Vector Directory Number extension) for each member of this coverage answer group.

Group Name

Enter the group name you want to use to identify this group.

Tip:

Enter the extension numbers that are group members. This allows a **`list coverage answer group`** command to be used to list the telephones that will be alerted. The **`list`** command can be used in conjunction with the **`list station`**, **`list coverage path`**, and **`list hunt group`** commands to determine stations involved in call coverage. This makes it possible to follow call coverage for any extension, allowing the administrator to easily track call coverage paths.

Valid entries	Usage
Up to 27 characters	For example, typing pool, room 12, secy, and so on.

Group Number

A display-only field when the screen is accessed using an administration command such as **`add`** or **`change`**.

Name

This display-only field indicates the name assigned when the member's telephone is administered.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Coverage Path

This screen implements Call Coverage Paths. The screen provides the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's telephone rings before the call redirects to coverage.

Field descriptions for page 1

Figure 308: Coverage Path screen

```
change coverage path n                                         Page  1 of  x
                                COVERAGE PATH

            Coverage Path Number: n
                                        Hunt After Coverage: n
                Next Path Number: ___   Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?            n             n
              Busy?            y             y
      Don't Answer?            y             y Number of Rings:2
               All?            n             n
DND/SAC/Goto Cover?            y             y
  Holiday Coverage?            n             y  Holiday Table: 1

COVERAGE POINTS
Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: _________ Rng: Point2: _________Rng:
     Point3: _________ Rng: Point4: _________Rng:
     Point5: _________ Rng: Point6: _________Rng:
```

Coverage Path Number

A display-only field indicating the coverage path being administered.

Holiday Coverage

This field determines when to redirect call to coverage for an inside or outside call.

Valid entries	Usage
y	Type **y** to send the call to an announcement.
n	Type **n** to send the call to the next point in the coverage path.

Holiday Table

This field determines when to redirect call to coverage for an inside or outside call.

Valid entries	Usage
y/n	If the **Holiday Table** field is set to **y** for either inside or outside calls, the system uses a holiday table to route the call. Type the number of the holiday table to use.

Hunt After Coverage

Valid entries	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned on the **Station** screen of the last coverage point.
n	Coverage treatment is terminated; the call is left at the last available location (principal or coverage point).

Linkage

Display-only fields that show the (up to) two additional coverage paths in the coverage path chain.

Next Path Number

Enter the next coverage path in a coverage path chain. If the coverage criteria of the current coverage path is not satisfied, the system steps down this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. No path number here indicates that this path is the only path for the principal. See "Call Coverage" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Valid entries	Usage
1 to **9999** or blank	Enter the next coverage path in a coverage path chain.

COVERAGE CRITERIA

COVERAGE CRITERIA are the conditions that, when met, cause the call to redirect to coverage. Assign one of the following:

Valid entries	Usage
Active	Calls redirect if at least one call appearance is busy.
Busy	Calls redirect if all call appearances that accept incoming calls are busy.
Don't Answer	Calls redirect when the specified number of rings has been exceeded.
All	Calls redirect immediately to coverage and overrides any other criteria with a **y** in this column.
DND/SAC/ Goto Cover	Must be assigned before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features. Allows a calling user, when calling to another internal extension, to redirect a call immediately to coverage by pressing a **Go to Cover** button. Allows a principal temporarily to direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the **Send All Calls** (or **Do Not Disturb**) button. **Send All Calls** also allows covering users to temporarily remove their telephones from the coverage path.

Number of Rings

Enter the number of rings.

Valid entries	Usage
1 to **99**	This is the number of rings a user's telephone rings before the system redirects the call to the first point in the coverage path.

COVERAGE POINTS

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially beginning with Point 1 (do not leave gaps). Each path can have up to six coverage points.

Valid entries	Usage
extension	Redirects the call to an internal extension or announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to **h999**	Redirects the call to the corresponding hunt-group. For example, enter **h32** if you want a coverage point routed to hunt group 32. (See Hunt Group on page 1119 for more information.)
c1 to **c750** **c1** to **c1000** (S8300/S87XX Servers)	Redirects the call to the corresponding coverage answer group. For example, enter **c20** if you want a coverage point routed to call coverage answer group 20. (See Coverage Answer Group on page 879 for more information.)
	1 of 2

Valid entries	Usage
r1 to **r999** **r1** to **r1000** S8300/S87XX Servers	Redirects the call to the corresponding remote coverage point number. For example, enter **r27** if you want a coverage point routed to remote coverage point 27. (See Remote Call Coverage Table on page 1438 for more information.)
v + extension	Redirects the call to the corresponding VDN extension. For example, enter **v12345** if you want the last administered coverage point to be the VDN associated with extension 12345. Note that a Vector Directory Number can be used only as the last administered point in a coverage path.
	2 of 2

If calls redirect to an AUDIX in a DCS network, administer a unique **Hunt Group** screen. Assign the AUDIX extension in the **Group Extension** field. If the AUDIX is connected to the local node, set the **Message Center** field to **audix**; if the AUDIX is connected to another node, set the **Message Center** field to **rem-audix**.

If calls redirect to Message Center (a special Uniform Call Distribution hunt group), AUDIX, or to the attendant, do not list any subsequent coverage points. These calls will normally queue and never redirect to another coverage point. Calls to any hunt group will queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full or there is no queue.

If the Coverage of Calls Redirected Off-Net feature is not enabled, a remote coverage point will function as the last point in the coverage path, because the system will no longer have control of the call once it has redirected off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, a call redirected off-net can be monitored by the system and brought back for further call coverage processing.

Rng

Valid entries	Usage
1 to **99** or blank	Enter the number of rings at this coverage point before the system redirects the call to the next point in the coverage path.

Terminate to Coverage Pts. with Bridged Appearances

Valid entries	Usage
y	Allows a call to alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Crisis Alert System Parameters

This screen allows you to define the system parameters associated with sending crisis alert messages.

Field descriptions

Figure 309: Crisis Alert System Parameters screen

```
change system-parameters crisis-alert                        page 1 of x
                          CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
    Every User Responds? n

ALERT PAGER
            Alert Pager? y
  Originating Extension: 7768
      Crisis Alert Code: 911
                Retries: 5
   Retry Interval (sec): 30
            Main Number: 303-555-0800

                      Pager Number          Pin Number
                      1: 3035559001         1: 7614567890
                      2: 123456789012345    2: ppp1234567890pp
                      3: 123456789012345    3: ppp1234567890pp

                  DTMF Duration - Tone (msec): 100   Pause (msec): 100
```

ALERT STATION

Every User Responds

Controls who needs to respond to a crisis alert.

Valid entries	Usage
y	If set to **y**, all users who have a crisis alert button are notified and must clear the alert for every emergency alert. Assign crisis alert buttons only to attendant consoles and stations that must be notified of an emergency call.
n	If set to **n**, all users are notified, but only one user needs to acknowledge an alert. This user might be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

ALERT PAGER

Alert Pager

Valid entries	Usage
y/n	Enter **y** to use Crisis Alert to a Digital Pager.

Crisis Alert Code

Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 through 3 digits	The numbers in this field are the first 3 digits in the crisis alert pager message. Avaya recommends you enter the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

DTMF Duration - Tone (msec)

The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Main Number

The main telephone number to the location or a location code. This field is optional and does not require an entry. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **-** (dash)	Enter a number up to 15 digits to identify the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and not included in the message sent to the pager. This entry is the last group of digits displayed in the pager message.

Originating Extension

Used as the extension originating the call to send a crisis alert message to a pager. Displays when the **Alert Pager** field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 to 7 digits	Requires a valid unassigned extension according to the dial plan.

Pager Number

Displays when the **Alert Pager** field is **y**. One of these fields must have a number or the screen cannot be submitted.

Valid entries	Usage
1 to 15 digits **-** (dash)	Any dashes are for display purposes only and not included in the message sent to the pager One of the pager number fields must have a number or the screen cannot be submitted.

Pause (msec)

The length of time between DTMF tones for each digit. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
20 to **2550**	Enter a number in increments of 10.

Pin Number

This field can be used for any combination of the pager pin number and pauses or left blank. Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
digits **0** to **9** **p**(ause) **#**(pound) *****(star)	Enter a number up to 15 digits. A pause (about 2 seconds) is for timing of the message. For instance, after the pin number you might want to have a pause to allow time for the pager service to set up the correct pager message box. If the pager service requires you to submit a PIN number, enter it here.

Retries

Displays when the **Alert Pager** field is **y**.

Valid entries	Usage
0 to **10**	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Retry Interval (sec)

Displays when the **Alert Pager** field is **y**. This field is not used unless the **Retries** field is **1** to **10**.

Valid entries	Usage
30 to **60**	The administrable time period (in seconds) between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

CTI Link

The **cti-link** commands are available only if, on the **System Parameters Customer-Options (Optional Features)** screen, either the **ASAI Link Core Capabilities and/ or Computer Telephony Adjunct Links** field is **y**.

Field descriptions for page 1

Figure 310: CTI Link screen when Type field is ASAI or ADJLK

```
add cti-link next                                              Page 1 of x
                                 CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI
     Port: 1C0501                                                 COR: 1
     Name: ASAI CTI Link 1









BRI OPTIONS
               XID? y     Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

Figure 311: CTI Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                              Page 1 of x
                                 CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI-IP
                                                                  COR: 1
     Name: ASAI CTI Link 1
```

CTI Link

A display-only field indicating the CTI link number.

Valid entries	Usage
1 to system max	Avaya Communication Manager on a DEFINITY Server CSI, DEFINITY G3i, S8300 Server, S87XX Fiber-PNC Servers.

Extension

This field displays the extension for this link.

Type

For each link that you want to add to your system, you must specify the CTI link type.

Valid entries	Usage
ADJLK	Enter the CTI link type.
ADJ-IP	
ASAI	
ASAI-IP	

Port

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter 7 characters to specify a port, or an x.

Valid entries	Usage
01 to **64**	First and second numbers are the cabinet number
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH.

Name

Enter a name associated with this CTI link.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

BRI Options

XID

Appears when the **Type** field is **ASAI** or **ADJLK**. Used to identify Layer 2 XID testing capability.

MIM Support

Management Information Message Support. A display-only field that appears when the **Type** field is **ASAI** or **ADJLK**.

Fixed TEI

Appears when the **Type** field is **ASAI** or **ADJLK**. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For ASAI, enter **y**.

CRV Length

Appears when the **Type** field is **ASAI** or **ADJLK**. Enter **1** or **2** to indicate the length of CRV for each interface.

Field descriptions for page 2

Figure 312: CTI Link screen when Type field is ASAI-IP or ADJ-IP

```
add cti-link next                                                 Page 2 of x
                                 CTI LINK
FEATURE OPTIONS

    Event Minimization?        Special Character for Restricted Number?
                         Send Disconnect Event for Bridged Apppearance?
                                       Two-Digit Aux Work Reason Codes?
                                           Block CMS Move Agent Events?
```

Block CMS Move Agent Events

Valid entries	Usage
y/n	When this option is set to **y**, if CMS sends an agent-move-while-staffed message (MVAGSFD8), ASAI does not send the associated agent Logout Event Report (C_Logout), Login Event Report (C_login) and Agent Work Mode Change event report messages to report the changes involved with the move of agents while staffed. Default is **n**.

Event Minimization

This option can be used when event reports normally would be sent on multiple associations, but the adjunct does not need to see more than one. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link. When enabled, this option allows only a single such event to be sent. The selection of the association on which the event will be sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled). Use the **Station** screen to change this option. The new option settings take effect the next time the ASAI link is activated.

Valid entries	Usage
y/n	Enter **y** to control the behavior for that particular link.

Send Disconnect Event for Bridged Appearance

Valid entries	Usage
y/n	Enter **y** to indicate that an event report is sent when a bridged appearance disconnects.

Special Character for Restricted Number

Enables an ASAI CTI link to indicate the calling number restricted presentation within an event report. For further information, see *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Valid entries	Usage
y/n	When set to **y** and a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then "*" is appended to the calling party number in the ASAI message.

Two-Digit Aux Work Reason Codes

Valid entries	Usage
y/n	Enter **y** to enable sending two digit Reason Codes over the ASAI link. All messages that include Aux Work Reason Codes will allow a codes of 1 to 99. This field can only be set to **y** when **Two-Digit Aux Work Reason Codes?** on the Feature-Related System Parameters screen is set to **y**. Default is **n**.

Customer Options

See **System Parameters Customer-Options (Optional Features)**.

Data Module

The following section provides descriptions of standard fields on **Data Module** screens. Some of the fields are used for specific data module types; others are used for all data modules. Unique fields and fields requiring special consideration are listed with the appropriate data module descriptions in this book.

Field descriptions for page 1

Figure 313: Data Module screen

```
change data-module nn                                           Page   1 of x
                                DATA MODULE

   Data Extension: 30            Name: 27                             BCC:
             Type: data-line___   COS: 1       Remote Loop-Around Test?
             Port: _______        COR: 1         Secondary data module?
              ITC: restricted__    TN: 1                  Connected to: dte


 ABBREVIATED DIALING
   List1:

 SPECIAL DIALING OPTION:



 ASSIGNED MEMBER (Station with a data extension button for this data module)

           Ext     Name
       1: 1002    27 character   station name
```

BCC

(Bearer Capability Class) A display-only field used with Data Line, Netcon, Processor Interface, Point-to-Point Protocol, Processor/Trunk (**pdm** selection), and System Port Data Modules. Appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. The value in this field corresponds to the speed setting of the data module. This field can be compared with the BCC value in an associated routing pattern when attempted calls utilizing the data module fail to complete. The BCC values must be the same.

See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a detailed description of Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities (ISDN-PRI Interworking).

Valid entries	Usage
1	Relates to 56-bkps
2, 3, 4	Relates to 64 kbps

Board

Used with Announcement Data Modules. Enter the five character announcement circuit pack number that identifies the physical circuit pack to which the announcement module is connected. You can enter **x** in this field to indicate that there is no hardware associated with this port assignment.

The five character announcement board number is comprised of:

Characters	Meaning	Value
1 to 2	Cabinet Number	**1** to **64** (S87XX Series IP-PNC)
3	Carrier	**A** to **E**
4 to 5	Slot Number or X	**0** to **20**

Broadcast Address

Used with Ethernet data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information. Does not appear for S87XX Series IP-PNC.

Connected Data Module

Used with Processor Interface (used with DEFINITY CSI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Connected to

Used with Data Line and Processor/Trunk (**pdm** selection) Data Module. This field shows what the Asynchronous Data Unit (ADU) is connected to.

Valid entries	Usage
dte	Data Terminal Equipment. Used with Data Line and Processor/Trunk Data Modules.
isn	Information Systems Network. Used with Data Line and Processor/Trunk Data Modules.

COS

Does not appear for **ethernet**. Enter the desired class of service.

Valid entries	Usage
0 to **15**	Select the allowed features.

COR

Does not appear for **ethernet**. Enter the desired class of restriction.

Valid entries	Usage
0 to **995**	Select the allowed restriction.

Data Extension

A display-only field indicating the extension assigned to the data module.

Enable Link

Used with Point-to-Point, and Processor Interface data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information. This field is in different locations on the screen for different data module types.

Establish Connection

Used with Point-to-Point, and Processor Interface (used with DEFINITY CSI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

IP Address Negotiation

Used with Point-to-Point data modules. Does not appear for S87XX Series IP-PNC. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

ITC

(Information Transfer Capability) Used with 7500, Announcement, data-line, Netcon, Processor/ Trunk (**pdm** selection), Processor Interface, and System Port Data Modules. Appears only when, on the **Trunk Group** screen, the **Comm Type** field is **56k-data** or **64k-data**. Indicates type of transmission facilities to be used for ISDN calls originated from this endpoint. Does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1’s density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1’s density digital transmission (that is, digital information is sent exactly as is).

Link

Used with Ethernet, Point-to-Point, and Processor Interface (used with DEFINITY CSI only) data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information. This field is in different locations on the screen for different data module types.

Valid entries	Usage
1 to **99**	Enter a communication interface link number.

Maintenance Extension

Used with Netcon and Processor Interface Data Modules.

Valid entries	Usage
Enter the extension number required to perform maintenance functions on the standby netcon physical channel in a duplicated system.	The standby remote loop around tests fails if this field is not administered.

MM Complex Voice Ext

Used with 7500 and World Class BRI Data Modules. Does not appear on S87XX Series IP-PNC. This field contains the number of the associated telephone in the multimedia complex. This field appears only after you set the **Multimedia** field to **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext** on the **Station** screen.

Valid entries	Usage
Valid values conform to your dial plan	Once you complete the field on the **Station** screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone.

Multimedia

Used with the 7500 and World Class BRI Data Modules. Appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **MM** field is **y**.

Valid entries	Usage
y/n	Enter **y** to make this data module part of a multimedia complex.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter the name of the user associated with the data module. The name is optional and can be blank. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Network uses 1's for Broadcast Addresses

Used with Ethernet data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Node Name

Used with Ethernet (not on S87XX Series IP-PNC) and Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

PDATA Port

Used with System Port Data Modules. Enter a seven-digit alphanumeric port location to which the data module is connected. Used to relate the physical PDATA port to which the mode 3 portion of the system port is connected. This entry must be assigned to a port on a PDATA Line Board.

Valid entries	Usage
01 to **22**	First and second characters are the cabinet number
01 to **64**	First and second characters are the cabinet number (S87XX Series IP-PNC)
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number in the carrier
01 to **12**	Sixth and seventh characters are the circuit number

Physical Channel

Used with Netcon and Processor Interface Data Modules. The Physical Channel number is referred to on associated system forms as the Interface Link number.

Valid entries	Usage
01 to **08**	For Processor Interface Data Modules, enter the 2-digit circuit number of the Processor Interface port. A multi-carrier cabinet system supports the use of two Processor Interface circuit packs, the first circuit pack (mounted in Control Carrier A) supports physical channels or links 01 through 04; the second (mounted in Control Carrier A) supports physical channels or links 05 through 08. A single-carrier cabinet system supports one Processor Interface circuit pack and physical channels or links 01 through 04 only.
01 to **04**	For DEFINITY CSI configurations. For Netcon Data Modules, enter a netcon data channel.

Port

Used with 7500, Data Line, Ethernet, Processor/Trunk, PPP, System Port, and World Class BRI Data Modules. Specifies a port location to which the data module is connected.

Characters	Meaning	Value
1-2	Cabinet Number	**01** to **64** (S87XX Series IP-PNC)
3	Carrier	**A** to **E**
4-5	Slot Number	**0** to **20**
6-7	Circuit Number	**01** to **31** (S87XX Series IP-PNC (tdm, pdm) configurations) **01** to **16** (ppp for S87XX Series IP-PNC) **01** to **08** (system-port for S87XX Series IP-PNC) **17/33** (Ethernet on S87XX Series IP-PNC)

Note:

You can enter **x** in the **Port** field to indicate that there is no hardware associated with the port assignment (also known as Administration Without Hardware (AWOH). These stations are referred to as "phantom stations." If this data module is designated as a secondary data module (Secondary data module set to **y**) An **x** cannot be entered into this field. The port of a primary data module cannot be changed to **x** if a secondary data module is administered.

Remote Loop-Around Test

Used with Processor/Trunk Data Modules. Appears when the **Type** field is **pdm**, or **tdm**.

Valid entries	Usage
y/n	For Processor/Trunk Data Modules, enter **y** if the data module supports a loop-back test at the EIA interface. In general, Avaya equipment supports this test but it is not required by Level 2 Digital Communications Protocol. Enter **n** to abort a request for this test.

Secondary data module

Used with Processor/Trunk Data Modules. Appears only when the **Type** field is **pdm**. The primary data module must be administered before the secondary data module can be added. If the **Port** field is **x**, the **Secondary Data Module** field cannot be **y**.

Valid entries	Usage
y	This PDM is the secondary data module used for Dual I-channel AUDIX networking.
n	This is the primary PDM, or if this data module is not used for AUDIX networking.

Subnet Mask

Used with Point-to-Point data modules (for S87XX Series IP-PNC). See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

TN

Valid entries	Usage
1 through **100**	Enter the Tenant Partition number.

Type

Enter the type of data module.

Valid entries	Usage
7500	Assigns a 7500 Data Module. The 7500 data module supports automatic TEI, B-channel, maintenance and management messaging, and SPID initialization capabilities. BRI endpoints, both voice and/or data, are assigned to either the ISDN-BRI - 4-wire S/T-NT Interface circuit pack or the ISDN-BRI - 2-wire U circuit pack. Each can support up to 12 ports. Since BRI provides multipoint capability, more than one ISDN endpoint (voice or data) can be administered on one port. For BRI, multipoint administration allows for telephones having SPID initialization capabilities, and can only be allowed if no endpoint administered on the same port is a fixed tie endpoint and no station on the same port has B-channel data capability. Currently, multipoint is restricted to 2 endpoints per port.
announcement	Assigns an announcement data module. The announcement data module is built-in to the integrated announcement circuit pack and is administered using the **Announcement Data Module** screen. This data module allows the system to save and restore the recorded announcements file between the announcement circuit pack and the system memory.
data-line	Assigns a Data Line Data Module. The **Data Line Data Module (DLDM)** screen assigns ports on the Data Line circuit pack (DLC) that allows EIA 232C devices to connect to the system. The DLC, with a companion Asynchronous Data Unit (ADU), provides a less expensive data interface to the system than other asynchronous DCP data modules. The DLC supports asynchronous transmissions at speeds of Low and 300, 1200, 2400, 4800, 9600, and 19200 bps over 2-pair (full-duplex) lines. These lines can have different lengths, depending on the transmission speed and wire gauge. The DLC has 8 ports. The connection from the port to the EIA device is *direct*, meaning that no multiplexing is involved. A single port of the DLC is equivalent in functionality to a data module and a digital line port. The DLC appears as a data module to the Digital Terminal Equipment (DTE) and as a digital line port to the server running Avaya Communication Manager. The DLC connects the following EIA 232C equipment to the system: ● Printers ● Non-Intelligent Data Terminals ● Intelligent Terminals, Personal Computers (PCs) ● Host Computers ● Information Systems Network (ISN), RS-232C Local Area Networks (LANs), or other data switches.
	1 of 2

Valid entries	Usage
ethernet	Assigns an Ethernet data module. The **Ethernet Data Module** screen assigns the 10BaseT port on the Control-LAN (C-Lan) circuit pack. This port provides a TCP/IP connection to network hub or LAN. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information on Ethernet data modules.
ni-bri	Assigns an NI-BRI Data Module.
pdm	Assigns a DCE interface for Processor/Trunk Data Modules. These screens assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). One screen is required for assigning MPDMs (700D), 7400B, 7400D or 8400B Data Module, and another screen for MTDMs (700B, 700C, 700E, 7400A). One screen must be completed for each MPDM, 7400B, 7400D, 8400B or MTDM. The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface for connection to equipment such as data terminals, CDR output devices, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. It also provides a Digital Communications Protocol (DCP) interface to the digital switch. (DCE is the equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.) The MTDM provides an Electronic Industries Association (EIA) Data Terminal Equipment (DTE) interface for connection to off-premises private line trunk facilities or a switched telecommunications network and a DCP interface for connection to the digital switch. (DTE is the equipment comprising the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host computer, the terminal, the host, and their associated modems or data modules make up the DTE.) The MTDM or 7400A Data Module also can serve as part of a conversion resource for Combined Modem Pooling.
ppp	Assigns a Point-to-Point Protocol data module. The **PPP Data Module** screen assigns a synchronous TCP/IP port on the Control Lan (C-Lan) circuit pack. These ports are tailored to provide TCP/IP connections for use over telephone lines. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information on Point-to-Point data modules.
system-port	Assigns a System Port Data Module.
tdm	Assigns a DTE interface for Processor/Trunk Data Modules. See the **pdm** entry above.
wcbri	Assigns a World Class BRI Data Module.
	2 of 2

DESTINATION

CHAP

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

CHAP Secret

Appears when the **CHAP** field is **y**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Valid entries	Usage
character string	Enter 1 to 30 characters; first character cannot be @.

Digits

This field appears when the **Type** field is **ppp**. Used with Point-to-Point data modules.

Valid entries	Usage
An extension, or Trunk Access Code (TAC) and extension of destination connection, or blank	Enter the number that the local data module dials to establish a connection to a far-end data module in a private network. .

Node Name

Appears when the **Type** field is **ppp**. Used with Point-to-Point data modules. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504 for more information.

ABBREVIATED DIALING

List1

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Supports Data Hot Line. This field can be left blank.

Valid entries	Usage
e	Enhanced
g	Group. You also must enter a group list number.
p	Personal. You also must enter a personal list number.
s	System.

SPECIAL DIALING OPTION

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

HOT LINE DESTINATION

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Appears only when the **Special Dialing Option** field is **hot-line**. Entry in this field supports Data Hot Line

Valid entries	Usage
0 to **999**	This number is associated with the AD List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

DEFAULT DIALING

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Only appears when the **Special Dialing Option** field is **default**. When the user goes off-hook and enters a carriage return following the DIAL prompt, the system dials the AD number. The data call originator can also perform data-terminal dialing by specifying a dial string that might or might not contain alphanumeric names.

Valid entries	Usage
0 to **999**	Enter a list number associated with the abbreviated dialing list.

CIRCUIT SWITCHED DATA ATTRIBUTES

Used with 7500 and World Class BRI Data Modules.

Note:

These fields represent defaults needed for modem pooling conversion resource insertion when the endpoint does not support data query capability and administered connections. These fields have no significance for data modules providing data query [all Avaya -supported ISDN-BRI data modules (7500 and ADM)]. For Avaya ISDN-BRI or World Class ISDN-BRI data modules, use the default settings.

Default Duplex

Used with 7500 and World Class BRI Data Modules. Used to identify the duplex mode.

Valid entries	Usage
full	Allows simultaneous two-way transmission.
half	Allows only one transmission direction at a time.

Default Mode

Used with 7500 and World Class BRI Data Modules. Used to identify the data mode.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

Default Speed

Used with 7500 and World Class BRI Data Modules. Used to identify the data rate.

Valid entries	Usage
1200 **2400** **4800** **19200**	
56000 **64000**	Can be entered when the **Default Mode** field is **sync**.

ASSIGNED MEMBER

Ext and Name

Used with Data Line, Announcement, Netcon, Processor/Trunk, Processor Interface, and System Port Data Modules. Displays the extension number and name of the user (previously administered) with associated **Data Extension** buttons who shares the module.

DATA MODULE CAPABILITIES

Default Data Applications

Used with 7500 and World Class BRI Data Modules. Used to identify mode to be Used with originating data calls when the mode is not specified with the calling parameters. This mode is also used for terminating trunk calls that do not have bearer capability specified or administered connections. See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for additional information.

Valid entries	Usage
M0	Mode 0. Use this setting for a WCBRI endpoint used as an administered connection.
M1	Mode 1
M2_A	Mode 2 asynchronous
M2_S	Mode 2 synchronous
M3/2	Mode 3/2 adaptable

Default ITC

Used with 7500 and World Class BRI Data Modules.

Valid entries	Usage
restricted	For a WCBRI endpoint used as an administered connection.
unrestricted	

MM Complex Voice Ext:

Used with 7500 and World Class BRI Data Modules. This display-only field contains the number of the associated telephone in the multimedia complex. It only appears when the **Multimedia** field is **y**. This field is left blank until you enter the data module extension in **MM Complex Data Ext f**ield on the **Station** screen. Once you complete the field on the **Station** screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone. Valid values conform to your dial plan.

Field descriptions for page 2 - Type data-line

This version of page 2 appears when the **Type** field is **data-line**.

Figure 314: Data Line Data Module screen— if KYBD Dialing is y

```
change data-module nn                                                 Page 2 of x
                              DATA MODULE
        CAPABILITIES
                       KYBD Dialing? y                        Configuration? n
                           Busy Out? n
        SPEEDS
              Low? y            1200? y            4800? y            19200? y
              300? y            2400? y            9600? y        Autoadjust? n
        OPTIONS
                    Permit Mismatch? n                          Dial Echoing? y
                Disconnect Sequence: two-breaks                  Answer Text? y
                             Parity: even              Connected Indication? y
```

See DLC Option Settings on page 915 for additional information when assigning entries for the remaining fields on the screen.

CAPABILITIES

Busy Out

This option should be enabled for DTEs that are members of a hunt group and to allow "busy out" when DTE turns power off so that calls do not terminate on that DTE.

Valid entries	Usage
y/n	Enter **y** to place the DLC port in a busied-out state once the DTE control lead to the DLC is dropped.

Configuration

Appears when the **KYBD Dialing** field is **y**. This option normally is enabled for "originate/ receive" DTE such as non-intelligent terminals and disabled for intelligent devices such as computers. The **KYBD Dialing** field must be **y** with this option.

Valid entries	Usage
y/n	Enter **y** to allow the viewing and changing of options from the DTE.

KYBD Dialing

This option must be enabled to allow data endpoints to originate calls via the EIA 232C interface and obtain ASCII feedback text. When enabled, the user gets the dial prompt. This option normally is enabled for "originate/receive" DTE that has a need to set up data calls. If this option is disabled, originations cannot be done at the DTE and text feedback does not occur at the DTE during call setup/take down. Data call answering is still allowed but without text feedback.

Note:

ADU-type hunt groups connecting the system to terminal servers on a host computer should have these hunt group extensions assigned as "no" keyboard dialing.

Valid entries	Usage
y/n	Enter **y** to allow keyboard dialing. This enables the data endpoint to receive and transmit text during call origination or termination. Low must be **n**.

SPEEDS

Enter **y** to select operating speeds as follows:

Valid entries	Usage
Low	Enter **y** to instruct the DLC to operate at a low speed from 0 to 1800 bits per second (bps). Enter **n** if the **KYBD Dialing** field is **y**.
300, 1200, 2400, 4800, 9600, or 19200	Enter **y** beside the desired operating speed. Enter **n** if the speed is not desired. The DLC can be any one of these speeds. The speed is matched for the duration of the call, from call setup to call takedown. When multiple speeds are selected (select three or more, do not select just two speeds) and autoadjust is disabled, the DTE's speed must be the highest selected speed. This is required because all feedback text is delivered to the DTE at the highest selected speed.
Autoadjust	Appears when the **KYBD Dialing** field is **y**. Enter **y** which tells the DLC port to automatically adjust to the operating speed and parity of the DTE it is connected to. Enter **n** if this option is not desired. Autoadjust can be selected with any of the speeds selected in the previous step. Autoadjust allows the DLC port to determine the speed and parity of the DTE and then match itself to this speed. Autoadjust only applies to calls originated by the user through Keyboard Dialing.

OPTIONS

Answer Text

Appears when the **KYBD Dialing** field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

- INCOMING CALL
- ANSWERED
- DISCONNECTED
- DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

Connected Indication

Appears when the **KYBD Dialing** field is **y**. This option generates a "CONNECTED" message to the DTE when the connection has been established. If the **KYBD Dialing** field is **n**, the connected indication is provided by the DLC activating its EIA 232C control lead.

Valid entries	Usage
y/n	Enter **y** to select this option.

Dial Echoing

Appears when the **KYBD Dialing** field is **y**.

Valid entries	Usage
y/n	Enter **y** to echo characters back to the DTE. Dial echoing should be disabled when keyboard dialing is done by an intelligent device.

Disconnect Sequence

Appears when the **KYBD Dialing** field is **y**. Selects the sequence for a disconnect.

Valid entries	Usage
long-break	A long-break is greater than 2 seconds.
two-breaks	Two-breaks is within 1 second.

Parity

Appears when the **KYBD Dialing** field is **y**. Select the desired type of parity. The DLC generates the parities when call setup text is sent to the DTE. The DLC does not check the parity when receiving dialing characters. Parity has nothing to do with the far end; it is used by the DLC to terminal communications during call setup. Set to match the connected DTE.

Valid entries	Usage
even	Set to match the connected DTE.
odd	
mark	
space	

Permit Mismatch

This option allows the EIA interface to operate at a rate different than that agreed to in the data module handshake. (The data module handshake is always the highest compatible rate as determined by the reported speed option of each data module.) Permit Mismatch eliminates the need to change the DTE/DLC speed every time a call is placed to/from an endpoint operating at a different speed. When this option is enabled, the DLC reports the highest optioned speed and all the lower speeds (or the previously selected autoadjust speed) during the handshake process.

Valid entries	Usage
y/n	Enter **y** to instruct the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module.

CAUTION:

Caution must be used when using this option to send information from a DTE/DCE that is transmitting data at higher rates than that of the far end. Sustained usage of this type transmission results in loss of data. Whenever this option is enabled, the DTE must match the highest speed selected for the associated DLC port.

This option is intended to be used by a DTE device operating locally at a higher baud rate than that of its far-end connection but transmitting relatively low amounts of data (for example, a user typing at a terminal). Also, this option can be selected whether or not Keyboard Dialing is selected.

Note:

The Low speed setting is not reported as an available speed when the **Permit Mismatch** field is **y**.

DLC Option Settings

The following provides additional information on the option settings for DLCs when used with the following types of devices:

- Printers
- Non-intelligent terminals
- Data terminals and personal computers
- Host computers
- Information Systems Network (ISN)

Printers

A DLC port with a companion ADU, when attached to a printer, usually terminates a data call. Therefore, in this connection, the printer is the endpoint device. The originating device might be attached to a DCP mode 2 data module (such as the MPDM) or the DLC. A Z3A ADU extends the range of the EIA 232C connection.

When a receive-only printer (or any printer that does not generate the Transmit Data and DTR leads) is used, the ADU must be powered from a small plug-mounted transformer (2012D, or equivalent) connected to pins 7 and 8 of the modular jack. (See **ADU User Manual** for details.)

An ADU cannot be used if the printer has hardware flow control using the Clear To Send (CTS) lead. An ADU can be used, however, if the printer is using software flow control.

A printer connected to a DLC is usually assigned as a line. Table 13 lists the option settings for printer connections.

Table 13: DLDM screen settings for printer connection

Field on screen	Option	Comments
Speed	Highest speed at which the Printer operates	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	yes	If printer is member of Hunt Group
Permit Mismatch	yes	No, if printer is low speed
Parity	-	Don't care
Dial Echoing	-	Don't care
Disconnect Sequence	-	Don't care
		1 of 2

Table 13: DLDM screen settings for printer connection (continued)

Field on screen	Option	Comments
Answer Text	-	Don't care
Connected Indication	-	Don't care
Configuration	no	
		2 of 2

Non-intelligent terminals

A non-intelligent terminal connected to the DLC usually is assigned as a line. Table 14 lists the option settings for non-intelligent terminals.

Table 14: DLDM screen settings for connection to non-intelligent terminals

Field On screen	Option	Comments
Speed	All speeds at which the terminal can operate; autoadjust	Subject to distance limitations; Autoadjust when the **KYBD Dialing** field is **y** and the Terminal can generate an ASCII "return"
KYBD Dialing	yes	
Busy Out	no	Yes, if terminal is member of a hunt group
Permit Mismatch	yes	-
Parity	Same as DTE	
Dial Echoing	yes	Only if the **KYBD Dialing** field is **y**
Disconnect Sequence	2	Depends on terminal
Answer Text	yes	
Connected Indication	-	Don't care
Configuration	yes	

Data terminals and personal computers

An intelligent data terminal or a personal computer (PC) attached to a DLC can either originate or terminate a data call. A single ADU at the site of the originating device extends the distance signals can travel to the switch (the model ADU depends on the terminal connector). An analog telephone can be attached to this arrangement whenever an ADU uses the standard building wiring. Table 15 lists the option settings used for data terminal and personal computer connections.

Table 15: DLDM screen settings for connection to data terminal or personal computer

Field on screen	Option	Comments
Speed	All speeds at which the Data Terminal or PC can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	yes	
Busy Out	no	Yes, if device is accessed through a hunt group
Permit Mismatch	yes	No, if device does not support XON/XOFF flow control
Parity	Same as DTE	
Dial Echoing	no	These devices can dial in the ASCII stream without human intervention
Disconnect Sequence	Long <BREAK>	-
Answer Text	no	These devices might not want to see any text
Connected Indication	-	Don’t care
Configuration	yes	

Host computers

A host computer can originate and terminate a data call. For this application, the number of DLCs required depends on the number of ports needed. An MADU can be used (instead of 8 ADUs) to complete the connection. Table 16 lists option settings for a port that has a terminating connection to a host computer or an originating connection from a host computer.

Note:

If the **KYBD Dialing** field is **n**, the rest of the option settings are irrelevant.

Table 16: DLDM screen settings for terminating connection to host computer

Field on screen	Option	Comments
Speed	All speeds at which the computer can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	-	Don't care
Permit Mismatch	-	Don't care
Parity	-	Don't care
Dial Echoing	-	Don't care
Disconnect Sequence	-	Don't care
Answer Text	-	Don't care
Connected Indication	-	Don't care
Configuration	-	Don't care

Field Descriptions for page 2 - Type 7500, WC-BRI, NI-BRI

This version of page 2 appears when **Type** is 7500, WC-BRI, and NI_BRI

Figure 315: 7500, World Class BRI, and NI-BRI Data Module screen

```
change data-module nn                                             Page  2  of  x
                                 DATA MODULE
  BRI LINK/MAINTENANCE PARAMETERS
                   XID? y      Fixed TEI? n       TEI: ___
           MIM Support? y     Endpt Init? y     SPID: 300_____    MIM Mtce/Mgt? y
```

BRI LINK/MAINTENANCE PARAMETERS

Country Protocol

Used with World Class BRI data modules. Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port. The following table lists the valid protocol entries. For a list of country codes, see the Country code table on page 1579.

Country/Area	Protocol
Australia	2
ETSI (Europe)	etsi
Japan	3
Singapore	6
United States (Bellcore National ISDN)	1

Endpt ID

Used with World Class BRI and NI-BRI data modules. Appears only if the **Endpt Init** field is **y**. This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires that the last 2 digits of the Service Profile Identifier (SPID) be between **00** and **63** and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires that the SPID programmed into the endpoint contain at least 9 digits. (For example, if the **SPID** field is **1234**, and the **Endpt ID** field is set to **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.)

Valid entries	Usage
00 to **62**	Enter a 2-digit number. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

Endpt Init

Used with 7500, World Class BRI, and NI-BRI Data Modules. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the Service Profile Identifier (SPID), administered into the system and entered into the ISDN-BRI terminal. For a ISDN-BRI terminal to become operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. This means that the SPID of the new or re-used terminals must be programmed to match the administered SPID value.

Valid entries	Usage
y/n	Indicates the terminal’s endpoint initialization capability.

Fixed TEI

Used with 7500, World Class BRI, and NI-BRI Data Modules. Used to indicate whether the endpoint has Fixed Terminal Equipment Identifier (TEI) capability. TEI identifies a unique access point within a service. For Fixed TEI stations, the TEI must be administered. Terminals with automatic TEI capability, the associated TEI is assigned by the system.

Valid entries	Usage
y/n	Enter **y** to indicate the endpoint has Fixed Terminal Equipment Identifier (TEI) capability.

MIM Mtce/Mgt

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Management Information Message (MIM) Support. Entering **y** indicates the terminal supports MIM Maintenance and Management capabilities, other than endpoint initialization.

MIM Support

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Used to support two types of capabilities: MIM endpoint initialization capability (SPID support), and other Maintenance/Management capability.

SPID

Used with 7500, World Class BRI, and NI-BRI Data Modules. Appears only if the **Endpt Init** field is **y**. The Service Profile Identifier (SPID) is a variable parameter of up to 10 digits. The SPID must be different for all terminals on the ISDN-BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first ISDN-BRI on a port, any other ISDN-BRI assignment to that port is blocked.

Valid entries	Usage
0 to **9999999999**	Assign a Service Profile Identifier (SPID) for this data module.

TEI

Used with 7500, World Class BRI, and NI-BRI Data Modules. Appears only if the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	Enter a 1 to 2-digit number.

XID

(Exchange identification) Used with 7500, World Class BRI, and NI-BRI Data Modules. Used to identify layer 2 XID testing capability.

Valid entries	Usage
y/n	Avaya recommends setting to **n**.

Date and Time

Use this screen to set the system date and time, to select the daylight savings plan number, if any, and to show whether the current time is standard time or daylight savings. Settings on this screen affect the internal clock and timestamp of the server running Avaya Communication Manager. You should update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

For additional information, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

Field descriptions for page 1

Figure 316: Date and Time screen

```
set time                                                               Page 1
                           DATE AND TIME
DATE
      Day of the Week: _______          Month: ________
     Day of the Month: __                Year: ____

TIME
     Hour: __ Minute: __      Second: __                 Type: ________________

            Daylight Savings Rule: _

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
         and MEASUREMENTS
```

Day of the Month

Valid entries	Usage
1 to **31**	Enter the current day of the month. The system clock uses this as the current date.

Day of the Week

Valid entries	Usage
Sunday through **Saturday**	Enter the current day of the week. The system clock uses this as the current day.

Daylight Savings Rule

This field displays which daylight savings rule is in use for your system.

Valid entries	Usage
0 to **15**	Enter the appropriate rule number. The system clock uses this as the current daylight savings rule. These rules are defined on the **Daylight Savings Rules** screen.

Hour

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the current hour to be used by the system clock.

Minute

Valid entries	Usage
0 to **59**	Enter the current minute. The system clock uses this as the current minute.

Month

Valid entries	Usage
January through **December**	Enter the current month. The system clock uses this as the current month.

Second

This display-only field shows the seconds and cannot be modified. It resets to zero when you save the information on this screen.

Type

Valid entries	Usage
daylight-savings	Enter **daylight-savings** to indicate daylight savings time is in effect.
standard	Enter **standard** to indicate standard time is in effect.

Year

Valid entries	Usage
1990 to **2099**	Enter the current year. The system clock uses this as the current year.

Related topics

To update the date and time for the change to or from daylight savings time, use the **Daylight Saving Rule** screen. See Chapter 1: System Basics for instructions on how to set up daylight savings rules.

Daylight Savings Rules

Use this screen to enter up to 15 customized daylight savings rules. You can specify the day, month, date, time, and increment each daylight savings rule goes into effect and the day, month, date, and time it stops. Rule 0 makes no adjustment to the system clock for daylight savings and cannot be modified. Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after March 8 at 2:00 a.m. with a 01:00 increment. Daylight Savings Time stops on the first Sunday on or after November 1 at 2:00 a.m., also with a 01:00 increment (used as a decrement when switching back to Standard time. Telephone displays are affected by these settings.

Field descriptions for page 1

Figure 317: Daylight Savings Rules screen

```
change daylight-savings-rules                               Page 1 of 2
DAYLIGHT SAVINGS RULES
Rule      Change Day                    Month  Date   Time    Increment

0:   No Daylight Savings

1:   Start: first __Sunday_ on or after _March___ 8_ at 2:00_    ___01:00__
     Stop:  first __Sunday_ on or after November 1_ at 2:00_
2:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
3:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
4:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
5:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
6:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
7:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
```

Change day (Start)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move ahead to begin daylight savings. If you enter **Day** in this field, the clock will change on the exact date entered in the next two fields.

Change day (Stop)

Valid entries	Usage
Sunday through **Saturday** or **Day**	Enter the day of the week you want the clock to move back to return to standard time. If you enter **Day** in this field, the clock will change on the exact date entered in the next two fields.

Date (Start)

Valid entries	Usage
0 to **31**	Enter the day of the month you want the clock to move ahead to begin daylight savings.

Date (Stop)

Valid entries	Usage
0 to **31**	Enter the date you want the clock to move back to return to standard time.

Increment (Start)

Valid entries	Usage
0 to **23**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.
0 to **9**	Enter the number of minutes you want the clock to move ahead for daylight savings and to move back to return to standard time.

Month (Start)

Valid entries	Usage
January through **December**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.

Month (Stop)

Valid entries	Usage
January through **December**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.

Rule

This display-only field indicates the daylight savings rule number.

Time (Start)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move ahead to begin daylight savings.
0 to 59	Enter the minute you want the clock to move ahead to begin daylight savings.

Time (Stop)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move back to return to standard time.
0 to **59**	Enter the minute you want the clock to move back to return to standard time.

DCS to QSIG TSC Gateway

Use the **DCS to QSIG TSC Gateway** screen to determine when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. This screen maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the **QSIG MWI-Prefix** screen. It also assigns the voice mail number to be used when a DCS served-user node interrogates a QSIG message center.

This screen only appears if the Interworking with **DCS** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Field descriptions for page 1

Figure 318: DCS to QSIG TSC Gateway screen

```
change isdn dcs-qsig-tsc-gateway                              Page 1 of x

                               DCS TO QSIG TSC GATEWAY

                                AAR/                                     AAR/
                                ARS                                      ARS
Mach  Sig  TSC    Voice Mail   Access    Mach  Sig  TSC    VoiceMail    Access
 ID   Grp  Index   Number       Code      ID   Grp  Index   Number      Code
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
 __   __   ___   ____________   _____    ___   ___  ____   _________    ____
```

AAR/ARS Access Code

This field can be left blank.

Valid entries	Usage
0 to **9**, *, #	Enter up to 4-digit access code.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the **Processor Channel** screen.

Sig Grp

You must complete the **Signaling Group** field for each machine ID.

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number between 1 and 110 for DEFINITY CSI.
1 to **650**	Enter the assigned signaling group number between 1 and 650 for S8300/S87XX Servers.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the **Signaling Group** screen.

Voice Mail Number

This field can be left blank.

Valid entries	Usage
0 to **9**	Enter the complete Voice Mail Dial Up number up to 15 digits.

Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. This screen enables you to determine the beginning digits and total length for each type of call that Avaya Communication Manager needs to interpret. The **Dial Plan Analysis Table** and the **Dial Plan Parameters** screen work together to define your system's dial plan.

Note:

In Communication Manager 5.0 and later, you can administer dial plans per-location. Typing the command **`change dialplan analysis n`** displays the all-locations **Dial Plan Analysis** screen. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change dialplan analysis location n`**, where ***n*** represents the number of a specific location. For details on command options, see online help, or *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Field descriptions for page 1

Figure 319: Dial Plan Analysis Table screen

```
change dialplan analysis                                          Page 1 of x
                              DIAL PLAN ANALYSIS TABLE
                                    Location: All           Percent Full: 7

       Dialed  Total  Call          Dialed  Total  Call         Dialed  Total  Call
       String  Length Type          String  Length Type         String  Length Type
         00       2    attd
         1        3    dac
         2        4    ext
         3        4    ext
         3        1    aar
         4        1    ars
         4        5    ext
         5        5    ext
         5        7    ext
         6        5    ext
         7210     7    ext
         8        7    ext
         9        1    fac
         *        3    fac
         #        3    fac
```

Call Type

Valid entries	Usage
aar	Automatic Alternate Routing — Used to route calls within your company over your own private network. In order to use this code in your dial plan, the ARS/AAR Dialing without FAC feature must be enabled on the **System Parameters Customer-Options (Optional Features)** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type aar, as soon as the dialed digits have reached the administered length, the digits are treated as if an AAR feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the AAR Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 930, extensions of 3xxx cannot be dialed directly. Whenever a user dials the first digit of 3, the system immediately interprets the dialed string as an AAR string and transfers control to AAR. Extensions of 3xxx can only be accessed using AAR Digit Conversion. That is, you must dial a longer AAR number from which AAR Digit Conversion deletes leading digits to form a number of the form 3xxx.
ars	Automatic Route Selection — Used to route calls that go outside your company over public networks. ARS is also used to route calls to remote company locations if you do not have a private network. In order to use this code in your dial plan, the ARS/AAR Dialing without FAC feature must be enabled on the **System Parameters Customer-Options (Optional Features)** screen. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.) When dialing digits of Call Type ars, as soon as the dialed digits have reached the administered length, the digits are treated as if an ARS feature access code (FAC) was dialed. Control is transferred and the digits are routed according to the ARS Analysis and Digit Conversion forms. In the example shown on the Dial Plan Analysis Table on page 930, extensions of 4xxxx cannot be dialed directly. Whenever a user dials the first digit of 4, the system immediately interprets the dialed string as an ARS string and transfers control to ARS. Extensions of 4xxxx can only be accessed using ARS Digit Conversion. That is, you must dial a longer ARS number from which ARS Digit Conversion deletes leading digits to form a number of the form 4xxxx.
attd	Attendant — Defines how users call an attendant. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code. Beginning with the November 2003 release of Communication Manager (2.0), you can also administer the attendant access code by entering an appropriate **fac** or **dac** entry on the **Dial Plan Analysis** screen, and then entering the actual access code on the **Feature Access Code (FAC)** screen. Location-specific attendant access codes can be administered on the **Locations** screen.
	1 of 3

Valid entries	Usage
dac	Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. Dial access codes can start with any number from 0 to 9, * or # and can contain up to 4 digits. If an extension entry and a DAC entry have the same Dialed String, the extension entry can be longer than the DAC entry only if all of the trunk groups covered by that DAC entry have **Dial Access** on the **Trunk Group** screen set to **n**. You can use the DAC to activate or deactivate a Communication Manager feature or to seize a trunk from a trunk group, or both. In the first case, the DAC functions as a FAC, in the second as a TAC. For example, you can define the group 300 to 399 for dial access codes, and allow both FAC and TAC in that range. You can use 4-digit DACs for ordinary trunk access, but they do not work for attendant control of trunk groups, trunk-ID buttons, or DCS, and only the last 3 digits of the codes can be recorded in CDR records. See also the description below for **fac**.
ext	Primary extension — Defines extension ranges that can be used on your system. Extension can have a first digit of 0 through 9 and can be 1 to 7 digits in length. Extension cannot have the same first digit as a 1-digit ARS or AAR feature access code (FAC). When a dial plan has mixed station numbering, extensions of various lengths (all with the same first digit) are mapped on the Dial Plan Analysis table. The system then employs an inter-digit time-out to ensure that all dialed digits are collected.
fac	Feature access code only — A FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. Avaya recommends that a FAC have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems might occur when, for example, 3-digit FACs and 4-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it *must* have the longest total length in the dial plan.
	2 of 3

Valid entries	Usage
pext	Prefixed extension — Is made up of a prefix (first digit) that can be a **0** to **9** (* and # not allowed) and an extension number of up to 5 digits in length. The maximum length of a prefix and extension combination is 6 digits. You cannot administer a dial access code with the same first digit as a prefixed extension. The purpose of the prefix is to identify the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits (extension number) are then processed. A prefixed extension allows the use of extensions numbers with any dialed string (the extension length must be specified on the table). The "prefixed extension" cannot have the same dialed string as the ARS or AAR facility access code (FAC).
udp	Works identically to **ext**, with this exception: ● If dialed digits match the **Call Type udp**, Communication Manager automatically checks the UDP Table first to see if there is a match, regardless of the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen. If there is no match, Communication Manager then checks the local server. ● If dialed digits match the **Call Type** of **ext**, Communication Manager checks the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen. - If the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen is **udp-table-first**, Communication Manager checks the UDP Table first to see if there is a match. If there is no match, Communication Manager then checks the local server. - If the value in the **UDP Extension Search Order** field on the **Dial Plan Parameters** screen is **local-extensions-first**, Communication Manager checks the local server first to see if there is a match. If there is no match, Communication Manager then checks the UDP Table. **Note**: The **udp Call Type** allows Communication Manager to recognize strings of 14 and 15 digits, which are longer than the maximum extension length of 13 digits. However, **udp** can be used with any length.
	3 of 3

Dialed String

The dialed string contains the digits that Avaya Communication Manager will analyze to determine how to process the call. This field allows you to enter up to four digits, so you can allocate blocks of 1000 numbers even when using a 7-digit dial plan

.

Valid entries	Usage
0 to **9**, * and **#**	Enter any combination of 1 to 4 digits. the following restrictions apply: ● The digits * and # can only be used as first digits, and only for the Call Types **fac** and **dac**. ● For Call Type **attd**, if the Total Length is 2, the Dialed String must be 2 digits long. ● Two Dial Plan entries can use the same Dialed String only if the Dialed String is 1 digit long. Longer Dialed Strings must all be unique. ● A new entry cannot be administered if it causes an existing extension, feature access code, or trunk access code to become inaccessible.

Location

This is a display-only field. Typing the command **`change dialplan analysis`** displays the all-locations screen, and populates this field with **all**. The ***n*** specifies that dialed strings beginning with the value **`n`** are displayed first. To access a per-location screen, type **`change dialplan analysis location`** ***n***, where ***n*** represents the number of a specific location. This field then displays the number of the specified location. For details on command options, see online help, or Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers, 03-300431.

Valid entries	Usage
1 to **64**	Defines the location of the server running Avaya Communication Manager that uses this Dial Plan Analysis Table. On the System Parameters Customer-Options (Optional Features) screen, the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this Dial Plan Analysis Table is the default for all port network (cabinet) locations. Appears only if the **Multiple Locations** field is **n** on the System Parameters Customer-Options (Optional Features) screen.

Percent Full

Displays the percentage (0 to 100) of the system's memory resources that have been allocated for the dial plan that are currently being used.

Total Length

Valid entries	Usage
1 to **2** for **attd** **1** to **4** for **dac** **1** to **4** for **fac** **1** to **7** for **ext** **2** to **6** for **pext**	Enter the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

Dial Plan Parameters

The **Dial Plan Parameters** screen works with the **Dial Plan Analysis Table** to define your system's dial plan.

It also controls the appearance of digit extensions on station displays. These multi-digit extensions can be hard to read when displayed as a block. Avaya Communication Manager allows you to select the display format for 6-13 digit extensions.

Field descriptions for page 1

Figure 320: Dial Plan Parameters screen

```
change dialplan parameters                                          Page 1 of x
                              DIAL PLAN PARAMETERS

                   Local Node Number: 2
                     ETA Node Number:
                 ETA Routing Pattern:
          UDP Extension Search Order: local-extensions-first
        AAR/ARS Internal Call Prefix:
  AAR/ARS Internal Call Total Length:
Retry ARS Analysis if All-Location Entry Inaccessible? n

EXTENSION DISPLAY FORMATS

                           Inter-Location/SAT          Intra-Location
 6-Digit Extension:        xx.xx.xx                    xx.xx.xx
 7-Digit Extension:        xxx-xxxx                    xxx-xxxx
 8-Digit Extension:        xx.xx.xx.xx                 xx.xx.xx.xx
 9-Digit Extension:        xxx-xxx-xxx                 xxx-xxx-xxx_
10-Digit Extension:        xxx-xxx-xxxx                xxx-xxx-xxxx_
11-Digit Extension:        xxxx-xxx-xxxx               xxxx-xxx-xxxx
12-Digit Extension:        xxxx-xxx-xxxx               xxxxxxxx-xxxx
13-Digit Extension:        xxxxxxxxxxxxx               xxxxxxxxxxxxx
```

AAR/ARS Internal Call Prefix

The digits entered in this field are concatenated with the calling or called extension. Appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ARS/**

AAR Dialing Without FAC field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
0 to **9** up to 8 digits**,** or blank	Enter a string from 1 to 8 digits long (not including * or #).

AAR/ARS Internal Call Total Length

The total length of the internal call digit string. Appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ARS/AAR Dialing Without FAC** field is **y**. (Contact your Avaya technical support representative to discuss the ARS/AAR Dialing Without FAC feature before enabling it.)

Valid entries	Usage
6 to **10** or blank	Enter the total length of the internal call digit string, which includes the Internal Call Prefix and the calling or called extension.

Note:

If either the **AAR/ARS Internal Call Prefix** or the **AAR/ARS Internal Call Total Length** field is non-blank and valid, the other must also be non-blank and valid. In addition, the longest extension length on the **Dial Plan Analysis** screen, plus the length of the **ARS/AAR Internal Call Prefix**, must equal or be greater than, the **ARS/AAR Internal Call Total Length** value.

ETA Node Number

Enter the number of the destination server for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Valid entries	Usage
1 to **999** or blank	Enter the number of a destination server.

ETA Routing Pattern

Enter the number of the routing pattern to reach the destination server.

Valid entries	Usage
1 to **640** or blank	Avaya DEFINITY ServerCSI.
1 to **999** or blank	Avaya S8300 and S87XX Servers.

Local Node Number

Enter a number to identify a specific node in a server network. This entry must match the DCS switch node number and the CDR node number if they are specified.

Valid entries	Usage
1 to **63**	Enter the number of a specific node in a network.
blank	Can be blank if automatic restoration, DCS, and CDR are not used.

Retry ARS Analysis if All-Location Entry Inaccessible

This field appears on the **Dial Plan Parameters** screen in Communication Manager Release 4.0.x or later.

Valid entries	Usage
y	The system finds and uses the best possible entry in the per-location ARS table, if the all-location table points to a trunk group that cannot be accessed because the network has fragmented.
n	The system does not retry ARS analysis when a trunk group cannot be accessed because the network has fragmented.

UDP Extension Search Order

Appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Uniform Dialing Plan** field is **y**. Specifies the first table to search to match a dialed extension. If the dialed extension is not found in the specified place, it then is searched for in the "other" place.

Valid entries	Usage
local-extensions -first	Searches the local server first to match a dialed extension; if not found, then uses the UDP tables to route the call.
udp-table-first	Searches the UDP tables for an off-switch conversion; if not found, then searches the local server for the dialed extension.

EXTENSION DISPLAY FORMATS

Extension Display Format (6-13 digits)

Use these fields to specify how the system punctuates extensions for display. The punctuation field is divided into two columns, one for **Inter-Location/SAT** displays, and one for **Intra-Location** displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (**.**) are used on SAT screens in place of blanks. The following table gives the maximum number of punctuation marks permitted for each extension length:

Note:

The number of punctuation marks that the system allows is determined by the number of “x”s in the format:

- If the format contains fewer than 6 x's, no punctuation marks can be entered.

- For 6 or more x's, the maximum number of punctuation marks is determined by the following table.

Extension Length	Max Punctuation Marks	Max Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid entries	Usage
xx.xx.xx	This field can contain all ‘x’ characters (no punctuation) or you can use a combination of ‘x’ characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions will display. If the format contains fewer than 6 x's, no punctuation marks can be entered. You must specify a format or accept the default. You cannot leave this field blank. The default values for the 8-, 9-, 10-, 11-, 12-, and 13-digit fields are those shown in Figure 320.

Digit Absorption

This screen implements up to 5 digit absorption lists. The screen might be required for each CO and FX trunk group connected to a step-by-step CO. Each outgoing digit string from the server running Communication Manager to the step-by-step CO is treated according to entries in the "Absorption Treatment Assignment" section of the screen.

Note:

If the **Digits** field on the **Trunk Group** screen is blank, you cannot administer Digit Absorption.

Field descriptions for page 1

Figure 321: Digit Absorption screen

```
change digit absorption                                          Page 1 of x
                              DIGIT ABSORPTION

                           List Number: __
  ABSORPTION TREATMENT INFORMATION (All selections must be from same group)
                Choice      Meaning
    Group I.      A       Digit not absorbed.
                  B       Digit absorbed repeatedly.
                  C       Digit absorbed once with no further absorption.
    Group II.     A       Digit not absorbed.
                  D       Digit absorbed only if it is the first digit.
                  E       Digit absorbed only if it is the second digit and
                          the first digit was already absorbed.
                  F       Digit asorbed only if it is the first or second digit.

  ABSORPTION TREATMENT ASSIGNMENT (select treatment (A-F) for each digit below)
          0: A          2: A          4: A          6: A          8: A
          1: A          3: A          5: A          7: A          9: A
```

Absorption Treatment Assignment

Valid entries	Usage
A to **F**	Enter a desired treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Absorption Treatment Information

This is a display-only section. It shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II. Enter the assignment on the next section on the screen.

List Number

A display-only field indicating the Digit Absorption List number (**0** to **4**). The list number is referenced from a field entry on the associated trunk group.

Display Parameters

Use this screen to establish how extensions of 6 to 13 digits are punctuated. There are 26 possible **Display Parameters** screens, numbered from 1-25. This screen is linked to the value that is entered in the **Display Parameters** (**Disp Parm**) field on the **Locations** screen.

Field descriptions for page 1

Figure 322: Display Parameters screen

```
 change display-parameters 5                                       Page 1 of 1
                               DISPLAY PARAMETERS

EXTENSION DISPLAY FORMATS

   Note: If a format is blank, the corresponding format administered
         on the Dial Plan Parameters form will be used

                               Inter-Location       Intra-Location
        6-Digit Extension:     ________             ________
        7-Digit Extension:     ________             ________
        8-Digit Extension:     ___________          ___________
        9-Digit Extension:     ____________         ____________
       10-Digit Extension:     _____________        _____________
       11-Digit Extension:     _____________        _____________
       12-Digit Extension:     _____________        _____________
       13-Digit Extension:     _____________        _____________

       Default Call Appearance Display Format: inter-location
```

EXTENSION DISPLAY FORMATS

The fields in this section of the **Display Parameters** screen override similar fields on the **Dial Plan Parameters** screen. If you leave these fields on the **Display Parameters** screen blank, the values on the **Dial Plan Parameters** screen apply.

Extension Display Format (6-13 digits)

Use these fields to specify how the system punctuates extensions for display. The punctuation field is divided into two columns, one for **Inter-Location** displays, and one for **Intra-Location** displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (**.**) are used on SAT screens in place of blanks.

Note that he number of punctuation marks that the system allows is determined by the number of “x”s in the format:

- If the format contains fewer than 6 x's, no punctuation marks can be entered.
- For 6 or more x's, the maximum number of punctuation marks is determined by the following table.

The following table gives the maximum number of punctuation marks permitted for each extension length:

Extension Length	Max Punctuation Marks	Max Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid entries	Usage
xx.xx.xx or blank	This field can contain all ‘x’ characters (no punctuation) or you can use a combination of ‘x’ characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions will display. If the format contains fewer than 6 x's, no punctuation marks can be entered. The default is blank.

Default Call Appearance Display Format

This field only affects call appearances on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances are not affected by this field.

Valid entries	Usage
inter-location	The system displays the complete extension on downloadable call appearance buttons. This is the default.
intra-location	The system displays a shortened version of the extension on downloadable call appearance buttons.

Inter-Location

Use this field to specify punctuation for calls between locations. This is the default.

Intra-Location

Use this field to specify punctuation for calls within a location.

Valid entries	Usage
y	Enter **y** when the **Signaling Mode** field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

DS1 Circuit Pack

Use this screen to administer all DS1 circuit packs.

Field descriptions for page 1

Figure 323: DS1 Circuit Pack screen

```
add ds1 nnnn                                                    Page 1 of x
                                DS1 CIRCUIT PACK

             Location: _____                                Name: _______________
             Bit Rate: _____                         Line Coding: ____
    Line Compensation: _                            Framing Mode: ___
       Signaling Mode: ___________                     D-Channel: __
              Connect: _________                       Interface: ___________
         Interconnect: ___                         Peer Protocol: ________
                                                Country Protocol: ____
                                                Protocol Version: _
Interface Companding: _____                                 CRC? _
            Idle Code: ________
                                 DCP/Analog Bearer Capability: ________
                                            T303  Timer(sec):
                                           Disable Restarts?:

    MMI Cabling Board: _____  MMI Interface: ESM

MAINTENANCE PARAMETERS

       Slip Detection? _                   Near-end CSU Type: ________
                                             Block Progress Indicator? n
```

Figure 324: DS1 Circuit Pack screen for Croatia and South Africa

```
add ds1 nnnn                                                Page 1 of x
                                DS1 CIRCUIT PACK

             Location: ______                        Name: ____________
             Bit Rate: ______                 Line Coding: ____

       Signaling Mode: __________

         Interconnect: ____                Country Protocol: ____

 Interface Companding: ____
            Idle Code: ____

               Received Digital Metering Pulse Minimum (ms):
               Received Digital Metering Pulse Maximum (ms):
                      Received Digital Metering Pulse Value:
       Slip Detection: __               Near-end CSU Type: ______
                                              Block Progress Indicator? n
```

The following screen is valid *only* for the TN2242.

Figure 325: DS1 Circuit Pack screen for Channel Associated Signaling

```
add ds1 nnnn                                                Page 1 of x
                                DS1 CIRCUIT PACK

              Location: 01A13                         Name: _______________
              Bit Rate: 2.048                  Line Coding: cmi

        Signaling Mode: CAS
          Interconnect: pbx
                                            Country Protocol: 3

  Interface Companding: mulaw
             Idle Code: 11111111



 MAINTENANCE PARAMETERS

        Slip Detection? n
```

Bit Rate

Use this field to select the maximum transmission rate for DS1 circuit packs that support either T-1 or E-1 service. For circuit packs that only support one of these services, the field is a display-only field.

Note:

Once an **`add ds1`** operation is complete (that is, the **DS1** screen has been submitted) you can't change the **Bit Rate** field with a **`change ds1`** command. Instead, execute a **`remove ds1`** command. Then use the **`add ds1`** command to administer the circuit pack again. You'll have to re-enter all the information for the circuit pack.

TN464C (and later release) circuit packs have an option switch that must be set to match the entry in the **Bit Rate** field.

Valid entries	Usage
1.544	Use for T-1 service.
2.048	Use for E-1 service.

Channel Numbering

The ETSI and ISO QSIG specifications require that B-channels on an E1 be encoded as 1 to 30 in the Channel ID IE. Prior to the existence of this field, Avaya Communication Manager only used this scheme for Country Protocols 2a (Australia) and 13a (Germany 1TR6). This field appears when the **Signaling Mode** field is **isdn-pri**, the **Bit Rate** field is **2.048**, the **Connect** field is **pbx**, and the **Interface** field is **peer-master** or **peer-slave**.

Valid entries	Usage
timeslot	
sequential	If Avaya Communication Manager is connected via QSIG trunks to a switch/server supporting the ETSI QSIG or ISO QSIG specifications, this field must be **sequential**. When the **Signaling Mode** field is **isdn-pri** and the **Bit Rate** field is **2.048**, but the **Channel Numbering** field does not display because of the setting of other fields, it is set internally to **sequential** for 2a (Australia) and 13a (Germany).

Connect

In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this DS1 link. This field only appears when the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
pbx	Enter **pbx** if this DS1 link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.
line-side	Enter **line-side** when Communication Manager is acting as the network side of an ISDN-PRI interface. Use **line-side** to connect to Roll About Video equipment.
network	Enter **network** when the DS1 link connects Communication Manager to a central office or any other public network switch.
host	Enter **host** when the DS1 link connects Communication Manager to a computer.

Country Protocol

The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using.

This field appears if the **Signaling Mode** field is **CAS** or **isdn-pri**. For the Japanese 2Mbit trunk circuit pack, this is a display-only field if the **Signaling Mode** field is **CAS**.

Note:

For a list of country codes, see the Country code table on page 1579.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

CRC

This field indicates whether a cyclic redundancy check (CRC) will be performed on transmissions that the DS1 circuit pack receives. This field does not display for all circuit packs.

Valid entries	Usage
y	Enter **y** when the **Signaling Mode** field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

D-Channel

The Japanese 2Mbit trunk circuit pack, when administered to support ISDN-PRI signaling, allows you to assign the D-channel to any channel from 1 to 31 in an E-1 facility. You cannot submit the screen if this field is blank. Using the `change ds1` command, you can change this field if the D-channel is not used in a signaling group. This field appears only when the **Location** field indicates the circuit pack is a Japanese 2Mbit trunk circuit pack and the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
1 to **31**	Enter the number of the channel that will be used as the D-channel.

DCP/ANALOG Bearer Capability

This field appears when the **Signaling Mode** field is **isdn-pri**. This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**.

Valid entries	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability.
speech	Provides speech encoding in the information transfer capability.

Disable Restarts

Use this field to control whether outgoing RESTART messages are sent. This field appears when one of the following is true:

- **Country Protocol** is **3 (Japan)**
- **Country Protocol** is **ETSI**
- **Peer Protocol** is **QSIG**

This field and the **Protocol Version** field are mutually exclusive. Only one of the fields can be displayed. You can also use this field to disable QSIG restarts.

Valid entries	Usage
y	Outgoing restarts are disabled, i.e., RESTART messages are not sent.
n	Outgoing RESTART messages are sent. This is the default.

DMI-BOS

The DMI/BOS protocol is used for high-speed digital communications between a host computer and Avaya Communication Manager. With this 24-channel protocol, channels 1 to 23 of the DS1 link carry data and channel 24 carries control signaling. DMI/BOS has greater capacity than a robbed-bit 24-channel facility. This field appears only when the **Signaling Mode** field is **common-chan**.

Valid entries	Usage
y	Enter **y** to activate the Digital Multiplexed Interface-Bit Oriented Signaling (DMI-BOS) format.
n	Enter **n** to use an Avaya proprietary format.

Framing Mode

Use this field to select either superframe (sf or d4) or extended superframe (esf) for T1 service on the DS1 link. The framing mode you use must match the mode used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

This field only appears if the **Bit Rate** field is **1.544** (that is, if you're using T-1 service). If you're using E-1 service, Avaya Communication Manager automatically selects CEPT1 framing.

Tip:

Avaya recommends using ESF when your service provider supports it, especially if you might someday upgrade the facility to ISDN. The ESF format provides enhanced performance measurements and uses a sophisticated error-checking method to ensure data integrity.

Valid entries	Usage
d4	Enter **d4** to use the basic DS1 superframe (sf). Avaya recommends this mode only for voice traffic.
esf	Enter **esf** to use the Extended Superframe format. Avaya recommends this mode for digital data traffic. If you enter **esf** for a TN464F, TN767E, or a later suffix DS1 circuit pack, a second page of the **DS1 Circuit Pack** screen becomes available to administer ESF Data Link options.

Idle Code

CAUTION:

Customers: The entry in the **Country Protocol** field sets the default idle code. Do not change the default without assistance from Avaya or your network services provider.

For some circuit packs, this is a display-only field.

Valid entries	Usage
any 8-digit string of 0's and 1's	This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Interconnect

For E-1 service using channel-associated signaling, the entry in this field tells Avaya Communication Manager whether the DS1 circuit pack is using a public or private network protocol. The entry in this field must agree with the entry in the **Group Type** field on the **Trunk Group** screen. This field appears only when the **Signaling Mode** field is **CAS**.

Valid entries	Usage
pbx	If **pbx** is selected, the board operates as a tie trunk circuit pack.
CO	If **CO** is selected, the board operates as a CO or DID circuit pack. Use **CO** for Enterprise Mobility User (EMU)/EC500 administration.

Interface

This field only appears when the **Connect** field is **pbx**; that is, when this DS1 link is providing an ISDN-PRI connection in a private network. The **Interface** field controls how your server negotiates glare with the far-end switch. The servers at either end of the DS1 link must have complementary settings in this field: if not, the D-channel won't even come up. For example, if the Avaya S8XXX Server at one end of the link is administered as **network**, the other end must be administered as **user**.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

The entry in this field must match the companding method used by the far-end switch. This field does not appear for all DS1 circuit packs.

Valid entries	Usage
alaw	Enter **alaw** for E-1 service.
mulaw	Enter **mulaw** for T-1 service.

Interworking Message

This field determines what message the server sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

ITN-C7 Long Timers

This field controls the T302 and T303 timers. It only appears if the **Signaling Mode** field is **isdn-pri**.

Valid entries	Usage
y	Use if you want to increase the length of the long timers.
n	Leave n if you want to use the default long timers.

Line Coding

This field selects the type of line coding used on this facility. The setting in this field must match the setting on the far-end of the link, or you must have an intervening CSU to convert the line coding protocols. Voice calls will work even if line coding does not match, but a single data call will bring down the DS1 facility. For the TTC 2Mb CMI Trunk circuit pack, this is a display-only field showing **cmi** (coded mark inversion).

The following information is for reference. Talk with your network service provider or your Avaya technical support representative to find the appropriate protocol for your application.

CAUTION:

If you change this field, you must busy out the DS1 circuit pack. You must also change the following screens: **Route Pattern***,* **Access Endpoint***,* **PRI Endpoint***,* **Signaling Group***, and* **Trunk Group***.*

Note:

When the DS1 circuit pack is used for ISDN service, the ISDN D-channel data is inverted when **ami-basic** or **ami-zcs** is entered and not inverted when **b8zs** or **hdb3** is entered.

Valid entries	Usage
b8zs (bipolar eight zero substitution)	Enter **b8zs** for T-1 facilities that support voice and/or data traffic. Enter **b8zs** if you need a 64K clear channel.
ami-zcs (alternate mark inversion - zero code suppression)	Enter **ami-zcs** only for T-1 facilities that carry voice traffic: Avaya does not recommend this for digital-data applications. If you anticipate upgrading this facility to ISDN, use b8zs line coding if possible.
ami-basic (alternate mark inversion-basic)	Enter **ami-basic** for unrestricted E-1 facilities.
hdb3 (high density bipolar 3)	Enter **hdb3** for restricted E-1 facilities.
cmi (coded mark inversion)	Used in Japan, **cmi** is the only type of line coding you can use with the Japanese 2Mbit trunk circuit pack. This field becomes a display-only field when you are administering the Japanese 2Mbit trunk circuit pack.

Line Compensation

The appropriate entry in this field varies with the type of cable used, so work with your network service provider to determine the correct setting in your situation. The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable terminated on a DSX-1 cross-connect.

Valid entries	Usage
1	Length: 000 to 133 (ft), 000 to 40.5 (m)
2	Length: 133 to 266 (ft), 40.5 to 81.0 (m)
3	Length: 266 to 399 (ft), 81.0 to 122 (m)
4	Length: 399 to 533 (ft), 122 to 163 (m)
5	Length: 533 to 655 (ft), 163 to 200 (m)

The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable directly connecting to DS1 interfaces.

Valid entries	Usage
1	Length: 0000 to 0266(ft), 000 to 081(m)
2	Length: 0266 to 0532(ft), 081 to 162(m)
3	Length: 0532 to 0798(ft), 162 to 243(m)
4	Length: 0798 to 1066(ft), 243 to 325(m)
5	Length: 1066 to 1310(ft), 325 to 400(m)

Location

This display-only field shows the port address specified in the `add` command when the circuit pack was first administered.

MMI Cabling Board

This field appears only if the **MMCH** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
slot address (cabinet, carrier, slot)	Enter the slot location (cabinet, carrier, slot) of the multimedia interface circuit pack that is connected to the Expansion Services Module (ESM).

MMI Interface

This display-only field appears if the **MMCH** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen and there is a value in the **MMI Cabling Board** field.

Name

Use this field to assign a significant, descriptive name to the DS1 link. Avaya recommends putting the vendor's circuit ID for the link in this field, because that information helps you troubleshoot problems with the link, but you could also use this field to indicate the function or the destination of this DS1 facility. In that case, put the DS1 link circuit ID in the **Name** field of the trunk group associated with this link.

Valid entries	Usage
1 to 15 characters	Enter a name for the DS1 link. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Peer Protocol

This allows you to administer the peer level protocol that will operate in a private network. This field appears if the **Interface** field is **peer-master** or **peer-slave**. To enter **Q-SIG**, the **Basic Call Setup** field on the **System Parameters Customer-Options (Optional Features)** screen must be **y**

Valid entries	Usage
Q-SIG	This implements QSIG Network Basic Call.
TTC	For private networking. Requires a Digital Trunk (Japan 2 MB TTC) (TN2242) circuit pack.

Protocol Version

In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, this field selects the protocol that matches your network service provider's protocol. See Public network signaling administration for ISDN-PRI Layer 3 on page 957 to see which countries support which protocols.

This field appears only when:

- The **Signaling Mode** field is **isdn-pri** and the **Connect** field is **network.**

- The **Signaling Mode** field is **isdn-pri,** the **Connect** field is **pbx**, and the **Interface** field is **user** or **network**.

Valid entries	Usage
a, b, c, d	The entry in this field must match the protocol used by your network service provider, so work with your vendor to determine the appropriate entry.

WARNING:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1** screen, you cannot set the **Send Connected Number** field to r (restricted) on the **ISDN-PRI Trunk Group** screen, as this causes display problems.

Public network signaling administration for ISDN-PRI Layer 3

The table below shows Avaya Communication Manager public-network access connections for ISDN-PRI Layer 3.

Admin value	Country	Protocol supported	B-channel mtce msg
1-a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1-b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607	Restart
1-c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United States	Telecordia SR-4287	Service
2-a	Australia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2-b	Australia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
			1 of 2

Admin value	Country	Protocol supported	B-channel mtce msg
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/ DPNSS through external converter)	Restart
10-b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	None
11	Spain	Telefonica ISDN Specification	Restart
12-a	France	VN4 (French National PRI)	None
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13-a	Germany	FTZ 1 TR 6 (German National PRI)	None
13-b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI-a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI-b		ETS 300 102	None
			2 of 2

Received Digital Metering Pulse Maximum (ms)

This field appears only when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 960. The default value depends on the **Country Protocol** **f**ield's entry. For a list of country codes, see the Country code table on page 1579.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be greater than the **Received Digital Metering Pulse Minimum** field.

Received Digital Metering Pulse Minimum (ms)

This field appears only when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Interconnect** field is **co** or **pbx**, and the **Country Protocol** field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Incoming digital PPM signaling default per Country Protocol code on page 960. The default value depends on the **Country Protocol** field's entry. For a list of country codes, see the Country code table on page 1579.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be less than the **Received Digital Metering Pulse Maximum** field.

Received Digital Metering Pulse Value

This field appears when the **Signal Mode** field is **cas** (Channel Associated Signaling), the **Country Protocol** field is **21**, and the **Interconnect** field is **co** or **pbx**.

Valid entries	Usage
0, **1**	Work with your network services provider to determine the appropriate entry.

Table 17: Incoming digital PPM signaling default per Country Protocol code

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
0	null	NA	NA	NA
1	U.S.	NA	NA	NA
2	Australia	80	180	0
3	Japan	NA	NA	NA
4	Italy	120	150	1
5	Netherlands	90	160	0
6	Singapore	NA	NA	NA
7	Mexico	20	180	1
8	Belgium	20	180	1
9	Saudi Arabia	NA	NA	NA
10	UK	NA	NA	NA
11	Spain	20	220	0
12	France	NA	NA	NA
13	Germany	NA	NA	NA
14	Czech Republic	20	420	1
15	Russia CIS	NA	NA	NA
16	Argentina	10	180	1
17	Greece	100	180	1
18	China	NA	NA	NA
				1 of 2

Table 17: Incoming digital PPM signaling default per Country Protocol code (continued)

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
19	Hong Kong	NA	NA	NA
20	Thailand	20	180	1
21	Macedonia Croatia	120 20	180 80	1 1
22	Poland	100	150	0
23	Brazil	NA	NA	NA
24	Nordic	NA	NA	NA
25	South Africa	160	240	0, 1
				2 of 2

Side

This field controls how your server running Communication Manager resolves glare at layer 3 over an ISDN-PRI link in QSIG private networks. It appears if the **Interface** field is **peer-master** or **peer-slave**.

The default value of the field changes depending upon which value the **Interface** field contains.

CAUTION:

It is critical that administration on this server correctly pairs with the administration of the far-end switch/server. If the far-end is administered as the b side, this field should be set to a regardless of whether the layer 2 designation is peer-master or peer-slave, and vice versa.

Valid entries	Usage
a	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).
b	Enter **b** if the **Interface** field is **peer-slave** (this server releases the contested circuit and looks for another when glare occurs).

Signaling Mode

This field selects the signaling method used for the DS1 link. This mode must match the method used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

Valid entries	Usage
CAS (Channel Associated Signaling)	Enter **CAS** for out-of band signaling with E-1 service. This setting yields 30 64-kbps B-channels for voice or data transmission. Channel 0 is used for framing while channel 16 carries signaling. Enter CAS for Enterprise Mobility User (EMU)/EC500 administration.
robbed-bit	Enter **robbed-bit** for in-band signaling with T-1 service. This setting yields 24 56-kbps B-channels for voice transmission.
isdn-pri	Enter **isdn-pri** for either T-1 or E-1 ISDN service. This setting supports both Facility Associated Signaling and Non-Facility Associated Signaling.
isdn-ext	Enter **isdn-ext** for either T-1 or E-1 ISDN service. This setting supports only Non-Facility Associated Signaling. **Note**: NFAS is primarily a feature for ISDN-T1 connections offered by service providers in North America and Hong Kong. However, it can also be used on private-network connections, and in that context it is possible to set up NFAS using ISDN-E1 interfaces.
common-chan	Enter **common-chan**, for out-of-band signaling with T-1 service. This setting yields 23 64-kbps B-channels for voice or data transmission. Channel 24 is used for signaling.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
2 to **10**	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

MAINTENANCE PARAMETERS

Alarm When PRI Endpoint Detached

Use this field for DS1 circuit packs connected to Roll-About Video equipment. This field appears only when the **Connect** field is **line-side**.

Valid entries	Usage
y/n	Enter **y** if you want the server running Avaya Communication Manager to generate an alarm when the DS1 board detects a loss of signal (for example, if the video equipment is disconnected).

EC Configuration

Appears when **Echo Cancellation** is **y** on the **DS1 Circuit Pack** screen.

Valid entries	Usage
1 to **15**	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

EC Direction

Direction of echo cancellation. Appears when **Echo Cancellation** is **y** on the **DS1 Circuit Pack** screen.

Valid entries	Usage
inward/outward	Enter **a** if the **Interface** field is **peer-master** (this server overrides the far-end when glare occurs).

Echo Cancellation

Appears when **DS1 Echo Cancellation** is **y** on the **System Parameters Customer-Options (Optional Features)** screen and circuit packs support echo cancellation.

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation.

Near-end CSU Type

This field appears only when the DS1 circuit pack is a TN767D or TN464E or later suffix model, the **Bit Rate** field is **1.544**, and the **Country Protocol** field is **1** (U.S.) or **3** (Japan). This field does not display for all circuit packs.

Valid entries	Usage
other	Enter **other** if no channel service unit is attached to the DS1 facility or if the CSU is an external unit. No options are available on page 2 for administering an external CSU.
integrated	Enter **integrated** if a 120A CSU module is attached to the DS1 board. This integrated channel service unit (ICSU) can accept software-administrable option downlinks (that is, it can respond to test codes from technician's equipment and report its status). When you enter **integrated**, fields for administering options on the ICSU appear on page 2 of the **DS1 Circuit Pack** screen.

Slip Detection

Slips — synchronization errors — slow digital transmissions and can result in data loss. The server maintains a slip-count record for each DS1 interface to detect errors and evaluate their severity (the type of alarm). If as many as 50 percent of those spans administered for slip detection are experiencing slips (with respect to the primary), then a decision is made to switch to the secondary.

CAUTION:

Always enter **y** for DS1 circuit packs that serve as primary or secondary synchronization references.

Valid entries	Usage
y	Enter **y** to allow maintenance software to measure the slip-rate of this circuit pack and determine whether it's excessive. Typically, enter **y** for DS1 spans used for data applications and for spans used as synchronization references. This excludes all T1-spans connecting channel banks, unless the channel bank is externally timed. This entry enables switching between the primary and secondary synchronization references and an internal high-accuracy clock.
n	Enter **n** for DMI-BOS links or when testing is not required. Normally, enter **n** for DS1 spans that are used exclusively for voice and that do not serve as the primary or secondary synchronization source.

Block Progress Indicator

This field allows you to block sending of the progress indicator in the SETUP message. It appears when the **Country Protocol** field is set to 1 and the **Protocol Version** field is set to `b`.

Valid entries	Usage
y	Enter **y** to prevent the progress indicator from being sent in the SETUP message.
n	Enter **n** to allow the progress indicator to be sent.

Field descriptions for page 2

Figure 326: DS1 Circuit Pack screen

```
add ds1 nnnn                                                   Page 2 of x
                               DS1 CIRCUIT PACK

ESF DATA LINK OPTIONS

                      Network Management Protocol:
  Send ANSI-T1.403 One-Second Performance Reports?
                              Far-end CSU Address:


INTEGRATED CSU OPTIONS
                                     Transmit LBO:
                                     Receive ALBO:
                                     Upon DTE LOS:

CPE LOOPBACK JACK OPTIONS
                   Supply CPE Loopback Jack Power?
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Page 2 does not appear for all DS1 circuit packs. For those circuit packs that support it, this page appears only when the **Framing Mode** field is **esf** or the **Near-end CSU Type** field is **integrated**.

ESF DATA LINK OPTIONS

Far-end CSU Address

This field, which, appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
a **b**	Enter **b**. This field administers the transmit direction address used for the **`ESF data link`** command with both integrated and external channel service units (CSU).

Network Management Protocol

This field appears only if the **Framing Mode** field is **esf**.

Valid entries	Usage
tabs	The entry in this field, used only with circuit packs that have an integrated channel service unit (CSU), allows the data link to be remotely monitored.

Send ANSI-T1.403 One-Second Performance Reports

This field selects whether your DS1 circuit pack will send error reports to the far-end server/ switch. These reports are useful for network management, and are sent at 1-second intervals when enabled. This field appears only if the **Framing Mode** field is **esf**. It is used only with circuit packs that have an integrated channel service unit (CSU).

Valid entries	Usage
y/n	Enter **n**. Consult your Avaya technical support representative if you think you might want to use these reports.

INTEGRATED CSU OPTIONS

Receive ALBO (Receive Automatic Line Build-Out)

This field increases the strength of incoming signals by a fixed amount to compensate for line losses.

Valid entries	Usage
26db	To set this field correctly, measure the signal loss on this specific facility. However, you can enter **26db** for most applications. **36db** is occasionally appropriate, mainly on campus networks that don't conform to public telephone network standards.
36db	

Transmit LBO (Transmit Line Build-Out)

This field reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your Communication Manager server (measured by cable length from the smart jack) and the nearest repeater. Where another server/switch is at the end of the circuit, as in campus environments, use the cable length between the 2 switches to select the appropriate setting from the table below. This field appears if the **Near-end CSU Type** field is **integrated**.

Valid entries	Usage
0db	For distances of 2,001 to 3,000 feet
-7.5db	For distances of 1,001 to 2,000
-15db	For distances of 0 to 1,000 feet
-22.5db	For mid-span repeaters

Upon DTE LOS

DTE stands for "Data Terminal Equipment." This field tells Communication Manager what to do if the outgoing signal from the DS1 circuit pack (the data terminal equipment) to the network is lost.

Valid entries	Usage
loopback	Enter **loopback** to return the network signal to the network. This prevents any alarms from being generated at the far-end
ais	Enter **ais** (Alarm Indicator Signal) to send an unframed all-ones signal (the AIS or Blue Alarm) to the far-end server/switch. This option alerts your network service provider to the problem immediately and aids in troubleshooting.

CPE LOOPBACK JACK OPTIONS

Supply CPE Loopback Jack Power

If a CPE (Customer Premise Equipment) Loopback Jack is installed, the DS1 board should supply power to the equipment during loopback testing.

Valid entries	Usage
y/n	Enter **y** if a CPE Loopback Jack is installed. If not, you must enter **n**.

Related topics

See Setting up digital trunks on page 491 for instructions.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Duplicate Station

Use this screen to add telephones is to copy the information from an existing telephone and modify it for each new telephone. For example, you can configure one telephone as a template for an entire work group. Then, you merely duplicate the template **Station** screen to add all the other extensions in the group. Note that only telephones of the same model can be duplicated. The duplicate command copies all the feature settings from the template telephone to the new telephones.

Note:

For more information, see Duplicating telephones in Chapter 3, *Managing telephones*. For field descriptions of specific fields, see the **Station** screen.

Figure 327: Duplicate Station screen - page 1

```
duplicate station 1234567890123                                    Page 1 of 2
                                     STATION


                                                        Security           Endpt
Extension       Port    Name                            Code               ID
1234567890123   1234567 123456789012345678901234567 12345678           12
```

Figure 328: Duplicate Station screen - page 2

```
duplicate station 1234567890123                                    Page 2 of 2
                                     STATION

Extension       Room       Jack  Cable
1234567890123   1234567890 12345 12345
```

Duplicate Vector

Use this screen to define the vector numbers and names for the vectors to be duplicated from the master vector and to display one VDN extension that the vector is assign to. An asterisk (*) appears if the vector is assigned to two or more VDNs.

Upon submission of this screen, copies of the master vector are created, numbered and named as specified on the screen, with all steps populated exactly the same as the original. After the vector duplicates are created, you can use the **`change vector`** command to add to or otherwise edit the vector(s), including changing the type vector fields as required. *Goto* references, particularly, should be reviewed for appropriateness in the copies.

Figure 329: Duplicate Vector screen - page 1

```
duplicate vector 1                                                 Page 1 of x
                             DUPLICATE VECTOR

        Vector    Name               Assigned to VDN          More VDN's
Master    1      Number 9
          3
```

Vector

The first row displays the existing master vector showing the vector number and name (if assigned) for the master vector specified in the duplicate vector command line. In the next row, enter the number of an unassigned vector between 1 and 2000 (1 to 256 for S8300/S8400 platforms).

The lines following the master vector are for defining the vector numbers and names for the duplicates to be created. The screen shows 16 lines numbered 1 to 16 for specifying the vector numbers and (optionally) names for the vectors that will be copies/duplicates of the master vector. If a **`start`** ***`nnnn`*** entry is included on the command line, the specified ***`nnnn`*** number is to be used as the first vector number to be used for creating the duplicates. If a start number is entered on the command line without including a count entry, then only one vector number will be pre-entered as the vector number for the duplicate. If the start vector number specified is populated (has one or more steps administered), the first unused vector after the specified start vector number is pre-entered.

If a **`count`** ***`xx`*** entry is included in the command line, that count (***`xx`***) is to be used to define how many vector numbers (up to 16) are to be pre-entered on the **Duplicate Vector** screen to be used when creating the duplicates of the master vector. The pre-entered vector numbers are numbered sequentially beginning with the first unused vector found either starting with vector number 1 (if a **`start`** ***`nnnn`*** entry is not included) or starting at the specified start number (*nnnn*). If any of the vectors in that sequence are already defined with one or more steps assigned, then those numbers (defined vectors) are to be skipped when listing the numbers for the vector duplicates. If the vectors chosen for the pre-entered listing have a name assigned (without any steps populated), the vector names are shown on the **Duplicate Vector** screen along with the pre-assigned vector numbers. You can change the listing of one or more pre-entered vector numbers to replace the vector numbers chosen by the system, or to add additional vector numbers for duplicates. You can use any unassigned vector number in this field.

Name

Enter a name for the new vector. Entry of a vector name is optional so that duplicates can be created without a vector name entered. Any pre-assigned vector names can also be replaced with a different name which is to be used when creating the duplicates.

Assigned VDN

This field displays an extension number (up to 13 digits) of the first VDN (in numerical extension order) to which the vector is assigned to, if any. The **Assigned VDN** and **More VDN's** columns are populated for the master vector and any of the duplicate vectors which may already be assigned to one or more VDNs. Pre-entered vector numbers have these columns populated when the screen first appears. User-entered vector numbers appear in these columns when tabbing to the next vector number field.

More VDN's

This field displays an asterisk (*) if there is at least one more VDN with that vector assigned.

Enable File Transfer

Use this screen to enable SFTP on TN799BP Control Lan (C-LAN) and VAL circuit packs.

Field descriptions for page 1

Figure 330: Enable File Transfer screen

```
enable filexfer a03                                                  Page 1
                             ENABLE FILE TRANSFER
             Login:______
          Password:___________
          Password:___________
            Secure?
```

Login

Valid entries	Usage
3 to 6 alphanumeric characters	Enter your login.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Enter your password.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Repeat your password for verification. Entry must be identical in both **Password** fields.

Secure

Valid entries	Usage
y/n	Enter **y** to enable SFTP instead of FTP or TFTP. If the circuit pack does not support a secure session, no session is enabled. Default is **y**.

Enable Session

Use this screen to enable SSH instead of Telnet.

Field descriptions for page 1

Figure 331: Enable Session screen

```
enable session                                                      Page 1
                             ENABLE SESSION
             Login:______
          Password:___________
          Password:___________
            Secure?
     Time to Login:
```

Login

Valid entries	Usage
3 to 6 alphanumeric characters	Enter your login.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Enter your password.

Password

Valid entries	Usage
7 to 11 characters, with at least one number	Repeat your password for verification. Entry must be identical in both **Password** fields.

Secure

Valid entries	Usage
y/n	Enter **y** to indicate that SSH is used instead of Telnet. Default is **y**.

Time to Login

This field appears only if the board in question is a TN2302.

Valid entries	Usage
0 to **255**	Enter the number of minutes allowed for login before the session times out. Default is blank.

Extended Pickup Group

This screen allows grouping of pickup groups into extended pickup groups. This allows users to answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Field descriptions for page 1

Figure 332: Extended Pickup Group screen

```
change extended-pickup-group n                              Page   1 of  x
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                          Pickup   Pickup Group
Number     Number                              Number     Number

 0:      _____                                 13:      _____
 1:      _____                                 14:      _____
 2:      _____                                 15:      _____
 3:      _____                                 16:      _____
 4:      _____                                 17:      _____
 5:      _____                                 18:      _____
 6:      _____                                 19:      _____
 7:      _____                                 20:      _____
 8:      _____                                 21:      _____
 9:      _____                                 22:      _____
10:      _____                                 23:      _____
11:      _____                                 24:      _____
12:      _____                                 25:      _____
```

Extended Group Number

This display-only field shows the number associated with a collection of pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Pickup Group Number

This field determines which call pickup groups can answer calls in the extended pickup group.

Valid entries	Usage
1 to **800** or blank (DEFINITY CSI) **1** to **5000** or blank (S8300 and S87XX Servers)	Enter the pickup group numbers for each of the pickup groups that you want to belong to this extended group.

Pickup Number

This display-only field shows the pickup number assigned to the pickup group number. This is the number users dial after the feature access code (FAC) to pick up calls in their extended pickup group.

Extensions Administered to have an MCT-Control Button

This screen lists the extensions that can take control of a Malicious Call Trace (MCT) request. In order to give a user the ability to take control of such requests, you need to add their extension to this list and assign them a **mct-control** feature button.

Field descriptions for page 1

Figure 333: Extensions Administered to have an MCT-Control Button screen

```
 display mct-group-extensions                                      Page 1 of 2

Extensions Administered to have an MCT-Control Button:
    1234567890123       1234567890123       1234567890123       1234567890123
 1: 41917           19:                 37:                 55:
 2:                 20:                 38:                 56:
 3: 41963           21:                 39:                 57:
 4:                 22:                 40:                 58:
 5: 41801           23:                 41:                 59:
 6: 41973           24:                 42:                 60:
 7: 43911           25:                 43:                 61:
 8:                 26:                 44:                 62:
 9: 41908           27:                 45:                 63:
10:                 28:                 46:                 64:
11:                 29:                 47:                 65:
12:                 30:                 48:                 66:
13:                 31:                 49:                 67:
14:                 32:                 50:                 68:
15:                 33:                 51:                 69:
16:                 34:                 52:                 70:
17:                 35:                 53:                 71:
18:                 36:                 54:                 72:
```

1 to 100

Enter the extension for a telephone or attendant console that you want to have an **MCT-Control** button. Note that you must also assign the **mct-control** button on the extension's **Station** or **Attendant Console** screen.

Note:

Page 2 contains elements 73 to 100.

Extensions to Call Which Activate Features by Name

With this screen, you can assign a dialed extension to a feature within Communication Manager. This extension is called a feature name extension (FNE). You must set up the FNE mapping. All extensions must fit your dial plan and because they are implemented system-wide. These extensions are paired with feature access codes (FACs). When a user calls the extension, the feature access code activates the feature. Administer the FACs on the **Feature Access Code (FAC)** screen. For more information about individual features, see *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

To administer this screen, type:

`change off-pbx-telephone feature-name-extensions [set <1-99>]` or blank.

The set number you designate in the command displays. If you do not enter a set number with the command, set 1 automatically displays.

Note:

The set number is not the same as the location number on the Communication Manager **Locations** screen.

Figure 334: Extensions to Call which Activate Features By Name screen - page 1

```
change off-pbx-telephone feature-name-extensions set x          Page 1 of 2

         EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

     Active Appearance Select: 8866123456789
          Automatic Call Back:
   Automatic Call-Back Cancel:
             Call Forward All:
  Call Forward Busy/No Answer:
          Call Forward Cancel:
                    Call Park: 8861123456789
        Call Park Answer Back: 8862123456789
                 Call Pick-Up: 8863123456789
         Conference on Answer: 8865123456789
         Calling Number Block:
       Calling Number Unblock:
        Directed Call Pick-Up: 8864123456789
        Drop Last Added Party:
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup:
       Held Appearance Select:
```

Figure 335: Extensions to Call which Activate Features By Name screen - page 2

```
change off-pbx-telephone feature-name-extensions                  Page 2 of 2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

      Idle Appearance Select:
          Last Number Dialed:
        Malicious Call Trace:
 Malicious Call Trace Cancel:
         Off-Pbx Call Enable:
        Off-Pbx Call Disable:
               Priority Call:
              Send All Calls:
       Send All Calls Cancel:
         Transfer On Hang-Up:
      Transfer to Voice Mail:
     Whisper Page Activation:
```

Field descriptions

Extension

Each **Extension** field is an extension that matches your dial plan. A user dials the extension from their Extension to Cellular telephone to activate an FAC administered for that feature.

Valid entries	Usage
0–**9** or blank	Type an extension number, up to seven digits, for the Communication Manager feature you want users to access from their Extension to Cellular telephones. Default is blank.

Note:

The **Transfer to Voice Mail** FNE is used when a user is active on a call and wants to transfer the other party to voice mail, or to the principal's voice mail, if this is a covered call. This FNE can also be used when a user goes off-hook for the first time and dials the **Transfer to Voice Mail** FNE to be connected to the voice mail administered in his coverage path. This is identical to dialing a Transfer to Voice Mail feature access code (FAC).

Feature Access Code (FAC)

This screen assigns feature access codes (FACs) that, when dialed, activate or cancel the system features. Each field on this screen has the same valid values, which must conform to feature access codes or dial access codes as defined by your dial plan.

Field descriptions for page 1

Figure 336: Feature Access Code (FAC) screen

```
change feature-access-codes                                       Page 1 of x
                                   FEATURE ACCESS CODE (FAC)
             Abbreviated Dialing List1 Access Code: ____
             Abbreviated Dialing List2 Access Code: ____
             Abbreviated Dialing List3 Access Code: ____
    Abbreviated Dial - Prgm Group List Access Code: ____
                          Announcement Access Code: ____
                           Answer Back Access Code: ____
                             Attendant Access Code: ____
          Auto Alternate Routing (AAR) Access Code: ____
            Auto Route Selection ARS) Access Code1: ____  Access Code 2: ____
                     Automatic Callback Activation: ____   Deactivation: ____
    Call Forwarding Activation Busy/DA: ____   All: ____   Deactivation: ____
       Call Forwarding Enhanced Status: ____   Act: ____   Deactivation: ____
____                         Call Park Access Code: ____
                           Call Pickup Access Code: ____
    CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                      CDR Account Code Access Code: ____
                            Change COR Access Code: ____
                       Change Coverage Access Code: ____
                         Contact Closure Open Code: ____     Close Code: ____
                        Contact Closure Pulse Code: ____
```

Abbreviated Dialing List1 Access Code

Used to access AD list 1.

Abbreviated Dialing List2 Access Code

Used to access AD list 2.

Abbreviated Dialing List3 Access Code

Used to access AD list 3.

Abbreviated Dial - Prgm Group List Access Code

Used to enter a group list from a telephone. The user's extension must be entered on the **Abbreviated Dial Group List** screen in order to program the group list.

Announcement Access Code

Used to record announcements.

Answer Back Access Code

Used to retrieve parked calls.

Attendant Access Code

This field only appears and is valid if an **attd** entry does not exist on the **Dial Plan Analysis** screen. You cannot have an entry in both the **Dial Plan Analysis** screen and the **Feature Access Code (FAC)** screen. While the **Dial Plan Analysis** screen allows administration of only one attd code that connects to one attendant, this field on the **Feature Access Code (FAC)** screen allows you to administer more than one attendant access code in a single distributed network. Attendant access numbers can start with any number from 0 to 9 and contain 1 or 2 digits.

Auto Alternate Routing (AAR) Access Code

Used to access AAR.

Auto Route Selection (ARS) Access Code1

Used to access ARS. You can have one ARS access code for local and one for long distance, and route accordingly.

(ARS) Access Code 2

Also used to access ARS.

Automatic Callback Activation/Deactivation

Used to activate/cancel Automatic Callback.

Call Forwarding Activation Busy/DA

Used to forward calls to an administered number if the user is busy or does not answer.

Call Forwarding Enhanced Activation/Deactivation

Enter feature access code numbers to allow users to activate and deactivate Enhanced Call Forwarding. The FACs for activation and deactivation must be administered together. One can't exist without the other. In contrast, the FAC for **Call Forwarding Enhanced Status** can exist by itself and without the others.

Call Forwarding Enhanced Status

Used to display the status of Enhanced Call Forwarding.

Call Park Access Code

Used to park an active call, which can then be retrieved from a different station using the answer back access code. Do not administer to have the same first digit as another feature access code that is longer in length.

Call Pickup Access Code

Used to answer a call directed to a pickup group.

CAS Remote Hold/Answer Hold-Unhold Access Code

Used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote server running Communication Manager. This FAC can also be used by an analog station. Flashing the switch-hook for the proper interval (between 200 and 1000 ms) while talking on an existing call causes the existing call to be placed on soft hold, allowing the analog user to dial the Answer Hold-Unhold FAC to Hard hold the call.

CDR Account Code Access Code

Used prior to entering an account code for CDR purposes.

Change COR Access Code

Used to allow users to change their class of restriction (COR) from a telephone. This field can only be used if the **Change COR by FAC** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Change Coverage Access Code

Used to change a coverage path from a telephone or remote station.

Contact Closure Close Code

Used to close a contact closure relay. Must be administered if the **Contact Closure Open Code** field is administered.

Contact Closure Open Code

Used to open a contact closure relay. Must be administered if the **Contact Closure Close Code** field is administered.

Field descriptions for page 2

Figure 337: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page   2 of   x
                              FEATURE ACCESS CODE (FAC)
                    Contact Closure Pulse Code:
                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code:
     Emergency Access to Attendant Access Code: *11
         EC500 Self Administration Access Code:
                     Enhanced EC500 Activation: *81     Deactivation:#81
           Enterprise Mobility User Activation: *55     Deactivation:#56
  Extended Call Fwd Activate Busy D/A: *23 All: *24     Deactivation:#23
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code: *88
             Group Control Restrict Activation: *15     Deactivation: #15
                    Hunt Group Busy Activation: *82     Deactivation: #83
                              ISDN Access Code:
                Last Number Dialed Access Code: *54
     Leave Word Calling Message Retrieval Lock: *48
   Leave Word Calling Message Retrieval Unlock: #45
```

Contact Closure Pulse Code

Used to pulse a contact closure relay.

Data Origination Access Code

Used to originate a data call from a voice station.

Data Privacy Access Code

Used to isolate a data call from call waiting or other interruptions.

Directed Call Pickup Access Code

Used to establish directed call pickup.

EC500 Self Administration Access Code

The Self Administration Feature (SAFE) Access Code allows users to self-administer their cell phone number for the Extension to Cellular feature. Users can add or change their cell phone number through this feature access code. An administrator can still enter or change cell phone numbers. The user calls the SAFE access code and enters their cell phone number. The administration sequence differs based on what phone is used to access SAFE. For more information, see *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205 and *Avaya Extension to Cellular User's Guide*, 210-100-700.

Emergency Access To Attendant Access Code

Used to gain access to the attendant in an emergency situation. Such calls alert as emergency calls. This field cannot be used if the **Emergency Access to Attendant** field is not enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Enhanced EC500 Activation

Type a feature access code number to allow users to activate Extension to Cellular remotely.

Enhanced EC500 Deactivation

Type a feature access code number to allow users to deactivate Extension to Cellular remotely.

Enterprise Mobility User Activation

Type a feature access code number to activate the Enterprise Mobility User feature for a particular user, associating the features and permissions of their primary telephone to a telephone of the same type anywhere within the customer's enterprise. For more information about Enterprise Mobility User, see Setting Up Enterprise Mobility User on page 171.

Enterprise Mobility User Deactivation

Type a feature access code number to deactivate the Enterprise Mobility User feature. For more information about Enterprise Mobility User, see Setting Up Enterprise Mobility User on page 171.

Extended Call Fwd Activate Busy D/A

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Activate All

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Deactivation

Used to deactivate call forwarding from a telephone or remote location.

Note:

An extension must have Station Security Codes administered to use the following FACs:

- Extended Call Forward All Activate
- Extended Call Forward Busy/Don't Answer Activate
- Extended Call Forward Deactivate
- Change Coverage

Extended Group Call Pickup Access Code

The feature access code (FAC) users enter when a call directed to another pickup group is to be answered. Users must enter a valid "Pickup Number" following the Extended Group Call Pickup Access Code to complete the operation.

Facility Test Calls Access Code

Used to place activate a facility test call.

SECURITY ALERT:

To ensure the security of your system, leave Facility Test Calls Access Code blank except when actually testing trunks.

Flash Access Code

Used to generate trunk flash. This code ensures that the flash signal is interpreted by the central office switch, rather than by Avaya Communication Manager.

Group Control Restrict Activation / Deactivation

Used to change the restriction level for all users with a given class of restriction. Requires console permissions.

Hunt Group Busy Activation/Deactivation

Hunt Group members can dial these codes to place themselves in a busy state, and to become available again.

ISDN Access Code

Used to place an ISDN call without using ARS, AAR, or UDP.

Last Number Dialed Access Code

Used to redial the last number dialed from this station.

Leave Word Calling Message Retrieval Lock

Used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. The **Lock Messages** field on the **Station** screen also must be enabled.

Leave Word Calling Message Retrieval Unlock

Used to unlock a telephone's display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

Field descriptions for page 3

Figure 338: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 3 of x
                             FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message: *60
           Leave Word Calling Cancel A Message: #60
   Limit Number of Concurrent Calls Activation:         Deactivation:
               Malicious Call Trace Activation:         Deactivation:
         Meet-me Conference Access Code Change:

PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:
                              Posted Messages:

                  Priority Calling Access Code:
                           Program Access Code:

       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:       Deactivation:
              Self Station Display Access Code:
                     Send All Calls Activation:       Deactivation:
         Station Firmware Download Access Code:
```

Leave Word Calling Send A Message

Used to send a leave word calling message.

Leave Word Calling Cancel A Message

Used to cancel a leave word calling message.

Limit Number of Concurrent Calls Activation/Deactivation

Used to limit concurrent calls on a station even when additional call appearances normally would be available.

Malicious Call Trace Activation

Used to activate a trace request on a malicious call.

Meet-me Conference Access Code Change

Allows the controlling user of a Meet-me Conference VDN to change the access code.

PASTE (Display PBX data on telephone) Access Code

Allows users to view call center data on display telephones. PASTE is used in conjunction with Avaya IP Agent.

Personal Station Access (PSA) Associate Code

Used to associate a telephone with the telephone features assigned to a user's extension.

Personal Station Access (PSA) Dissociate Code

Used to remove the association between a physical telephone and an extension number. You cannot provide the code until Personal Station Access (PSA) on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Per Call CPN Blocking Code Access Code

If CPN blocking is off for a trunk group, users can turn it on for a call by using this code. When they dial this code, the calling party number is not sent to the public network.

Per Call CPN Unblocking Code Access Code

If CPN blocking is on for a trunk group, users can turn it off for a call by using this code. When they dial this code, the calling party number is sent to the public network.

Posted Messages

Only appears if the **Posted Messages** field is set to **y** on the **System Parameters Customer-Options (Optional Features)** screen. Used to access the Posted Messages feature. See "Posted Messages" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information

Priority Calling Access Code

Used to enable priority calling, a special type of call alerting between internal telephone users, including the attendant. The called party hears a distinctive ringing when the calling party uses Priority Calling.

Program Access Code

Used to program abbreviated dial buttons on an individual telephone.

Refresh Terminal Parameters Access Code

Used to update terminal parameters on an individual telephone when system settings have changed.

Remote Send All Calls Activation/Deactivation

Used to activate or deactivate the Send All Calls feature. Requires console permissions.

Self Station Display Activation

The **Self Station** field is not active. If set to a valid FAC, a digital station displays its primary extension number when the FAC is entered.

Send All Calls Activation/Deactivation

Used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

Station Firmware Download Access Code

This field specifies the feature access code used for 2420/2410 DCP station firmware downloads.

Field descriptions for page 4

Figure 339: Feature Access Code (FAC) screen

```
change feature-access-codes                                    Page 4 of x
                             FEATURE ACCESS CODE (FAC)
                       Station Lock Activation:       Deactivation:
      Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                Remove:
  Station User Button Ring Control Access Code:
                          Telephone Activation: #*
             Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:       Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
           Whisper Page Activation Access Code:
```

Station Lock Activation/Deactivation

Used to activate or deactivate Station Lock.

Station Security Code Change Access Code

Enter the code the user must dial to change their Station Security Code. The SCC must be administered before the user can change it using this FAC. That is, a user cannot change a blank SCC.

Station User Admin of FBI Assign

Used to activate or deactivate Facility Busy Indicators.

Station User Button Ring Control Access Code

Used to control the ring behavior for each line appearance and bridged appearance from the station. Allows users to have their telephones ring either silently or audibly.

Terminal Dial-Up Test Access Code

Used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the server running Avaya Communication Manager. The Terminal Dial-Up test ensures that the terminal and each of its buttons can communicate with the server. This test is initiated by a user entering this feature access code. This test is mostly for use by terminal service personnel, but may be used by any station user.

In order to use this feature, simply lift the receiver on a supported terminal and dial the feature access code. The terminal's button lights are extinguished, the display clears, and the message waiting lamp lights. This lamp remains lit throughout the test. When a button is depressed during the test, the server responds with the appropriate tone, light, or display. Depressing another button clears the lamp, the tone and ringer associated with the previous button and lights the lamp, sends new tone and ringer associated with the new button. If the same button is depressed, the lamp, tone and ringer are turned OFF, which makes the button work as an ON/OFF button. The terminal remains in this test mode until you hang up the receiver. To use a Terminal Dial-up Test FAC on a telephone with bridged appearances, add a bridged-appearance of the principal telephone.

Terminal Translation Initialization Merge Code

Enter the digits that must be dialed to install (merge) a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must have been used, or an **X** administered in the **Port** field of the **Station** screen, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective. (If you try to use this and it doesn't work, check the **Station** screen for this extension. If there is still a port assigned, type **X** in the **Port** field, then try the TTI merge again.)

Terminal Translation Initialization Separation Code

Enter the digits that must be dialed to remove (separate) a station from a location without losing any of its feature settings.

Transfer to Voice Mail Access Code

Enter the digits that must be dialed to allow coverage to transfer the caller to the original call recipient's AUDIX mail where the caller can leave a message. Do not administer this code to have the same first digit as another feature access code that is longer in length.

Trunk Answer Any Station Access Code

Enter the access code that station users must dial to answer calls alerting on night bells.

User Control Restrict Activation/Deactivation

Used to change the restriction level for a specific extension. Requires console permissions.

Voice Coverage Message Retrieval Access Code

Allows users to retrieve voice messages for another user (for whom they are a coverage point) via a digital display module.

Voice Principal Message Retrieval Access Code

Allows users to retrieve their own voice messages for another user via a digital display module.

Whisper Page Activation Access Code

Allows users to place a page to another user's telephone, when active on a call. The paged user, and not the other parties on the call, hears the page.

Field descriptions for page 5

The feature access codes on this page pertain only to ACD call centers.

Figure 340: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page 5 of x
                        FEATURE ACCESS CODE (FAC)

                  Automatic Call Distribution Features

                   After Call Work Access Code: ___
                            Assist Access Code: ___
                           Auto-In Access Code: ___
                          Aux Work Access Code: ___
                             Login Access Code: ___
                            Logout Access Code: ___
                         Manual-In Access Code: ___
     Service Observing Listen Only Access Code: ___
     Service Observing Listen/Talk Access Code: ___
         Service Observing No Talk Access Code: ___
                   Add Agent Skill Access Code: ___
                Remove Agent Skill Access Code: ___
            Remote Logout of Agent Access Code: ___
```

Add Agent Skill Access Code

Enter the digits an agent must dial to be able to add a skill to their current skill set.

After Call Work Access Code

Enter the code the agent must dial when the agent will be performing work-related ACD activities.

Assist Access Code

Enter the digit the agent must dial to request assistance from the split supervisor.

Auto-In Access Code

Enter the code the agent must dial to become automatically available to receive another ACD call each time a call is released.

Aux Work Access Code

Enter the code the agent must dial when the agent will be performing non-ACD activities.

Login Access Code

Enter the code the agent must dial to gain access to the ACD functions. This is a system-wide code for all ACD agents.

Logout Access Code

Enter the logout code the agent must enter to exit ACD. This is a system-wide logout code for all ACD agents.

Manual-In Access Code

Enter the code the agent must dial to receive a single, new ACD call upon the completion of an ACD call.

Note:

The following two fields appear only if **Service Observing (Remote/By FAC)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Remove Agent Skill Access Code

This field appears only if **Service Observing (Remote/By FAC)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**. Enter the digits an agent must dial to be able to remove a skill from their current skill set.

Remote Logout of Agent Access Code

This field appears only if **Service Observing (Remote/By FAC)**, **Vectoring (Basic)**, and **Vectoring (Prompting)** on the **System Parameters Customer-Options (Optional Features)** screen are set to **y.** Enter the digits you need to dial to remotely logout an idle ACD or EAS agent.

Service Observing Listen Only Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen to other agent ACD calls without being heard on the ACD call.

Service Observing Listen/Talk Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to both listen and be heard on an ACD call.

Service Observing No Talk Access Code

The following field appears only if **Expert Agent Selection (EAS) Enabled** is optioned on the **Feature-Related System-Parameters** screen. Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen only without reserving a 2nd timeslot for potential toggle to talk and listen mode. When this FAC is used for activation, the observing connection is listen only. Any attempt to toggle to talk via the Service Observing (SO) feature button is denied.

Field descriptions for page 6

The feature access codes on this page pertain only to Call Vectoring/ Prompting features.

Figure 341: Feature Access Code (FAC) screen

```
change feature-access-codes                                   Page 6 of x
                             FEATURE ACCESS CODE (FAC)

                        Call Vectoring/Prompting Features

     Converse Data Return Code:

  Vector Variable 1 (VV1) Code:
  Vector Variable 2 (VV2) Code:
  Vector Variable 3 (VV3) Code:
  Vector Variable 4 (VV4) Code:
  Vector Variable 5 (VV5) Code:
  Vector Variable 6 (VV6) Code:
  Vector Variable 7 (VV7) Code:
  Vector Variable 8 (VV8) Code:
  Vector Variable 9 (VV9) Code:
```

Converse Data Return Code

Enter a 1 to 4 digit number (# can be used as the first digit). If there is data to be returned to the switch, the Converse Data Return Code is outpulsed before the data to be passed is outpulsed.

Vector Variable x (VVx) Code (1-9)

Enter a 1 to 4 digit number (# can be used as the first digit). Entry of the Vector Variable Code FAC allows you to change the variable on the **Variables for Vectors** screen.

Field descriptions for page 7

The feature access codes on this page pertain only to Hospitality features.

Figure 342: Feature Access Code (FAC) screen

```
change feature-access-codes                                     Page 7 of x
                             FEATURE ACCESS CODE (FAC)

                               Hospitality Features

                        Automatic Wakeup Call Access Code: *11___
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                       Verify Wakeup Announcement Access Code: ______
                             Voice Do Not Disturb Access Code: ______
```

Automatic Wakeup Call Access Code

Enter the access code the user must dial to schedule or cancel a wakeup call.

Housekeeping Status (Client Room) Access Code

Enter the access code the housekeeper dials from the client's room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing. You can assign a definition to the six codes on the **Hospitality** screen.

Housekeeping Status (Station) Access Code

Enter the access code the housekeeper must dial to provide room status. This access code must be dialed from designated telephones. There are four codes.

Verify Wakeup Announcement Access Code

Enter the access code the user can dial to verify a wakeup announcement.

Voice Do Not Disturb Access Code

Enter the access code the user must dial to enter or cancel a do not disturb request without using a display - through the use of voice prompting.

Field descriptions for Multimedia Features page

The feature access codes on this page pertain only to Multimedia Call Handling (MMCH).

Figure 343: Feature Access Code (FAC) screen

```
change feature-access-codes                                         Page 8 of x
                              FEATURE ACCESS CODE (FAC)
                                 Multimedia Features

                       Basic Mode Activation:
                    Enhanced Mode Activation:
                 Multimedia Call Access Code:
       Multimedia Data Conference Activation:  Deactivation:
        Multimedia Multi-Address Access Code:
            Multimedia Parameter Access Code:
```

Basic Mode Activation

If you enter this FAC when your system is an Enhanced multimedia complex, it will revert to a Basic multimedia complex. If you enter this FAC when your system is a Basic mode station it will do nothing.

Enhanced Mode Activation

If you enter this FAC when your system is a Basic multimedia complex, it will become an Enhanced multimedia complex. If you enter this FAC when your system is an Enhanced mode station it will do nothing.

Multimedia Call Access Code

If you enter this FAC from any voice station, it indicates to Avaya Communication Manager that you are making an Enhanced mode multimedia call. If you originate a multimedia call with the multimedia call access code, it will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System Parameters** screen.

Multimedia Data Conference Activation

If you enter this FAC from any voice station that is participating in a multimedia call, it will alert Avaya Communication Manager that you want to enable data collaboration with the other parties on the call. If you enter this FAC a second time, it will give denial treatment (since a collaboration session is already active). This FAC only applies to voice stations on servers equipped with ESM adjuncts.

Multimedia Data Conference Deactivation

If you enter this FAC from the telephone that enabled data collaboration on a multimedia mode call, it will deactivate the data session and revert to a voice and video call. If a user enters this FAC while participating in a data-collaboration multimedia call that the user did not initiate, the system responds with denial treatment.

Multimedia Multi-Address Access Code

The multimedia multi-address access code is similar to the multimedia call access code. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. For example, ISDN-BRI provided by a Central Office is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code will originate a call according to the Default Multimedia Parameters selected on the **Feature-Related System-Parameters** screen.

Multimedia Parameter Access Code

This FAC can be entered by any voice station to indicate to Avaya Communication Manager that you want to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

Field descriptions for MLPP features page

The feature access codes on this page pertain only to Multiple Precedence and Preemption (MLPP) calls.

Figure 344: Feature Access Code (FAC) screen

```
change feature-access-codes                                 Page 8 of x
                              FEATURE ACCESS CODE (FAC)
                                 MLPP Features

             Precedence Calling Access Code: 8


    WNDP PRECEDENCE ACCESS CODES:
                 Flash Override Access Code: 90
                          Flash Access Code: 91
                      Immediate Access Code: 92
                       Priority Access Code: 93
                        Routine Access Code: 94
```

Precedence Calling Access Code

Enter a feature access code that conforms to your dial plan, to be used to access the Multiple Level Precedence and Preemption feature.

WNDP PRECEDENCE ACCESS CODES

The system uses a feature access code to determine the precedence level for a call when the Worldwide Numbering and Dial Plan (WNDP) feature is active. Different feature access codes will be assigned for each PRECEDENCE level. When the WNDP feature is not active, the user will dial the PRECEDENCE CALLING feature access code followed by a digit indicating the precedence level of the call.

Flash Override Access Code

Enter a FAC to correspond with the Flash Override preemption level.

Flash Access Code

Enter a FAC to correspond with the Flash preemption level.

Immediate Access Code

.Enter a FAC to correspond with the Immediate preemption level.

Priority Access Code

.Enter a FAC to correspond with the Priority preemption level.

Routine Access Code

.Enter a FAC to correspond with the Routine preemption level.

Feature-Related System Parameters

This screen implements system parameters associated with various system features.

Note:

This screen used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a separate screen, which you can access with the command **`change system-parameters coverage-forwarding`**.

Field descriptions for page 1

Figure 345: Feature-Related System Parameters screen

```
change system-parameters features                                 page 1 of x

                         FEATURE-RELATED SYSTEM PARAMETERS
                            Self Station Display Enabled? n
                                 Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                   Call Park Timeout Interval (minutes): 10
     Off-Premises Tone Detect Timeout Interval (seconds): 20_
                               AAR/ARS Dial Tone Required? y
                      Music/Tone On Hold: music    Type: _______
           Music (or Silence) On Transferred Trunk Calls: all
                    DID/Tie/ISDN/SIP Intercept Treatment: attd

 Internal Auto-Answer of Attd-Extended/Transferred Calls? y
               Automatic Circuit Assurance (ACA) Enabled? y
                                     ACA Referral Calls: local
                               ACA Referral Destination: ________
            ACA Short Holding Time Originating Extension: ______
             ACA Long Holding Time Originating Extension: ______
          Abbreviated Dial Programming by Assigned Lists:
     Auto Abbreviated/Delayed Transition Interval(rings):
                Protocol for Caller ID Analog Terminals: Bellcore
 Display Calling Number for Room to Room Caller ID Calls?
```

AAR/ARS Dial Tone Required

A second dial tone provides feedback to the user that additional dialing can occur.

Valid entries	Usage
y/n	Enter **y** to indicate a second dial tone is to be given to the calling party on a incoming tie or DID trunk call that is to be routed via AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid entries	Usage
y	Enter **y** to allow programming by station's assigned list.
n	Enter **n** if using Program Access code to indicate which personal list is to be programmed.

ACA Referral Calls

Indicates where ACA referral calls generate. This field only appears when the **Automatic Circuit Assurance (ACA) Enabled** field is **y**.)

Valid entries	Usage
local	Local referral calls generate on and for the local switch.
primary	Primary referral calls generate on the local switch for remote servers/switches as well as the local switch.
remote	Remote referral calls generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number as defined on the **Dial Plan** screen. Also, **ACA** button status transmits to other servers/switches when in a DCS network.

ACA Referral Destination

The specified extension should be equipped with a display module. This field only appears if ACA Referral Calls is **local** or **primary**.

Valid entries	Usage
An extension	Enter the extension on a local server running Communication Manager that is to receive the ACA referral call.
attd	Enter **attd** for attendant.

ACA Remote PBX Identification

This field only appears if **ACA Referral Calls** is **remote**.

Valid entries	Usage
1 to **63**	Enter a number to identify the switch in a DCS network that makes the referral call. Do not define the remote server/switch identified in this field as **local** on the system's **Dial Plan** screen.

ACA Short Holding Time Originating Extension and ACA Long Holding Time Originating Extension

Valid entries	Usage
An unassigned extension	Do not use the same extension number for both fields. The specified extensions are assigned automatically by the system when the screen is submitted. These fields only display if **ACA Referral Calls** is **local** or **primary**.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid entries	Usage
1 to **16**	Enter the number of rings before an automatic abbreviated/ delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid entries	Usage
2 to **9**	Enter the number of times the callback call rings at the calling station before the callback call is canceled.

Automatic Circuit Assurance (ACA) Enabled

If **Automatic Circuit Assurance (ACA) Enabled** is **n**, associated ACA fields will not display.

Must have an **aca-halt** button administered on the user's station. If you enable this feature, complete the following ACA-related fields.

Valid entries	Usage
y	Enter **y** if ACA measurements will be taken.
n	Otherwise, enter **n**.

Call Park Timeout Interval (minutes)

Valid entries	Usage
1 to **90**	Enter the number of minutes a call remains parked before it cancels.

DID/Tie/ISDN/SIP Intercept Treatment

Valid entries	Usage
A recorded announcement extension	Toll charges do not apply to DID and private network calls routed to an announcement. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	For system security, Avaya recommends entering **attd** in this field. This routes intercept calls to the attendant and, if the attendant receives several of these, they will know a problem exists.

Display Calling Number for Room to Room Caller ID Calls

Valid entries	Usage
y/n	Enter **y** to display the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

This only applies to digital telephones (except BRI) with a headset or speakerphone capability.

Valid entries	Usage
attd-extended	Enter **attd-extended** to enable IAA for only attendant extended calls.
both	Enter **both** to enable IAA for station transferred and attendant extended calls.
none	Enter **none** to disable IAA for all calls.
transferred	Enter **transferred** to enable IAA for only station transferred calls.

Music (or Silence) On Transferred Trunk Calls

Valid entries	Usage
all	Enter **all** to allow all transferred trunk calls to receive music until the call is answered if the Music-on-Hold feature is available.
no	Enter **no** if trunk callers are to hear music (or silence if Music-on-Hold is not administered) while waiting to be transferred, and then ringback as soon as the transfer is completed till the call is answered.
call-wait	Enter **call-wait** if trunk calls transferred to stations that require the call to wait hear music (if administered); all other transferred trunk calls receive ringback tone.

Music/Tone on Hold

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic OnHold system, which does not require such a license, from Avaya or our business partners. This field does not appear if **Tenant Partitioning** is **y** on the **System Parameters Customer-Options (Optional Features)** screen. In that case, use the **Tenant** screen to establish Music on Hold.

Valid entries	Usage
music **tone** **none**	Indicates what a caller hears while on hold. Default is **none**. When **music** is entered, the **Type** field appears to define the music type.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid entries	Usage
5 to **25**	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the setting of the **Outpulse Without Tone** field on page 6 of this screen.

Port

Appears when **Music/Tone on Hold** is **music** and **Type** is **port**. Enter the necessary characters to indicate the port number that provides Music-on-Hold access.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Protocol for Caller ID Analog Terminals

Determines the protocol/tones sent to a Caller ID telephone.

Valid entries	Usage
Bellcore	Enter **Bellcore** for Bellcore protocol with 212 modem protocol tones. Used in the U.S. and similar countries.
V23-Bell	Enter **V23-Bell** for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Self Station Display Enabled

Use this field to control the use of the **inspect** button for digital display telephones.

Self Station Display allows a user to display the primary extension associated with a digital display telephone. There are two methods: (1) enter a feature access code (FAC), and (2) use the **inspect** button. In either case, the display shows the primary extension associated with the telephone where the FAC or **normal** or **exit** button is entered. In the case of the FAC, the display continues until a display-altering event occurs (for instance, going on-hook or receiving an incoming call). In the case of the **inspect** button, the display continues until the user presses the **normal** or **exit** button or until a display-altering event occurs.

Valid entries	Usage
y	The primary extension does display when the **inspect** button is pressed.
n	The extension does not display when the **inspect** button is pressed.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. See your Avaya technical support representative for assistance.

Valid entries	Usage
all	Enter **all** to enable all trunk-to-trunk transfers. This allows telephone users to set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location. This value is required for SIP Enablement Services (SES) support.
restricted	Enter **restricted** (restricted public) to restrict all public trunks (CO, WATS, FX, CPE, DID, and DIOD).
none	Enter **none** to restrict all trunks (except CAS and DCS) from being transferred.

Type

This field appears when **Music/Tone on Hold** is set to **music**.

Note:

If the **Tenant Partitioning** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**, you cannot administer the **Music/Tone on Hold** field. If the **Tenant Partitioning** field on the **System Parameters Customer-Options (Optional Features)** screen set to **y**, you must use the **Music Sources** screen to assign music to a port.

Valid entries	Usage
ext **group** **port**	• Indicate whether the source for Music on Hold is an announcement extension, an audio group, or a port on a VAL board. • Type **ext** and the corresponding extension number of the integ-mus announcement/audio source. • Type **group** and the corresponding Music-on-Hold analog group number. • Type **port** and the corresponding location of the Music-on-Hold analog/aux-trunk source. **Note:** After a valid value is entered, a blank field appears for entry of the appropriate source identifier (extension number, audio group number, or port number).

Field descriptions for page 2

Figure 346: Feature-Related System Parameters screen

```
change system-parameters features                                    Page 2 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

LEAVE WORD CALLING PARAMETERS
                            Maximum Number of Messages Per Station: 10
               Maximum Number of External Calls Logged Per Station: 0
                     Message Waiting Indication for External Calls? n
  Stations with System-wide Retrieval Permission (enter extension)
    1:1234567890123   9:1234567890123   17:1234567890123   25:1234567890123
    2:               10:                18:                26:
    3:               11:                19:                27:
    4:               12:                20:                28:
    5:               13:                21:                29:
    6:               14:                22:                30:
    7:               15:                23:
    8:               16:                24:

                   Prohibit Bridging Onto Calls With Data Privacy? n
                        Enhanced Abbreviated Dial Length (3 or 4): 3
            Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

LEAVE WORD CALLING PARAMETERS

Maximum Number of Messages Per Station

Valid entries	Usage
0 to **125**	The maximum number of LWC Messages that can be stored by the system for a telephone at a given time.

Maximum Number of External Calls Logged Per Station

When an external call is not answered, the server running Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers

.

Valid entries	Usage
0 to **15**	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in the **Maximum Number of Messages Per Station (when MSA not in service)** field.

Message Waiting Indication for External Calls

Provides a message waiting indication when external calls are logged.

Valid entries	Usage
y	The message waiting indication for a particular station is on whenever an external call is logged.
n	The log of external calls has no impact on the message waiting indication.

Default Multimedia Outgoing Trunk Parameter Selection

Does not appear on S87XX Series IP-PNC.

Valid entries	Usage
2x56	Sets default parameter for bandwidth and bearer for all video calls.
2x64	

Enhanced Abbreviated Dial Length (3 or 4)

The administrator might not be able to use all entry slots because of system capacity constraints.

Valid entries	Usage
3	A value of **3** makes 1000 Enhanced List entries available to the administrator
4	A value of **4** makes 10,000 entries available.

Prohibit Bridging Onto Calls with Data Privacy

Valid entries	Usage
y/n	Enter **y** to protect calls from bridge-on by any party, including Service Observing, Intrusion, Verify, and Bridging.

Stations With System-wide Retrieval Permission (enter extension)

An extension must be removed from this list before the station is removed from the system. The server running Communication Manager refers to the extensions on this list as "super-retrievers."

Valid entries	Usage
An assigned extension	Enter up to 10 telephone extension numbers that can retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is not allowed.
attd	An entry of **attd** gives retrieval permission to all attendants.

Field descriptions for page 3

Figure 347: Feature-Related System Parameters screen

```
change system-parameters features                              page 3 of x

                         FEATURE-RELATED SYSTEM PARAMETERS
TTI/PSA PARAMETERS

   WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

         Terminal Translation Initialization (TTI) Enabled? y_
              TTI State: ________        TTI Security Code:
         Enhanced PSA Location/Display Information Enabled?
                          Default COR for Dissociated Sets:
                             CPN, ANI for Dissociated Sets:
           Unnamed Registrations and PSA for IP Telephones?
               Customer Telephone Activation (CTA) Enabled?
  Don't Answer Criteria for Logged off IP/PSA/TTI Stations? n

EMU PARAMETERS
          EMU Inactivity Interval for Deactivation (hours): 1

CALL PROCESSING OVERLOAD MITIGATION
Restrict Calls:
```

TTI/PSA PARAMETERS

CPN, ANI for Dissociated Sets

Appears when the **Default COR for Dissociated Sets** field is non-blank. Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets, if no system-wide calling party information has been administered for those protocols on their respective administration screens.

Valid entries	Usage
1 to 20 digits	Enter the calling party number or automatic number identification for calls made from dissociated telephones.

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y	Enter **y** if you want the Customer Telephone Activation feature for your system.
n	Enter n if you do not want the Customer Telephone Activation feature for your system.

Default COR for Dissociated Sets

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
0 to **995** or blank	Specify the Class of Restriction (COR) that the system uses for calls made from dissociated telephones.

Don’t Answer Criteria for Logged Off IP/PSA/TTI Stations

Use this field to control call process handling for logged-off IP/PSA/TTI Terminals.

Valid entries	Usage
y	If this field is set to **y(es)**, the caller hears a ringback tone on a call to a logged-off IP/PSA/TTI terminal. If a coverage path is administered, the coverage for “Don’t Answer” is used. If Enhanced Call Forwarding is administered, the "Don’t Answer" path is used.
n	If this field is set to **n(o)**, the caller hears a busy tone on a call to a logged-off IP/PSA/TTI terminal. If a coverage path is administered, the coverage for “busy” is used. If Enhanced Call Forwarding is administered, the “Busy” path is used.

Enhanced PSA Location/Display Information Enabled

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
y	Enter **y**, if you want the system to display: ● PSA login and associated station information when a station is PSA associated. ● PSA logout and the port when a station is PSA dissociated.
n	Enter n if you do not want the system to display PSA information.

Terminal Translation Initialization (TTI) Enabled

For more information on TTI, see "Terminal Translation Initialization" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205. You should contact your Avaya technical support representative before making changes to TTI settings.

Valid entries	Usage
y	Enter **y** to start ACTR, TTI, and PSA transactions (extension and telephone moves between ports). You can administer this field only if the **Terminal Trans. Init. (TTI)** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**.
n	Enter **n** to remove existing TTI port translations and make sure no new TTI port translations are generated.

TTI Security Code

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**.

Valid entries	Usage
1 to 7 digits	Enter a one-digit to seven-digit number that a TTI user must use when the user accesses TTI from a telephone or data terminal. The system displays this field only when the **Terminal Translation Initialization (TTI)** field is **y**.

TTI State

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**. Enter the type of port translation that you want for the system to use for unadministered digital ports. The default is **voice**.

Valid entries	Usage
data	Enter **data**, if you want a stand-alone data module to the be the TTI port translation for your system. The activation and deactivation sequence is entered at data terminal.
resume	Enter **resume**, if you want TTI to be available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manually suspended.
suspend	Enter **suspend** to make TTI voice or TTI data translations temporarily unavailable. The system does not remove existing TTI translations.
voice	Enter **voice**, if you want voice or voice/data terminal to be the TTI port translation for the system. The activation and deactivation sequence is entered from a telephone.

Unnamed Registrations and PSA for IP Telephones

Valid entries	Usage
y/n	Enter **y** to allow IP telephones to use the Personal Station Access (PSA) feature, and allow IP telephones to register into the state sometimes known as "PSA dissociated," "TTI unmerged," or "TTI state," but which is called "Unnamed Registered" in the H.323 standards.

EMU PARAMETERS

EMU Inactivity Interval for Deactivation (hours)

Use this field to administer a system-wide administrable interval for EMU deregistration at the visited switch. The allowable entries are the digits between 1 and 24 for hours, or blank. An entry of 1 means that after 1 hour of inactivity, the telephone is dropped from the visited home server. Where the entry is blank, the timer is not used and the visited station remains active until deregistration by another means occurs. This timer is applicable to inter and intra-Communication Manager EMU registrations.

Note:

If SES is enabled for your system, this field is used as the inactivity timer for SIP Visiting Users. For more information on SES and SIP telephones, see *SIP Support in Avaya Communication Manager,* 555-245-206.

Valid entries	Usage
1 to **24** or blank	Enter the interval, in hours, after which a visiting user is dropped due to inactivity. Default is **1**.

CALL PROCESSING OVERLOAD MITIGATION

Restrict Calls

Indicate the type of calls to block first during overload traffic conditions on the system.

Valid entries	Usage
stations-first	Deny new traffic generated by internal stations, allowing inbound calls only (works best in call center environments).
all-trunk-first	Deny all out-bound calls to trunks, tie-lines and stations, and all station-originated calls.
public-trunks-first	Deny all in-bound calls from trunks and tie-lines.

Field descriptions for page 4

Figure 348: Feature-Related System Parameters screen

```
change system-parameters features                                page 4 of x
                            FEATURE-RELATED SYSTEM PARAMETERS
                             Reserved Slots for Attendant Priority Queue: 5_
                                                Time Before Off-Hook Alert: 10__
                                  Emergency Access Redirection Extension: ________
              Number of Emergency Calls Allowed in Attendant Queue: __
                                                      Call Pickup Alerting? n
                             Temporary Bridged Appearance on Call Pickup? y
                                           Call Pickup on Intercom Calls? y
                                                      Directed Call Pickup? n
                                                Extended Group Call Pickup: flexible
                 Deluxe Paging and Call Park Timeout to Originator? n
     Controlled Outward Restriction Intercept Treatment: tone
    Controlled Termination Restriction (Do Not Disturb): tone
              Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS           Authorization Code Enabled? y
                                         Authorization Code Length: 7
                            Authorization Code Cancellation Symbol? #
                                             Attendant Time Out Flag? n
                                          Display Authorization Code? _
                  Controlled Toll Restriction Replaces: station-station
          Controlled Toll Restriction Intercept Treatment: extension     3000
```

Call Pickup Alerting

This provides pickup group members with a visual indication on the Call Pickup status lamp of calls eligible to be answered via Call Pickup

.

Valid entries	Usage
y/n	Enter **y** to enable Call Pickup Alerting on a system-wide basis.

Call Pickup on Intercom Calls

Valid entries	Usage
y	Enter **y** to allow a user's or Agent LoginID's call, ringing as an intercom call, to be picked up using the Call Pickup or Directed Call Pickup features. This field controls the use of this feature throughout the system.
n	Enter **n** to prevent the use of these features to pickup an intercom call.

Controlled Outward Restriction Intercept Treatment

Enter the type of intercept treatment the caller receives when the call is outward restricted.

Valid entries	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. You select and record the message. The calling party receives indication that the call is receiving Intercept Treatment. Enter the extension number for the announcement in the associated field. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Enter the extension number for the extension in an associated field. Cannot be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed

Controlled Station-to-Station Restriction

Enter the type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid entries	Usage
announcement	If announcement is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
attendant	Intercepted calls are redirected to the attendant.
extension (cannot be a VDN extension)	If extension is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

Enter the type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid entries	Usage
announcement	If **announcement** is entered, complete an associated extension number field.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	If **extension** is entered, complete an associated extension number field. Cannot be a VDN extension,
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Paged calls that are to be parked require separate activation of the Call Park feature. All parked calls that time out return to the attendant.

Valid entries	Usage
y	Enter **y** to enable the Loudspeaker Paging - Deluxe feature that essentially integrates the Loudspeaker Paging and Call Park features. All parked calls that time out (not answered by paged party) return to the parking party.
n	Enter **n** to enable the Loudspeaker Paging feature.

Directed Call Pickup

Feature use by individual stations, attendants, or EAS agents can be controlled by COR.

Valid entries	Usage
y	Enter **y** to allow use of the Directed Call Pickup feature across the system.
n	Enter **n** to prevent feature use.

Emergency Access Redirection Extension

Valid entries	Usage
An assigned extension	Enter the assigned extension number (can be a VDN) where emergency queue overflow will redirect.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid entries	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup not supported.

Number of Emergency Calls Allowed in Attendant Queue

Valid entries	Usage
0 to **75**	Enter the number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid entries	Usage
2 to **342**	Enter the number of calls that can go in to the emergency queue.

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter **y** to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter **n** to prevent the temporary bridged appearance of calls answered with these features.

Time Before Off-Hook Alert

Valid entries	Usage
1 to **3000** seconds	Enter the time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

If this field is not enabled, the caller receives Intercept tone. This flag affects only remote users or incoming calls over trunks requiring an authorization code. This field only appears if **Authorization Codes Enabled** is **y**.

Valid entries	Usage
y/n	Enter **y** if a call is to be routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Authorization Code Cancellation Symbol

Enter the symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. This field only appears when **Authorization Code** is **y**.

Valid entries	Usage
#	Enter the cancellation code **#** if the main and tandem servers/switches are both of the same type.
1	Enter the cancellation code **1** if an Avaya System 85 or DIMENSION PBX switch is part of the complex/network.

Authorization Code Length

This field only appears and must be completed if **Authorization Codes Enabled** is **y**. This is the number of digits that must be assigned to the **Authorization Code (AC)** field on the **Authorization Code** screen.

SECURITY ALERT:
You enhance your system's security by using the maximum length for your authorization code.

Valid entries	Usage
4 to 13 digits	Enter a number that defines the number of digits (length) in the **Authorization Code** field.

Authorization Codes Enabled

This field cannot be administered if Authorization Codes is not enabled on the **System Parameters Customer-Options (Optional Features)** screen.

SECURITY ALERT:
To maintain system security, Avaya recommends that Authorization Codes be used.

Valid entries	Usage
y/n	Enter **y** to enable Authorization Codes on a system-wide basis.

Controlled Toll Restriction Intercept Treatment

Appears when the **Controlled Toll Restriction Replaces** field is **outward** or **station-to-station**. This field applies an intercept treatment to a toll call during the call processing.

Valid entries	Usage
announcement	A sub-field appears to the right if **announcement** is used. If the entry is **announcement**, enter the assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	A sub-field appears to the right if **extension** is used. If the entry is **extension**, enter the extension assigned to station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Toll Restriction Replaces

This field activates the Controlled Toll Restriction feature.

Valid entries	Usage
outward **station-station** **none**	The value that you choose for this field will be replaced by controlled toll restriction. In other words, if you choose station-station, you will not be able to use station-station restrictions unless you reset this field.

Display Authorization Code

This field applies only to DCP, not to BRI or hybrid sets.

SECURITY ALERT:
To enhance your system's security, set **Display Authorization Code** to **n**.

Valid entries	Usage
y	Enter **y** to allow authorization code digits to display on the set during the dialing.
n	Enter **n** if these digits should not display.

Field descriptions for page 5

Figure 349: Feature-Related System Parameters screen

```
change system-parameters features                                    page 5 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint: ____           Lines Per Page: 60       EIA Device Bit Rate:


SYSTEM-WIDE PARAMETERS
                                     Switch Name: ________________
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT:

MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n  MCT Voice Recorder Trunk Group: ___
      Delay Sending Release (seconds)?
SEND ALL CALLS OPTIONS
           Send All Calls Applies to: station
      Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? n
                                             UCID Network Node ID: ___
```

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

EIA Device Bit Rate

This field does not appear for S87XX Servers

Valid entries	Usage
1200 **2400** **4800** **9600**	Enter the required printer speed setting.

Endpoint

Valid entries	Usage
Data module extension	Does not appear for S87XX Series IP-PNC. Associated with the System printer.
eia	Does not appear for S87XX Series IP-PNC. If the DCE jack is used to interface the printer.
SYS_PRNT	Use this value if the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT on the **IP Services** screen.
blank	

Lines Per Page

Valid entries	Usage
24 to **132**	Enter the number of lines per page required for the report.

SYSTEM-WIDE PARAMETERS

COR to Use for DPT

Use this field to indicate the Class of Restriction to use for the Dial Plan Transparency feature (DPT).

.

Valid entries	Usage
station	This is the default. The FRL of the calling station determines whether that station is permitted to make a trunk call and if so, which trunk(s) it is eligible to access.
unrestricted	The first available trunk preference pointed to by ARS routing is used.

Emergency Extension Forwarding (min)

If an emergency call should drop (get disconnected), the public safety personnel will attempt to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

This Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same PBX as the extension that dialed 911. Customers who have several PBXs in a campus should assign emergency location extensions accordingly.

This field sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off (drops).

Valid entries	Usage
0 to **999**	Type a number between 0 and 999 that represents the time (in minutes) that an incoming trunk call will forward to the extension that made the initial 911 call. The default value for both new installs and upgrades is **10**.

Note:

If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes might remain on the timer.

Enable Dial Plan Transparency in Survivable Mode

Use this field to enable/disable Dial Plan Transparency (DPT) without changing or removing other feature administration associated with DPT.

Valid entries	Usage
y/n	Enter **y** to enable the Dial Plan Transparency feature should a media gateway register with a local survivable processor (LSP), or a port network register with an Enterprise Survivable Server (ESS). Default is **n**.

Enable Inter-Gateway Alternate Routing

For more information on Inter-Gateway Alternate Routing, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Valid entries	Usage
y/n	Enter **y** to enable the Inter-Gateway Alternate Routing feature. Default is **n**.

Switch Name

Valid entries	Usage
Any keyboard character	Enter up to 20 alpha-numeric characters for identification.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Valid entries	Usage
y/n	Enter **y** to provide an audible tone to the controlling station when an MCT recorder is actively recording a malicious call.

Delay Sending Release (seconds)

This field specifies the amount of time DEFINITY waits before sending an ISDN release message in response to receiving an ISDN disconnect message. This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Malicious Call Trace** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number in increments of 10.

MCT Voice Recorder Trunk Group

Assign the trunk group for MCT voice recorders.

Valid entries	Usage
1 to **666** or blank	group number for DEFINITY CSI
1 to **2000** or blank	group number for S87XX Series IP-PNC

SEND ALL CALLS OPTIONS

Auto Inspect on Send All Calls

Valid entries	Usage
y	If set to **y**, allows you to be presented automatically with Calling Party information for calls which are silently alerting their station because of the Send-All-Calls feature.
n	If set to **n**, you are not guaranteed a Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature.

Send All Calls Applies to

Valid entries	Usage
station	If set to **station**, any call to that station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to become Ring-Ping notification if **Redirect Notification** field is **y**.
extension	When set to **extension**, only the calls sent to that extension are placed to coverage.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Valid entries	Usage
y	If set to **y**, DEFINITY will generate UCID for each call when necessary.
n	If set to **n**, the DEFINITY will not generate a UCID for any call.

UCID Network Node ID

Enter a number unique to this server/switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank	This number is an important part of the UCID tag and must be unique to the server/switch.

Field descriptions for page 6

Figure 350: Feature-Related System Parameters screen

```
change system-parameters features                                     page 6 of x
                          FEATURE-RELATED SYSTEM PARAMETERS
       Public Network Trunks on Conference Call: 5             Auto Start? n
    Conference Parties with Public Network Trunks: 6             Auto Hold? n
 Conference Parties without Public Network Trunks: 6        Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180       Bridging Tone? n
                 Short Interdigit Timer (seconds): 3      Conference Tone? n
              Unanswered DID Call Timer (seconds): ____    Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30    Special Dial Tone? n
                 Long Hold Recall Timer (seconds):     Mode Code Interface? n
                      Reset Shift Timer (seconds): 0
      Station Call Transfer Recall Timer (seconds): 0      Recall from VDN? n
                               DID Busy Treatment: tone

                Allow AAR/ARS Access from DID/DIOD? _
                  Allow ANI Restriction on AAR/ARS? _
      Use Trunk COR for Outgoing Trunk Disconnect? _
                   7405ND Numeric Terminal Display? n          7434ND? n
DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                        Attendant Originated Calls:
    DTMF Tone Feedback Signal to VRU - Connection: _    Disconnection: _
```

7405ND Numeric Terminal Display

Valid entries	Usage
y/n	If enabled, this allows you to use 7405ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of voice messaging systems. This numeric display setting sends only numbers, and not names, to the messaging system.

7434ND

Valid entries	Usage
y/n	If enabled, this allows you to use 7434ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of messaging systems. Use this value if your voice messaging system operates in Bridged Mode.

Allow AAR/ARS Access from DID/DIOD

Valid entries	Usage
y/n	Enter **y** to allow calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only) If a call is placed over a Russian shuttle trunk or a Russian rotary trunk via an AAR or ARS entry with the **ANI Req** field set to **r**, then ANI is requested just like a **y** entry. However, if the ANI request fails, the call immediately drops. All other trunk types treat the **r** entry as a **y**.

Valid entries	Usage
y	The **ANI Req** field on the AAR and ARS Digit Analysis Table and the AAR and ARS Digit Conversion Table permits the additional value of **r**(estricted).
n	The **ANI Req** field on the two screens takes only the current values of **n** and **y**.

Attendant Originated Calls

Valid entries	Usage
internal **external** **priority**	Indicate which type of ringing applies to attendant-originated calls.

Attendant Tone

Valid entries	Usage
y/n	Enter **y** to provide call progress tones to the attendants.

Auto Hold

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Hold feature on a system-wide basis.

Auto Start

If this field is enabled, the **Start** buttons on all attendant consoles are disabled.

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Start feature.

Bridging Tone

Valid entries	Usage
y/n	Enter **y** to apply a bridging tone when calls are bridged on primary extensions.

Conference Parties with Public Network Trunks

If the value of the **Public Network Trunks on Conference Call** field is **0**, this field will not appear on the screen.

Valid entries	Usage
3 to **6**	Specify the maximum number of parties allowed in a conference call involving a public network subscriber.

Conference Parties without Public Network Trunks

Valid entries	Usage
3 to **6**	Enter a number to specify the maximum number of parties allowed in a conference call involving no public network trunks.

Conference Tone

Note:

Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones will be equivalent to no tone (silence) unless the tone is independently administered or customized on the **Tone Generation** screen.

Valid entries	Usage
y/n	Enter **y** to provide conference tone as long as three or more calls are in a conference call.

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid entries	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

Intrusion Tone

Valid entries	Usage
y/n	Enter **y** to apply an intrusion tone (executive override) when an attendant intrudes on the call.

Line Intercept Tone Timer (seconds)

Valid entries	Usage
0 to **60**	Enter a number to specify how long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

You can administer the system to remind a user that a call has been on hold for too long.

Valid entries	Usage
0 to **999**	Enter a number between **0** and **999**; **0** deactivates the timer. This value is the number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Note:
If you make a change to this field, you must log off and log back on to effect the permission changes to get to the Mode Code Related System Parameters on page 1343.

Valid entries	Usage
y/n	A **y** entry allows you to use the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to INTUITY AUDIX or other vendors' voice-mail systems.

Night Service Disconnect Timer (seconds)

Valid entries	Usage
10 to **1024** or blank	Enter a number or blank to indicate how long a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public Network Trunks on Conference Call

Indicates the number of public network trunks allowed on a conference call.

Valid entries	Usage
0 to **5**	If this field is **0**, the **Conference Parties with Public Network Trunks** field will not appear on the screen.

Recall from VDN

This feature is available when **Vectoring (Basic)** and **Vectoring (Prompting)** are set to **y**. Use this field to Indicate whether or not a call that is transferred to a VDN and then routed to a station is recalled to the originating station after the **Station Call Transfer Recall Timer** expires.

Valid entries	Usage
y	Calls are recalled from a VDN when the **Station Call Transfer Recall Timer** expires.
n	Calls are not recalled from a VDN when the **Station Call Transfer Recall Timer** expires.

Reset Shift Timer (seconds)

Used only for station-to-station calls or private network calls using ISDN trunks.

Valid entries	Usage
0 to **255**	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. Reset shift dial tone allows the user to dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. Enter **0** to disable this feature.

Short Interdigit Timer (seconds)

Valid entries	Usage
3 to **9**	Enter a number to limit the time that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Valid entries	Usage
y/n	Enter **y** to use the Special Dial Tone. You must have a TN2182 circuit pack.

Station Call Transfer Recall Timer (seconds)

Allows a user-transferred call (station-to-station, a trunk call, or a DCS call) to re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid entries	Usage
0 to **999**	Enter the time in seconds before a call redirects back to the station user who initiated the transfer operation. Enter **0** to disable this feature.

Unanswered DID Call Timer (seconds)

Enter number or blank to limit how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature. This timer interacts with the nonadministrable 50 second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Thus if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored. See Wait Answer Supervision Timer in this section.

Valid entries	Usage
A number between **10** and **1024**	Enter a number to indicate how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature.
blank	Disables the timer.

Use Trunk COR for Outgoing Trunk Disconnect

Use this field to indicate whether the Outgoing Trunk Disconnect Timer is set based on the COR of the originating station or of the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating caller's station.

Valid entries	Usage
n	Default. The Outgoing Trunk Disconnect Timer to be set based on the COR of the originating station. This is the default.
y	Enter **y** to enable the Outgoing Trunk Disconnect Timer to be set based on the COR of the trunk instead of the originating station.

DISTINCTIVE AUDIBLE ALERTING

The following Distinctive Audible Alerting fields appear when **Tenant Partitioning** on the **System Parameters Customer Options** screen is **n**. Use these fields to administer distinctive ring patterns for your system.

Attendant Originated Calls

This field appears when [Tenant Partitioning]() on the **System Parameters Customer Options** screen is **n**.

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls. Default is **external**.

Distinctive Audible Alerting (Internal, External, Priority)

This field appears when [Tenant Partitioning]() on the **System Parameters Customer Options** screen is **n**.

This is also known as Distinctive Ringing. Enter the number of rings for **Internal**, **External**, and **Priority** calls. For virtual stations, this applies to the mapped-to physical telephone. Defaults are as follows:

- **1**: Internal calls
- **2**:External and attendant calls
- **3**:Priority calls

Note:

SIP Enablement Services (SES) messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid entries	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

This field appears only if **DTMF Feedback Signals for VRU** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
0 to **9**, *****, **#**, **A**, **B**, **C**, **D**	Enter the code to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	Blank means that no tone is to be sent to the VRU.

Field descriptions for page 7

Figure 351: Feature-Related System Parameters screen

```
change system-parameters features                               page 7 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer?                    No Dial Tone Conferencing?
        Transfer Upon Hang-Up?        Select Line Appearance Conferencing?
Abort Conference Upon Hang-Up?                                      Unhold?
   No Hold Conference Timeout:     Maximum Ports per Expanded Meet-me Conf:
                                    External Ringing for Calls with Trunks?
ANALOG BUSY AUTO CALLBACK
                Without Flash?                         Announcement:
                                        Voice Mail Hunt Group Ext:

AUDIX ONE-STEP RECORDING
                            Recording Delay Timer (msec):
Apply Ready Indication Tone To Which Parties In The Call?
  Interval For Applying Periodic Alerting Tone (seconds):
```

CONFERENCE/TRANSFER

Abort Conference Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to abort the conference operation when they hang up.

Valid entries	Usage
y/n	Enter **y** to change a call placed on soft-hold in the conference-pending status to hard-held status if the user hangs up.

Abort Transfer

Stops the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when they hang up. If both the **Abort Transfer** and **Transfer Upon Hang-Up** fields are **y** and you press the **transfer** button and then dial the complete transfer-to number, hanging up the telephone transfers the call. You must select another non-idle call appearance to abort the transfer. If the **Transfer Upon Hang-Up** field is **y**, hanging up completes the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless telephones.

Valid entries	Usage
y/n	Enter **y** to abort the transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up or selecting another non-idle call appearance. The user must press the **Transfer** button again to complete the process unless **Transfer Upon Hang-up** is also set to **y**.

External Ringing for Calls with Trunks

Use this field to specify existing ringing behavior or external call ringing behavior on external trunk calls that are transferred or conferenced by stations or Attendants, or extended by the Attendant to an "on-switch" extension.

Valid entries	Usage
y	Enter **y** to activate external ringing for transferred external trunk calls.
n	Enter **n** to use existing ringing behavior. This is the default value.

Maximum Ports per Expanded Meet-me Conf

This field allows you to administer the maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit (i.e., not administrable on a per Expanded-Meet-me VDN basis). This field is hidden if **Maximum Number of Expanded Meet-me Conference Ports** is **0** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
3 to **300**	Enter the maximum number of parties allowed for each conference on your system.

No Dial Tone Conferencing

When another line is on hold or alerting, No Dial Tone Conferencing eliminates dial tone while setting up a conference.

Valid entries	Usage
y/n	Enter **y** to activate No Dial Tone Conferencing.

No Hold Conference Timeout

Controls the timeout of No Hold Conference call setup. The system Answer Supervision timer should be set to a value less than this.

Valid entries	Usage
20 to **120**	Enter the number of seconds.

Select Line Appearance Conferencing

Use this field to specify that the user can use the line appearance rather than the **Conf** button to include a call in a conference. If a user is on a call, and another line is on hold or an incoming call alerts on another line, the user can press the **Conf** button to bridge the calls together. Using the select line appearance capability, the user can press a line appearance button to complete a conference instead of pressing the **Conf** button a second time.

Valid entries	Usage
y/n	Enter **y** to activate Select Line Appearance Conferencing

Transfer Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to complete a transfer operation by hanging up.

Valid entries	Usage
y/n	Enter **y** so users can transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up. The user can also wait to hang up, speak with the other party, then press **Transfer** again to complete the process. With this field set to **y**, users of the Call Park FAC can park a call without having to press the **Transfer** button a second time.

Unhold

Allows the user to press the **hold** button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI telephones or attendant consoles.

Valid entries	Usage
y/n	Enter **y** to activate the unhold capability

ANALOG BUSY AUTO CALLBACK

With the Analog Busy Auto Callback Without Flash (ACB) feature enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path nor forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter **1** for ACB or **2** to cover to a voice mail hunt group extension.

Announcement

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter the extension of the announcement you want to play for the ACB feature. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter a voice mail hunt group extension to which the call is to be forwarded if the user enters 2 at the ACB announcement prompt.

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Note:

If the **Analog Busy Auto Callback Without Flash** field is set to **y**, the **Busy Auto Callback without Flash** field on the **Station** screen defaults to **y** (enabled) for all analog station types that allow Analog Auto Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

AUDIX ONE-STEP RECORDNG

On stations administered with this feature button, this feature allows users to activate and deactivate the recording of active calls to their Audix with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

This field is for administering who hears the Audix recording ready tone.

Valid entries	Usage
all, initiator, or **none**	Enter a value for which party or parties on the call should hear the ready-to-record indication tone. The default is **all**. This field cannot be left blank.

Interval For Applying Periodic Alerting Tone (seconds)

Appears only if the **Apply Ready Indication Tone To Which Parties In The Call** field is set to **all**.

Valid entries	Usage
0 to **60**	Enter a number from zero to 60 for the number of seconds desired between alerting tones, where zero disables the tone. The default value is a 15 second interval.

Recording Delay Timer (msecs)

Valid entries	Usage
0 to **4000** in increments of 100	Use this field to administer a delay interval before starting audix recording.

Field descriptions for page 8

Figure 352: Feature-Related System Parameters screen

```
change system-parameters features                               page 8 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS
                                                         PARAMETERS FOR CREATING
Send Non-ISDN Trunk Group Names as Connected Name?       QSIG SELECTION NUMBERS
 Display Connected Name/Number for ISDN DCS Calls?         Network Level:
       Send ISDN Trunk Group Name on Tandem calls?          Level 2 Code:
                                                            Level 1 Code:
               Send Custom Messages Through QSIG?
                         QSIG/ETSI TSC Extension:
MWI - Number of Digits Per Voice Mail Subscriber:
                                Feature Plus Ext:
                             National CPN Prefix:
                        International CPN Prefix:
                       Pass Prefixed CPN to ASAI:
   Unknown Numbers Considered Internal for AUDIX?
             UNSI Calling Name for Outgoing Calls?
               Path Replacement with Measurements?
                  QSIG Path Replacement Extension:
            Path Replace While in Queue/Vectoring?
```

ISDN PARAMETERS

Display Connected Name/Number for ISDN DCS Calls

Valid entries	Usage
y/n	Enter **y** to display the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

Valid entries	Usage
A valid extension	Administration of this field is required for proper termination of some Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension in order to send the indication to the appropriate server running Communication Manager. Appears only if the **ISDN Feature Plus** field is **y** on the System Parameters Customer-Options (Optional Features) screen.

International CPN Prefix

Allows you to apply prefixes to international calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to international calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and **#** or blank	Enter a number that allows you to apply prefixes to international calling numbers for display.

Maximum Length

Appears only if the **Unknown Numbers Considered Internal for AUDIX** field is **y**. Indicates the maximum length of an unknown private number. Any unknown number longer than the administered value is considered external. This field cannot be blank when it appears.

Valid entries	Usage
1 to **20**	Enter a number for the maximum length of an unknown private number.

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the **Basic Supplementary Services** field or the **ISDN Feature Plus** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**. This field provides an indication of the number of digits per AUDIX subscriber.

Note:

For QSIG-MWI, these routing digits and inserted digits must screen a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message- center server/ switch to a served-user switch), then every lamp update message places the **Inserted Digits** field (from the **Message Waiting Indication Subscriber Number Prefixes** screen) in front of the AUDIX subscriber number to screen a complete QSIG network number for the served user.

For Feature Plus MWI, the routing digits and inserted digits must screen a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (served user) switch. The **Inserted Digits** field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Enter a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the **Extension Length** field on the **Switch Interface Administration** screen of AUDIX.

National CPN Prefix

Allows you to apply prefixes to national calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and **#** or blank	Enter a number that allows you to apply prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

Passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Valid entries	Usage
y/n	Enter **y** to pass CPN information to ASAI.

Path Replacement While in Queue/Vectoring

Valid entries	Usage
y/n	Enter **y** to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS you are using, some calls can go unrecorded if you enable this capability. Please see your Avaya technical support representative for more information.

Path Replacement with Measurements

Valid entries	Usage
y/n	Allows QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG Path Replacement Extension

Enter the extension for the system to use as part of the complete number sent in the Path Replacement Propose message.

Valid entries	Usage
Extension	Enter an unused extension that conforms to your dial plan.

QSIG/ETSI TSC Extension

Valid entries	Usage
Enter any valid, unassigned extension.	This is the phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs), which are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing). ETSI protocol TSCs as well as QSIG TSCs are supported.

Send Custom Messages Through QSIG?

Valid entries	Usage
y/n	Enter **y** to provide appropriate display information, for example for the Posted Messages feature, over QSIG links.

Send ISDN Trunk Group Name on Tandem Calls

Valid entries	Usage
y/n	Enter **y** to provide consistent display information regardless of trunk type. If set to **y**, provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

Valid entries	Usage
y/n	Enter **y** to send a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Appears when, on the **System Parameters Customer-Options (Optional Features)** screen, either the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**. This field controls the treatment of an ISDN number whose numbering plan identification is "unknown" in a QSIG centralized AUDIX arrangement. This field works in conjunction with the Calling Party Number to INTUITY AUDIX field on the **Hunt Group** screen. The **Calling Party Number to INTUITY AUDIX** field on the **Hunt Group** screen must be **y** for this field to have an effect.

Valid entries	Usage
y	The unknown number is considered "internal" and AUDIX tries to find a calling party name match for the digit string. If a name match is found, AUDIX provides the calling party's name. If no name is found, AUDIX provides the calling party's telephone number.
n	The unknown number is considered "external" and AUDIX provides the calling party's telephone number.

USNI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	Enter **y** to send a name on outgoing calls over NI PRI trunks. Important: Be sure you have validated that your service provider's central office is capable of accepting calling name information from Communication Manager in this way. For example, if the central office has a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability might cause the central office to drop outgoing calls from your Avaya S8XXX Server. In this case, change the value in this field to **n**. Enter **n** to prevent sending calling name information with outgoing calls over NI PRI trunks. **n** is this field overrides a **y** in the **Send Name** field of the outgoing **Trunk Group** screen.

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS

Level 1 Code

Enter the first level regional code of the Avaya S8XXX Server in the network. Administer this field carefully. Communication Manager will not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **1** or **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is **1** or **2**.

In QSIG standards, this level 1 code is called the Level 0 Regional Code.

Level 2 Code

Enter the second level regional code of the Avaya S8XXX Server in the network. Administer this field carefully. The system will not check to ensure you have entered a code that supports your entry in the **Network Level** field. You cannot enter anything in this field unless the **Network Level** field is set to **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the **Network Level** field is 2.

In QSIG standards, this level 2 code is called the Level 1 Regional Code.

Network Level

Enter the value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/ Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid entries	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the **Send Calling Number** and/or **Send Connected Number** field is **y** or **r** with **private** specified for the **Numbering Format** field on the **ISDN Trunk Group** screen, the Calling Number and/or Connected Number IEs will not be sent. If the field is left blank but **private** has been specified in the **Numbering Format** field on the **ISDN Trunk Group** screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as **PresentationRestricted** with **NULL** for the associated digits.

Field descriptions for page 9

Figure 353: Feature-Related System Parameters screen

```
change system-parameters features                              page 9 of x
                          FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS

    CPN/ANI/ICLID Replacement for Restricted Calls:
   CPN/ANI/ICLID Replacement for Unavailable Calls:

DISPLAY TEXT
      Identity When Bridging: principal

INTERNATIONAL CALL ROUTING PARAMETERS
               Local Country Code: 1
        International Access Code: 011

ENBLOC DIALING PARAMETERS
Enable Enbloc Dialing without ARS FAC?

CALLER ID ON CALL WAITING PARAMETERS
Caller ID on Call Waiting Delay Timer (msec): 200
```

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN/ANI/ICLID Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

DISPLAY TEXT

Identity When Bridging

Use this field to determine whether the telephone display shows the literal identity of the bridged appearance or the virtual identity.

Note:

When you choose the **station** option, you must update the **Numbering — Public/Unknown Format** screen with the Extension Codes of the stations that display the caller's or answering party's assigned identification.

Valid entries	Usage
principal	The location from which the caller is bridging in. This is the default.
station	The caller’s and the answering party’s assigned identification.

INTERNATIONAL CALL ROUTING PARAMETERS

Local Country Code

Valid entries	Usage
1 to 3 digits or blank	Enter a valid PSTN E.164 country code for this node. The default is blank (no SBS signaling trunk groups are administered). For example, for an SBS node in the United States, enter **1**. For a list of country codes, see the International Telecommunications Union "List of ITU-T Recommendation E.164 Assigned Country Codes .

International Access Code

Valid entries	Usage
1 to 5 digits or blank	Enter the access code required by the PSTN to route calls out of the country. This code will be included with the telephone number received from the SBS terminating node if the Local Country Codes of the originating and terminating nodes are different. The default is blank (no access code is needed). For example, for an SBS node in the United States, enter **011**.

Note:

Once administered, these fields cannot be cleared until all trunk groups administered for SBS signaling have been removed. For details, see the **Trk Grp(s)** and **Signaling Group** screens.

ENBLOC DIALING PARAMETERS

Enable Enbloc Dialing without ARS FAC

Valid entries	Usage
y/n	Enter **y** to enable Enbloc Dialing without the need to dial a FAC. Default is **n**.

Minimum Digit Length

This field appears only when **Enable Enbloc Dialing without ARS FAC** is **y**.

Valid entries	Usage
1 to **20**	Enter the number of digits before Enbloc Calling Treatment is activated. Default is extension length plus 1.

CALLER ID ON CALL WAITING PARAMETERS

Caller ID on Call Waiting Delay Timer (msec)

Valid entries	Usage
5 to **1275** in increments of 5	Enter the desired delay in 5-millisecond intervals. Default is **200**.

Field descriptions for page 10

Figure 354: Feature-Related System Parameters screen

```
change system-parameters features                                page 10 of x
                         FEATURE-RELATED SYSTEM PARAMETERS
                 Pull Transfer: n           Update Transferred Ring Pattern? n
         Outpulse Without Tone? y             Wait Answer Supervision Timer? n
         Misoperation Alerting? n              Repetitive Call Waiting Tone? y
    Allow Conference via Flash? y    Repetitive Call Waiting Interval (sec): _
 Vector Disconnect Timer (min): __   Network Feedback During Tone Detection? y
   Hear Zip Tone Following VOA? y   System Updates Time On Station Displays? n
      Intercept Treatment on Failed Trunk Transfers? n
                    Station Tone Forward Disconnect: silence
                             Level Of Tone Detection: precise
           Charge Display Update Frequency (seconds): 30
         Date Format on 607/2400/4600/6400 Terminals: mm/dd/yy
On-hook Dialing on 607/2400/4600/6400/8400 Terminals? n

ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? y
   Apply Intercept Locally? _      Enforce PNT-to-PNT Restrictions? _
```

Allow Conference via Flash

Valid entries	Usage
y	Enter **y** to allow an analog station to use flash to conference calls together.
n	Enter **n** to prevent this.

Charge Display Update Frequency (seconds)

This applies only if you use Advice of Charge or Periodic Pulse Metering with display functions.

Valid entries	Usage
10 to **60** or blank	The amount of time (in seconds) between charge-display updates. Frequent display updates might have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display will not automatically show charge information. To see charge information for a call, the user must have a **disp-chrg** button and must press the button before the call drops.

Date Format on 607/2400/4600/6400 Terminals

The format of the date as displayed on the telephones.

Valid entries	Usage
mm/dd/yy	month/day/year
dd/mm/yy	day/month/year
yy/mm/dd	year/month/day

Hear Zip Tone Following VOA?

This tone alerts a telephone user that the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

Note:
This field does not effect auto-answer zip tone heard prior to the VOA.

Valid entries	Usage
y	Enter **y** to play zip tone following a VDN of Origin Announcement (VOA).
n	Enter **n** if you do not want zip tone following a VOA.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

Level of Tone Detection

For the most part, this option is no longer required in today's switching environment. It might be useful if your users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid entries	Usage
broadband	This is the least exact of the levels of tone detection. If Avaya Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Avaya Communication Manager interprets any tone which has a continuous "on" period of longer than 1 second as dial tone. Otherwise, the server accepts whatever the tone detector circuit pack reports.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Misoperation Alerting should not be enabled if Call Prompting is optioned.

Valid entries	Usage
y	Enter **y** for misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Enter **n** for standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

Valid entries	Usage
y/n	Enter **y** to provide audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Valid entries	Usage
y/n	Enter **y** allows users to dial without lifting the handset. If you enable this, users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

The next four fields control station-to-switch recall signal timing. If a flashhook interval (recall window) is required, the upper and lower bounds of the interval can be administered. An on-hook that lasts for a period of time greater than or equal to the lower bound and less than or equal to the upper bound will be treated as a recall flash. If an interval is not required, the **Disconnect Timing** value must be administered. An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect. Regardless, an on-hook lasting 50 to 150 ms coming from a 2500-type set will always be treated as a digit pulse unless **Ignore Rotary Digits** is **y** for that station.

Outpulse Without Tone

Valid entries	Usage
y	Enter **y** to indicate the server will outpulse digits even when a dial tone has not been received.
n	Enter "**n**" if the calling party should receive intercept tone if no dial tone is detected.

Pull Transfer

Valid entries	Usage
y/n	Enter **y** to enable the Pull Transfer feature on a system-wide basis. This allows either the transferring or transferred-to party to press the **Transfer** button to complete the transfer operation

Repetitive Call Waiting Interval (sec)

This field appears when the **Repetitive Call Waiting Tone** field is **y**.

Valid entries	Usage
1 to **99**	Enter a number to specify the number of seconds between call waiting tones.

Repetitive Call Waiting Tone

Valid entries	Usage
y/n	Enter **y** to indicate that a repetitive call waiting tone be provided to the called party for all types of call waiting access.

Station Tone Forward Disconnect

Tone Forward Disconnect applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid entries	Usage
busy **intercept** **silence**	When a station is the last party remaining off-hook on a call, that station receives the indicated tone or silence until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

This does not apply to telephones (such as BRI telephones) where the user sets the time.

Valid entries	Usage
y/n	Enter **y** to have the system automatically update the time on display telephones when background maintenance is run (for example, when the set is plugged in).

Update Transferred Ring Pattern

Valid entries	Usage
y/n	Enter **y** to change the ringing pattern from internal to external when an internal station transfers an external call. If most of your calls go through an attendant, you might want to set this to **y**, so your users will be able to distinguish an external call.

Vector Disconnect Timer (min)

Enter the number of minutes, or blank that a trunk should remain connected to a vector.

Valid entries	Usage
1 to **240**	The number of minutes that you enter determines when the trunk will be disconnected if the **Disconnect Supervision-In** or **Disconnect Supervision-Out** fields on the **Trunk Group** screen are **n**.
blank	Enter blank if you do not want Avaya Communication Manager to initiate a disconnect.

Wait Answer Supervision Timer

See Unanswered DID Call Timer (seconds) for more information.

Valid entries	Usage
y	Enter **y** to enable this feature on a system-wide basis. When **y** is entered in this field, calls to stations unanswered after 50 seconds are dropped.
n	When **n** is entered in this field, unanswered calls drop only when the calling party goes on-hook.

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Apply Intercept Locally

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment will be applied locally instead of on the originating server/switch.

Enforce PNT-to-PNT Restrictions

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that restrictions and denial of PNT-to-PNT connections will be enforced when the EDCS message is unavailable. A **y** in this field means restrictions will be enforced.

Italian Protocol Enabled

Valid entries	Usage
y/n	Enter **y** to enable the Italian DCS feature on a system-wide basis.

Field descriptions for page 11

Figure 355: Feature-Related System Parameters screen

```
change system-parameters features                                page 11 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS
  EAS
        Expert Agent Selection (EAS) Enabled? n
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension: ________          Delay: ___
   Message Waiting Lamp Indicates Status For: station
VECTORING
                   Converse First Data Delay: 0       Second Data Delay: 2
              Converse Signaling Tone (msec): 100          Pause (msec): 70_
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? n
                           BSR Tie Strategy? 1st_found
  Store VDN Name in Station's Local Call Log? n

SERVICE OBSERVING
             Service Observing: Warning Tone? n      or Conference Tone? n
    Service Observing Allowed with Exclusion? n
            Allow Two Observers in Same Call? n
```

CALL CENTER SYSTEM PARAMETERS

EAS

Direct Agent Announcement Delay

Only appears if **Expert Agent Selection (EAS)** or **ASAI Link Core Capabilities** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
0 to **99** or blank	Enter the number of seconds the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

Valid entries	Usage
Valid extension	Enter the extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

To enable this field, either no ACD or vectoring hunt groups might exist or, existing ACD or vectoring hunt groups must be "skilled." Only appears if **Expert Agent Selection (EAS)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
y/n	Enter **y** to enable Expert Agent Selection.

Message Waiting Lamp Indicates Status For

Only appears if **Expert Agent Selection (EAS)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a telephone, you need to indicate if the message is for at the telephone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Minimum Agent-LoginID Password Length

Enter the minimum number of digits that must be administered as an EAS Agent's LoginID password. Only appears if **Expert Agent Selection (EAS)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
0 to **9**	Entering a **0** or blank indicates no password is required.

VECTORING

Available Agent Adjustments for BSR

Controls the use of BSR available agent adjustments. The **Vectoring (Best Service Routing)** field must be **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to allow adjustments to available agents.

BSR Tie Strategy

This field appears only when **Vectoring (Best Service Routing)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Converse First Data Delay/Second Data Delay

The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. Only appears if **Vectoring (Basic)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
0 to **9**	Number of seconds for the delay.

Converse Signaling Tone/Pause

Only appears if **Vectoring (Basic)** and **DTMF** on the **System Parameters Customer-Options (Optional Features)** screen are **y**. In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum timer settings for the CONVERSANT or IR are 60 msec tone and 60 msec pause.

Valid entries	Usage
40 to **2550** (in increments of 10).	Values entered in the **Tone/Pause** fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits. ● **TN742B or later suffix analog board** — Tone and pause round up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone. ● **TN464F, TN767E or later suffix DS1 boards** — Tone and pause round up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers might stay in effect. To reset your timers to the system default, pull and reseat the circuit pack.
100	

Interflow-qpos EWT Threshold

Displays only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Lookahead Interflow (LAI)** field is **y**. Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources).

Valid entries	Usage
0 to **9** or blank	Number of seconds for this threshold

Prompting Timeout (secs)

Only appears if **Vectoring (Prompting)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
4 to **10**	Enter the number of seconds before the `Collect Digits` command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The "*" is interpreted as a "caller end-of-dialing indicator and the "#" is an indicator to clear all digits previously entered by the caller for the current "collect" vector step.

Valid entries	Usage
y/n	Enter **y** to reverse the star and pound digits by the "collect" vector step. If set to **y**, it does not affect any other DEFINITY vector step or other non-ACD DEFINITY feature (such as ARS) in that the "*" and "#" digit-processing is unchanged.

Store VDN Name in Station's Local Call Log

Specifies if Communication Manager sends a message telling the telephone to store the VDN name or the calling party's name in the station call log for any of the following telephones:

- 2420
- 4610
- 4620
- 4625

Valid entries	Usage
y	Communication Manager sends a message telling the telephone to store the VDN name in the station call log.
n	Communication Manager sends a message telling the telephone to store the calling party's name in the station call log. This is the default setting.

SERVICE OBSERVING

Allow Two Observers in Same Call

Use this field to set, on a system-wide basis, the number of service observers allowed in a call to two.

Valid entries	Usage
y/n	When set to **y**, two service observers can monitor the same EAS Agent LoginID or station extension, and up to two service observers can be on the same two-party call or in a conferenced call having more than two parties.

Service Observing: Warning Tone

Service Observing (Basic) on the **System Parameters Customer-Options (Optional Features)** screen must be **y** before this field can be administered.

CAUTION:
The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Valid entries	Usage
y/n	Enter **y** to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Conference Tone?** is set to **y**.

or Conference Tone

Service Observing (Basic) on the **System Parameters Customer-Options (Optional Features)** screen must be **y** before this field can be administered.

Valid entries	Usage
y/n	Enter **y** to assign a conference tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Warning Tone?** is set to **y**.

Service Observing Allowed with Exclusion

Allows Service Observing of a station with Exclusion active, either by Class Of Service or by manual activation of Exclusion. Default is **n**.

Valid entries	Usage
y	Enter **y** to allow Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
n	Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped. This is the default.

Field descriptions for page 12

Figure 356: Feature-Related System Parameters screen

```
change system-parameters features                           page 12 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AGENT AND CALL SELECTION
                     MIA Across Splits or Skills? n
                      ACW Agents Considered Idle? y
                      Call Selection Measurement: current-wait-time
Service Level Supervisor Call Selection Override? y
                             Auto Reserve Agents:
        Copy ASAI UUI During Conference/Transfer?
ASAI
Call Classification After Answer Supervision? n      Send UCID to ASAI? n

CALL MANAGEMENT SYSTEM
                       REPORTING ADJUNCT RELEASE
                                  CMS <appl mis>: R3V11
                                   IQ <appl ccr>:

                 ACD Login Identification Length: 0
                           BCMS/VuStats LoginIDs?
               BCMS/VuStats Measurement Interval: hour
       BCMS/VuStats Abandon Call Timer (seconds):
                 Validate BCMS/VuStats Login IDs? n
                        Clear VuStats Shift Data: on-login
             Remove Inactive BCMS/VuStats Agents? n
```

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid entries	Usage
y/n	Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid entries	Usage
all	Puts an agent on stand-by for all skills.
none	Agent is not on stand-by for any additional skills.
secondary-only	Puts an agent on stand-by only for secondary skills.

Call Selection Measurement

This field determines how Avaya Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

For information on Business Advocate, please contact your Avaya representative or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
current-wait-time	**Current Wait Time** selects the oldest call waiting for any of the agent's skills.
predicted-wait-time .	**Predicted Wait Time** is a feature of Business Advocate.

Copy ASAI UUI During Conference/Transfer

Displays when, on the **System Parameters Customer-Options (Optional Features)** screen, either the **ASAI Interface** or **ASAI Proprietary Adjunct Links** field is **y**.

Valid entries	Usage
y/n	Enter **y** to copy user-to-user (UUI) information during a conference or transfer calls.

Note:

When this field is set to **y**, the system actually copies *all* UUI information, not just ASAI UUI. Copying only occurs during a human-initiated conference or transfer. Communication Manager does not copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Valid entries	Usage
y/n	Enter **y** to remove an agent from the MIA queue for all the splits/skills/hunt groups that he or she is available in when the agent answers a call from any of his or her splits/skills/hunt groups.

Service Level Maximizer Algorithm

This field displays only if **Service Level Maximizer** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
weighted	Use the WSL algorithm for Service Level Maximizer calculations.
actual	Use the ASL algorithm for Service Level Maximizer calculations. This is the default.

Service Level Supervisor Call Selection Override

This field determines whether Avaya Communication Manager changes agents' call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold.

For information on Business Advocate, please contact your Avaya Account Executive or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
y	Enter **y** if you want to override the normal call handling preferences of a skill's assigned agents in this situation.
n	Enter **n** if you do not want to override agents' normal call handling preferences when the skill exceeds its Level 1 threshold. Service Level Supervisor requires Expert Agent Selection and Business Advocate.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid entries	Usage
y/n	Enter **y** to force the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection. The default value **n** always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Valid entries	Usage
y/n	Enter **y** to enable transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEM

REPORTING ADJUNCT RELEASE

CMS (appl mis)

Valid entries	Usage
R12	CMS R12 is connected to the mis1 link, and to the mis2 link for a second CMS. The **IQ** field must be blank.
R13	CMS R13 is connected to the mis1 link, and to the mis2 link for a second CMS. The **IQ** field must be blank.
R13.1	CMS R13.1 is connected to the mis1 link, and to the mis2 link for a second CMS. Reporting adjuncts CMS, Avaya IQ, or both can be connected. Only CMS R13.1 or R14 are allowed with Avaya IQ 4.0.
R14	CMS R14 is connected to the mis1 link, and to the mis2 link for a second CMS. Reporting adjuncts CMS, Avaya IQ, or both can be connected. Only CMS R13.1 or R14 are allowed with Avaya IQ 4.0.
blank	A CMS system is not connected. If any entry on the dial plan is set to greater than 7 digits, this field must be blank. CMS supports a maximum of 7 digits. This is the default.

IQ (appl ccr)

Valid entries	Usage
4.0	Enter the release of the Avaya IQ system that will be connected to the ccr1 link, and to the ccr2 link for a second Avaya IQ. With Communication Manager Release 4.0, only Avaya IQ Release 4.0 is valid. EAS and UCID must be active before this form is submitted for Avaya IQ connection. Reporting adjuncts CMS, Avaya IQ, or both can be connected. Only CMS R13.1 or R14 are allowed with Avaya IQ 4.0. When CMS is set to R13.1 the Avaya IQ Add Resource Screen Description field for Communication Manager must be set to Communication Manager 3.1. With Communication Manager 4.0 or 5.0, the Avaya IQ Description field should be set to Communication Manager 4.0 with an R14 CMS or no connected CMS.
blank	An IQ system is not connected. This is the default.

OTHER CALL MANAGEMENT SYSTEM FIELDS

ACD Login Identification Length

Enter the number of digits for an ACD Agent Login ID if **Expert Agent Selection (EAS)** on the **System Parameters Customer-Options (Optional Features)** screen is **n**. If **BCMS/VuStats Login IDs** is **y**, the ACD Login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of characters in the agent's login ID.

Valid entries	Usage
0 to **9**	For CMS, this field cannot be **0**.

BCMS/VuStats Abandon Call Timer (seconds)

Valid entries	Usage
1 to **10** or blank	Specify the number of seconds before calls are considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

BCMS/VuStats LoginIDs

This feature can be used when EAS is not optioned, or in addition to EAS login IDs. When this field is **y**, both BCMS and CMS use the same login ID for an agent.

Valid entries	Usage
y/n	Enter **y** to administer valid agent login IDs to monitor call activity by agent.

BCMS/VuStats Measurement Interval

You can enter **half-hour** or **hour** for polling and reporting measurement data if the **BCMS (Basic)** and/or the **VuStats** on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
half-hour	There are a maximum of 25 time slots available for measurement intervals. If **hour** is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval can be changed at any time, but will not go into effect until the current interval completes.
hour	

Clear VuStats Shift Data

Valid entries	Usage
on-login	Enter **on-login** to clear shift data for an agent when the agent logs in.
at-midnight	Enter **at-midnight** to clear shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid entries	Usage
y	Agents are removed from reports when they have no staff time during the previous 7 days.
n	Agents remain on the report even if they have no staff time for any period of time.

Validate BCMS/VuStats Login IDs

Valid entries	Usage
y	Enter **y** to allow entry only of login-IDs that have been entered on the **BCMS Login-ID** screen.
n	Enter **n** to allow entry of any ACD login of the proper length.

Field descriptions for page 13

Figure 357: Feature-Related System Parameters screen

```
change system-parameters features                           page 13 of x
                         FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER MISCELLANEOUS
                                Clear Callr-info:
               Allow Ringer-off with Auto-Answer?
```

Allow Ringer-off with Auto-Answer

Valid entries	Usage
y/n	Enter **y** to allow a user to use the **ringer-off** feature button to prevent ringing on EAS auto-answer calls.

Clear Callr-info

Use this field to specify when the collected digits Callr-Info display is to be removed from the agent/station display.

Valid entries	Usage
leave-ACW	Leaves the display up while the agent is in ACW (After-call work mode).
next-call	Clears the display when the next call is received. This is the default.
on-call-release	Clears the display on the 2nd line of a two-line display as soon as the call is released, either because of receiving call disconnect or the agent/ station user pressing the release button.

Field descriptions for page 14

Figure 358: Feature-Related System Parameters screen

```
change system-parameters features                              page 14 of x
                         FEATURE-RELATED SYSTEM PARAMETERS
REASON CODES
                                Aux Work Reason Code Type:
                                  Logoff Reason Code Type:
                         Two-Digit Aux Work Reason Codes?:

REDIRECTION ON IP CONNECTIVITY FAILURE

                Switch Hook Query Response Timeout:
            Auto-answer IP Failure AUX Reason Code:

MAXIMUM AGENT OCCUPANCY PARAMETERS
                Maximum Agent Occupancy Percentage:
           Maximum Agent Occupancy AUX Reason Code:

FORCED AGENT LOGOUT PARAMETERS
   Maxiumum Time Agent in ACW before Logout (sec.):
                     ACW Forced Logout Reason Code:
```

REASON CODES

Aux Work Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.

Logout Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System Parameters Customer-Options (Optional Features)** screen must be **y**.

Two-Digit Aux Work Reason Codes

Valid entries	Usage
y/n	Enter **y** to enable two-digit Reason Codes for agent state changes to Aux Work. Default is **n**.

REDIRECTION ON IP CONNECTIVITY FAILURE

Switch Hook Query Response Timeout

Valid entries	Usage
500 to **5000** (msec)	Assign the time on a system basis that the call processing will wait for a response to the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered. For details on selecting an appropriate timeout period, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779
blank	ROIF is not active.

Auto-answer IP Failure AUX Reason Code

Valid entries	Usage
0 to **99**	Enter the reason code assigned for auto-answer IP failure, as the reason the agent was put into Aux Work.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA).

When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available. To use MAO, Expert Agent Selection (EAS) must be enabled. For more information on MAO, see *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Maximum Agent Occupancy Percentage

Valid entries	Usage
0 to **100**	Enter the percentage for MAO. Default is **100**.

Maximum Agent Occupancy AUX Reason Code

Valid entries	Usage
0 to **99**	Enter a reason code value. Default is **9**. A different reason code can be used for this purpose, but Avaya recommends that you do *not* use reason code 0.

FORCED AGENT LOGOUT PARAMETERS

Maximum Time Agent in ACW before Logout (sec.)

This field is used for setting a maximum time the agent can be in ACW on a per system basis. You can only change the default if **Expert Agent Selection (EAS) enabled?** is set to **y** on the **Feature-Related System Parameters** screen, and the **Call Center Release** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **3.0** or later. When this timer expires, the agent is logged out. This system option applies only to EAS configurations.

Valid entries	Usage
30 to **9999** or blank	Indicate the maximum time an agent can be in ACW before being automatically logged out. Default is blank, meaning no timeout.

ACW Forced Logout Reason Code

This field is used to specify the reason for logging out the agent due to timeout in ACW when the Reason Codes feature is active. You can only change the default if, on the **System Parameters Customer-Options (Optional Features)** screen, **Reason Codes** is set to **y**, and the **Call Center Release** field is set to 3.0 or later. Additionally, the **Expert Agent Selection (EAS) enabled?** field on the **Feature-Related System Parameters** screen must be set to **y**.

Valid entries	Usage
0 to **9**	Enter a reason code value. Default is **0**.

Field descriptions for page 15

Figure 359: Feature-Related System Parameters screen

```
change system-parameters features                          Page 15 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

SPECIAL TONE

                              Special Dial Tone? y
```

Special Dial Tone

Valid entries	Usage
y/n	Enter **y** to enable an audible tone indicating the station is locked. Default is **n**.

Field descriptions for page 16

Figure 360: Feature-Related System Parameters screen

```
change system-parameters features                               Page 16 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                Automatic Exclusion Coverage/Hold? y
            Automatic Exclusion with Whisper Page? y
                              Recall Rotary Digit: 2
                    Password to Change COR by FAC: *
         Duration of Call Timer Display (seconds): 3

WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

IP PARAMETERS
                   Direct IP-IP Audio Connections? n
                             IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                           Re-try?
     T2 (Backward Signal) Activation Timer (secs):
```

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Activates automatic exclusion automatically by class of service when a user goes off-hook on a station with an assigned **Exclusion** button. This works only for stations on the local server running Communication Manager.

Valid entries	Usage
y	Enables automatic exclusion by a class of service.
n	Exclusion operates normally. See Exclusion on Telephone Feature Buttons Table on page 134 for more information.

Automatic Exclusion Coverage/Hold

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Administer a call timer button on the **Station** screen.

Valid entries	Usage
3 to **30**	Enter the length of time (in 3 second increments) that the call information remains on display after the call is terminated.

Password to Change COR by FAC

Appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Change COR by FAC** field is **y**. Avaya recommends using this password option.

Valid entries	Usage
4 to 8 digits	Requires the password option.
blank	Disables the password option.

Recall Rotary Digit

This establishes the digit to use for rotary telephones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary telephones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary digit.

Valid entries	Usage
0 to **9**	Enter the digit users can dial to generate recall dial tone. Use a number that is not the first digit in normal dialing patterns.

WIRELESS PARAMETERS

Radio Controllers with Download Server Permission

Enter the necessary characters for the port location of the circuit pack containing the radio controllers with download server permission.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth characters are the slot number

IP PARAMETERS

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the Avaya S8XXX Server in IP format, without going through the Avaya DEFINITY TDM bus. Default is **n**.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

The **Re-try** field applies to outgoing Russian MFP trunks. It allows the server running Communication Manager to resend Russian MFP calling party number and dialed number information to the CO. The server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

Valid entries	Usage
y/n	Enter **y** to resend address information on outgoing Russian MFP trunks.

T2 (Backward Signal) Activation Timer (secs)

The **T2 (Backward Signal) Activation Timer (secs)** field applies to outgoing Russian MFP trunks. This field sets the number of seconds that Communication Manager waits for confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks

Valid entries	Usage
5 to **20**	Enter the number of seconds the system waits to receive confirmation after sending the address information on outgoing Russian MFP trunks.

Field descriptions for page 17

Figure 361: Feature-Related System Parameters screen

```
change system-parameters features                              page 17 of x
                       FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
Invalid Number Dialed Intercept Treatment: announcement  7700
Invalid Number Dialed Display: Invalid Number
Restricted Number Dialed Intercept Treatment: announcement  7701
Restricted Number Dialed Display: Restricted No.
Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
Whisper Page Tone Given To: all

6400/8400/2420J LINE APPEARANCE LED SETTINGS
                    Station Putting Call On Hold: green   wink
                   Station When Call is Active: green   solid
       Other Stations When Call Is Put On Hold:
            Other Stations When Call Is Active:
                                        Ringing:
                                           Idle:
         Display Information With Bridged Call?
                             Pickup On Transfer?
```

INTERCEPT TREATMENT PARAMETERS

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Invalid Number Dialed Display

This field shows a name in either Latin or Asian characters for an invalid number calling in.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Restricted Number Dialed Intercept Treatment

This field controls whether an announcement or an intercept tone is played when an end-user dials an number restricted from them due to COS, COR, or FRL restrictions. Enter the type of intercept treatment the end-user hears after dialing a restricted number.

Valid entries	Usage
tone	Provides intercept tone when the end-user dials an restricted number. This is the default.
announcement	Provides a recorded announcement when the end-user dials a restricted number. You select and record the message. Enter the extension number for the announcement in the associated field.

Restricted Number Dialed Display

This field controls whether the system displays any string of alphanumeric characters assigned for calls that are denied because of COS/COR, or FRL restrictions.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

WHISPER PAGE

Whisper Page Tone Given To

Use this field to indicate who should hear a Whisper Page.

Valid entries	Usage
all	All parties hear the Whisper Page.
paged	The whisper page feature sends a beep to the paging and the paged party.

DIGITAL STATION LINE APPEARANCE LED SETTINGS

WARNING:
The following fields only change the LED operation for 84xx and 64xx model telephones. When the LED operation is changed using any of these fields, then IP Agent and IP Softphone using a station type of 84xx or 64xx does not work. For station types other than 84xx or 64xx, a change to the LEDs using these fields does not affect either IP Agent or IP Softphone.

Note:
The system generates a warning if the default values of the LED Settings field are changed. The warning message states "WARNING: Avaya Softphone will not operate correctly if this value is changed." You will see this warning message if you are running Avaya Communication Manager 3.1 or later.

Station Putting Call On Hold

Use this field to control the LED color and flash rate on the 8400 and 6400 series telephones for a call held on a Primary or Bridged Appearance. The LED for the color not selected is turned OFF. The default values are **green** and **wink**.

Valid entries	Usage
green or **red**	Indicate whether the LED is green or red.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for a call on hold.

Station When Call is Active

Use this field to control the red LED on the 8400 and 6400 series telephones, for a station active on a call. The default value is **steady**.

Valid entries	Usage
steady	When the value is steady, Communication Manager controls the red LED.
off	When the value is off, the red LED is always OFF.

Other Stations When Call Is Put On Hold

Use this field to control LED options for the other stations with a Bridged Appearance that has been placed on hold (e.g. the user of this station has not pushed the hold button). The default values are **green** and **wink**.

Note:

This field is for a DCP bridged appearance LED color and flash rate when a call on a bridged appearance is put on hold by another party on the DCP bridged appearance. Additionally, this field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values **red** and **flash** for this field.

Valid entries	Usage
green or **red**	Indicate the color of the LED. Default is **green**.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for the LED. Default is **wink**.

Other Stations When Call Is Active

Use this field to control a DCP bridged appearance LED for those non-active parties with a bridged appearance that is active. The default value is **green**.

Note:

This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses ICONs rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the value **red** for this field.

Valid entries	Usage
green or **red**	Select the LED color. Default is **green**.

Ringing

Use this field to control the LED color and flash rate while a call is ringing.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values **red** and **wink** for this field. The default values are **green** and **flash**.

Valid entries	Usage
green or **red**	Indicate the LED color.
off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Indicate the flash rate.

Idle

Use this field to control the LED of a station that is idle. The default value is **steady**.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. This value controls the red LED. Correct operation in the Japanese environment requires the administrator to select the value **off** for this field.

Valid entries	Usage
steady	LED is on. This is the default.
off	LED is off.

Display Information With Bridged Call

Use this field to control whether or not name and number for a bridged call are displayed on the telephone of the called party. A **y** entry indicates that the information is to be displayed; this field does not in any way control the content of the display.

Valid entries	Usage
y/n	Type **y** to display the name and number for an incoming call to the bridged appearance. Default value is **n**.

Pickup on Transfer

Valid entries	Usage
y	Enter **y** to allow bridged appearances of a station to pick up a call on hold because of a transfer.
n	Bridged appearances of another station are NOT allowed to pick up a call on hold because of a transfer.

Firmware Station Download

Use this screen to download firmware to multiple stations of the same telephone type, either 2420 or 2410 DCP telephones. Download firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

Field descriptions for page 1

Figure 362: Firmware Station Download screen

```
 change firmware station-download                                  page 1 of x

                              FIRMWARE STATION DOWNLOAD

Source File:

Schedule Download? y
     Start Date/Time:/:                    Stop Date/Time:/:
Continue Daily Until Completed? y
Download Set Type: 2420

Beginning Station:                Ending Station:
```

Source File

Valid entries	Usage
up to 32 alphanumeric characters	Display only field. This field displays the name of the file specified on the **TFTP Server Configuration** screen, and which exists in system memory.

Schedule Download

Valid entries	Usage
y/n	Enter **y** to schedule a time for firmware download to multiple DCP stations.

Start Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to begin.

Stop Date/Time

This field appears only when the **Schedule Download** field is set to **y**.

Valid entries	Usage
mm, dd, yyyy; hh, mm	Enter the month, day, year, and time at which you want the firmware download to end.

Continue Daily Until Completed

Valid entries	Usage
y/n	Enter **y** if you want the system to execute the firmware download each day at the scheduled time until all specified telephones have received the firmware.

Download Set Type

Valid entries	Usage
2410 DCP **2420 DCP**	Display only. Indicates the set type (2410 or 2420) of DCP telephones to which firmware is to be downloaded.

Beginning Station

Valid entries	Usage
up to 8 digits	Enter the first extension number in the range of telephones to which you want to download the firmware. Up to 1000 stations can be included in a scheduled download.

Ending Station

Valid entries	Usage
up to 8 digits	Enter the last extension number in the range of telephones to which you want to download firmware. Up to 1000 stations can be included in a scheduled download.

Group Paging Using Speakerphone

Use this screen to assign digital speakerphones to a paging group. Users can page all the telephones in the group simultaneously by dialing the group's extension.

Field descriptions for page 1

Figure 363: Group Paging Using Speakerphone screen

```
change group-page 1                                                  Page 1 of x
                          GROUP PAGING USING SPEAKERPHONE

    Group Number: 1                              Group Extension: 1234567890123
      Group Name:                                            COR: 1
GROUP MEMBER ASSIGNMENTS                                      TN: 1
  Extension       Name                     Extension       Name
  1234567890123   12345678901234567890     1234567890123   12345678901234567890

 1: 41752          x41752 4a1823 24port 17:
 2: 41153                               18:
 3: 41750    st2 4a1802                 19:
 4: 41529    Prince Charles             20:
 5: 41527    x41752 port 4a1805         21:
 6: 41706    EXT 41706                  22:
 7: 41534    Thunder x41534             23:
 8:                                     24:
 9:                                     25:
10:                                     26:
11:                                     27:
12:                                     28:
13:                                     29:
14:                                     30:
15:                                     31:
16:                                     32:
```

COR

Valid entries	Usage
0 to **995**	Enter a class of restriction. In order to page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Valid entries	Usage
An extension number	Assign a telephone to the group by entering its extension number in this field.

Group Extension

Valid entries	Usage
An extension number	Assign the extension users will dial to page the members of this group.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a name that's informative to users, because it appears on callers' telephone displays when they page the group. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Group Number

This field displays the identifying number the server running Communication Manager assigns to the group when it is created.

Name

When you save your changes, Avaya Communication Manager fills in this display field with the name assigned to each extension on the **Station** screen.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

TN

This field allows group paging to be partitioned by tenant. Enter the tenant number for this paging group.

Related topics

See Paging Over Speakerphones on page 523 for complete instructions.

See "Group Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Holiday Table

Use this screen to define individual holidays or holiday ranges.

Field descriptions for page 1

Figure 364: Holiday Table screen

```
change holiday table x                                    Page   1 of x
                                     HOLIDAY TABLE

Number:  1                    Name:

______START______      ______END________
Month Day Hour Min     Month Day Hour Min     Description
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
__    __  __   __      __    __  __   __      _____________________________
```

Description

Valid entries	Usage
Up to 27 characters.	Enter a phrase to describe the holiday.

End Day

Valid entries	Usage
1 to **31**	Enter the ending day of the holiday.

End Hour

Valid entries	Usage
0 to **23**	Enter the ending hour of the holiday using a 24-hour clock.

End Min

Valid entries	Usage
0 to **59**	Enter the ending minute of the holiday.

End Month

Valid entries	Usage
1 to **12**	Enter the ending month of the holiday.

Name

Display-only field identifying the name of the table.

Valid entries	Usage
Up to 27 characters	Description of the holiday table.

Number

Display-only field identifying the holiday table number.

Valid entries	Usage
1 to **10**	Holiday table number.

Start Day

Valid entries	Usage
1 to **31**	Enter the starting day of the holiday.

Start Hour

Valid entries	Usage
0 to **23**	Enter the starting hour of the holiday using a 24-hour clock.

Start Min

Valid entries	Usage
0 to **59**	Enter the starting minute of the holiday.

Start Month

Valid entries	Usage
1 to **12**	Enter the starting month of the holiday.

Hospitality

This screen is used to implement the system parameters associated with the hospitality features. To use and administer the Hospitality-related features, **Hospitality** must be **y** on the System Parameters Customer-Options (Optional Features) screen. Contact your Avaya representative for assistance.

Field descriptions for page 1

Figure 365: Hospitality screen

```
change system-parameters hospitality                            Page   1 of  x
                              HOSPITALITY

                       Message Waiting Configuration: act-nopms
               Controlled Restrictions Configuration: act-nopms
               Housekeeper Information Configuration: act-nopms
                     Number of Housekeeper ID Digits: 0
                                    PMS Log Endpoint:
                           Journal/Schedule Endpoint:
             Client Room Coverage Path Configuration: act-nopms
              Default Coverage Path for Client Rooms:
             Forward PMS Messages to Intuity Lodging? n

                              PMS LINK PARAMETERS
                                       PMS Endpoint:
          PMS Protocol Mode: transparent ASCII mode? n
               Seconds before PMS Link Idle Timeout: 20
Milliseconds before PMS Link Acknowledgment Timeout: 500
                   PMS Link Maximum Retransmissions: 3
           PMS Link Maximum Retransmission Requests: 3
                   Take Down Link for Lost Messages? y
```

Client Room Coverage Path Configuration

This indicates whether the server and the Property Management System (PMS) exchange coverage path information for guest stations.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept coverage path information. This field does not apply to normal mode. When upgrading from a release that does not support this feature, the field is set to **act-pms** if the PMS protocol mode is administered for transparent or ASCII mode.

Controlled Restrictions Configuration

This indicates whether controlled restriction information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Default Coverage Path for Client Rooms

This applies only to stations with a "client room" class of service in the "occupied" mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with "client room" class of service.

Valid entries	Usage
1 to **9999** or blank	Enter the coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid entries	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

Housekeeper Information Configuration

This indicates whether housekeeper information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept housekeeper information.

Journal/Schedule Endpoint

This is a valid data extension number that is assigned to the data module connected to the Journal/Schedule printer.

Valid entries	Usage
Valid data extension number	Cannot be a VDN extension. This extension can be the same as the PMS/Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the **IP Services** screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the **IP Services** screen.
blank	

Message Waiting Configuration

This indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	Message waiting is active on the server and information between the PMS and server is being transmitted.

Number of Housekeeper ID Digits

Valid entries	Usage
0 to **6**	Enter the number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

This is a valid data extension number that is assigned to the data module connected to the PMS/Log printer.

Valid entries	Usage
Valid data extension	Cannot be a VDN extension. This extension is dialed by the server to send housekeeping and PMS events to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the **IP Services** screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the **IP Services** screen.
blank	

PMS LINK PARAMETERS

ASCII mode

The **PMS Protocol Mode** field must be **transparent**.

Valid entries	Usage
y/n	Enter **y** when the ASCII-only mode is being used for the PMS message set.

Milliseconds Before PMS Link Acknowledgment Timeout

This regulates how quickly the system responds to a message from the PMS (also known as "pace timing"). This value is also used as the "inquiry message" (ENQ) time-out value. In most cases, keep this as short as possible.

Valid entries	Usage
100 to **1500**	Enter the time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Link Maximum Retransmission Requests

Valid entries	Usage
1 to **5**	Enter the number of times that the server will allow the PMS to request acknowledgment for a message that it sent.

PMS Link Maximum Retransmissions

Valid entries	Usage
1 to **5**	Enter the number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

PMS Log Endpoint

Valid entries	Usage
Valid extension	Enter the data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	Use this value if the PMS is connected over a TCP/IP link, and this link is defined as PMS on the **IP Services** screen.
blank	

PMS Protocol Mode

Valid entries	Usage
normal	Indicate the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.
transparent	

Seconds Before PMS Link Idle Timeout

Valid entries	Usage
5 to **20**	Enter the idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Take Down Link for Lost Messages

Valid entries	Usage
y/n	Enter **y** to cause the PMS link to come down if messages are being lost. Monitor your PMS error log if you use n.

Field descriptions for page 2

Figure 366: Hospitality screen

```
change system-parameters hospitality                           Page   2 of x

                                        HOSPITALITY
           Dual Wakeup? y    Daily Wakeup? y  VIP Wakeup? y
                                 VIP Wakeup Per 5 Minutes: _____
                          Room Activated Wakeup With Tones?
                  Time of Scheduled Wakeup Activity Report: ______
                   Time of Scheduled Wakeup Summary Report: ______
         Time of Scheduled Emergency Access Summary Report: ______
                                        Announcement Type:
                         Integrated Announcement Extension:
        Length of Time To Remain Connected To Announcement: ____
           Extension To Receive Failed Wakeup LWC Messages: ______
          Routing Extension On Unavailable Voice Synthesis: ______
                  Display Room Information in Call Display?
                        Automatic Selection of DID Numbers?
                       Custom Selection of VIP DID Numbers?
                                 Number of Digits from PMS:
                                         PMS Sends Prefix?
                     Number of Digits in PMS Coverage Path:
                                    Digit to Insert/Delete:
```

Announcement Ports

This field appears only when the **Announcement Type** field is **voice-synthesis**. For the **voice-synthesis** announcement type, this indicates the equipment location of two ports on the voice synthesizer circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Announcement Type

This indicates the type of automatic wakeup announcement the hotel guest will receive. Allowable entries are as follows:

Valid entries	Usage
external	Applicable when using an announcement adjunct. If **external** is used, complete the **Auxiliary Board for Announcement** field.
integrated	Applicable when using the TN750B or TN750C announcement circuit pack. If **integrated** is used, complete the **Integrated Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the **Recorded Announcements** screen) or a VDN.
	1 of 2

Valid entries	Usage
mult-integ	Multi-integrated; applicable when using the TN750B or TN750C announcement circuit pack. **mult-integ** allows the automatic wakeup feature to use integrated announcement circuit packs to play any one of multiple announcements to different extensions during a wakeup call. If **mult-integ** is used, complete the **Default Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the **Recorded Announcements** screen) or a VDN.
voice-synthesis	If **voice-synthesis** is used, complete the **Announcement Ports** field.
music-on-hold	If **music-on-hold** is used, no other field appears.
silence	If **silence** is used, no other field appears.
	2 of 2

Automatic Selection of DID Numbers

This field assigns a 2 to 5-digit number to a guest's telephone number that is not associated with the room number.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

Auxiliary Board for Announcement

This field appears only when the **Announcement Type** field is **external**. This indicates the equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Custom Selection of VIP DID Numbers

This field allows you to select the DID number assigned to a room when a guest checks in. This field can be accessed only if the **Automatic Selection of DID Numbers** field is **y**.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Daily Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request daily wakeup calls.

Default Announcement Extension

This field appears only when the **Announcement Type** field is **mult-integ**. This indicates the default wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
valid extension or VDN	Enter the extension of the announcement you want to use for default wakeup calls.

Digit to Insert/Delete

Enter the leading digit that can be deleted and inserted back as described in the following text. The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. In order to accommodate PMS devices that are based on room number and not extension, this leading digit can be deleted on messages from Avaya Communication Manager to the PMS, and then inserted back on messages from the PMS to Communication Manager.

Note:

The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Valid entries	Usage
0 to **9**	Enter the leading digit that can be deleted and inserted back as described in the following text.

Display Room Information in Call Display

This indicates the type of guest room information displayed on telephone displays.

Valid entries	Usage
y	If this field is set to **y**, the telephones will display the name and room number. The extension number and room number are not always the same number.
n	If this field is set to **n**, the telephones will display the name and extension number.

Dual Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request two wakeup calls within one 24-hour time period.

Extension to Receive Failed Wakeup LWC Messages

This indicates where unsuccessful wakeup LWC messages will be stored. This is usually administered to an unassigned extension (cannot be a VDN extension) or to the attendant (attd). In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wakeup calls.

Valid entries	Usage
Unassigned extension	Enter the extension where unsuccessful wakeup LWC messages will be stored.

Integrated Announcement Extension

This field appears only when the **Announcement Type** field is **integrated**. This indicates the wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
Valid extension or VDN	Enter the extension of the announcement you want to use for wakeup calls.

Length of Time to Remain Connected to Announcement

This applies only after the guest has heard the announcement completely one time, but continues to listen.

Valid entries	Usage
0 to **300**	Enter the length of time in seconds that a hotel guest will be connected to an announcement.

Number of Digits from PMS

This indicates the number of digits being sent from the PMS to the server to identify room numbers.

Note:

If the **Number of Digits from PMS** field is blank and the **PMS Sends Prefix** field is set to **n**, the server will not support an extension that starts with **0**.

Valid entries	Usage
1 to **4**	When using normal mode, digits **1** through **4** are valid.
1 to **5**	When using transparent or ASCII mode, digits **1** through **5** are valid.
blank	If using mixed numbering in the server, leave this field blank.

Number of Digits in PMS Coverage Path

This indicates whether the coverage paths are **3** or **4** digits long. There can be up to 9999 coverage paths.

Valid entries	Usage
3 to **4**	Indicate whether the coverage paths are 3 or 4 digits long.

PMS Sends Prefix

This indicates if the PMS sends a prefix digit to the server as part of the room numbering plan.

Note:

If the **PMS Sends Prefix** field is set to **n** and the **Number of Digits from PMS** field is blank, the server will not support an extension that starts with **0**.

Valid entries	Usage
y/n	Enter **y** or **n** to indicate if the PMS sends a prefix digit to the server as part of the room numbering plan.

Room Activated Wakeup with Tones

Valid entries	Usage
y/n	Enter **y** if wakeup calls can be activated via tones that prompt users for the time they wish to waken. This allows room activated wakeup calls without the use of a speech synthesizer or a display telephone.

Routing Extension on Unavailable Voice Synthesis

Valid entries	Usage
Assigned extension (cannot be a VDN extension) or **attd**	A call will be placed to this extension if a voice synthesis port is not available during voice synthesis entry of wakeup requests.

CAUTION:

Set the following reports for a time other than when the system does its scheduled maintenance tests. To make sure the times do not overlap, enter the command **`display system-parameters maintenance`** and check when the system is set to run tests.

Time of Scheduled Emergency Access Summary Report

This indicates the time of day that the Emergency Access Summary Report will be printed on the Journal/ Schedule printer.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Activity Report

This indicates the time of day that the Wakeup Activity Report will be printed on the Journal/ Schedule Printer. This report summarizes the wakeup activity for each extension that had wakeup activity for the past 24 hours.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Summary Report

This indicates the time of day that the Wakeup Summary Report will be printed on the Journal/Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wakeup calls and a list of extensions to which wakeup calls were attempted but did not complete during the hour.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

VIP Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request VIP wakeup calls.

VIP Wakeups Per 5 Minutes

This field appears if the **VIP Wakeup** field is **y**.

Valid entries	Usage
1 to **50**	Enter the number of VIP Wakeup calls allowed in a 5-minute interval.

Field descriptions for page 3

Figure 367: Hospitality screen

```
change system-parameters hospitality                            Page 3 of x

                                   HOSPITALITY

ROOM STATES:

                  Definition for Rooms in State 1: Rooms in State 1
                  Definition for Rooms in State 2: Rooms in State 2
                  Definition for Rooms in State 3: Rooms in State 3
                  Definition for Rooms in State 4: Rooms in State 4
                  Definition for Rooms in State 5: Rooms in State 5
                  Definition for Rooms in State 6: Rooms in State 6

HOSPITALITY FEATURES

                                             Suite Check-in? n
```

ROOM STATES

Definition for Rooms in State 1 through 6

Enter up to a 30-character definition for each room status. For example, you could define state 1 as 'clean, ready to use' and state 2 as 'occupied, needs cleaning.'

The definitions for room states (Field descriptions for page 3), are for Attendant Room Status only. If you are not using Attendant Room Status, you do not need to complete these fields.

HOSPITALITY FEATURES

Suite Check-in

This field allows attendants to have the system automatically check-in several related extensions with one **`check-in`** command.

Valid entries	Usage
y/n	Enter **y** to use the Suite Check-in feature. See "Hospitality" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Hunt Group

Hunt groups allows calls to be answered by users (agents) at a predefined group of telephones or devices.

Use the **Hunt Group** screen to create a hunt group, identified by a hunt group number, and to assign hunt group member users by their extension numbers.This screen can also be used to implement associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing. The total number of pages can vary, depending on system configuration. See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207, for the maximum number of hunt groups supported by each configuration.

When a call comes into a hunt group, the system checks for the busy or idle status of extension numbers in the hunt group when answering. A Uniform Call Distribution (UCD) type hunt group selects the "most idle" extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the "most idle" agent or the "least occupied" agent with the highest skill level for the call's skill.

Note:

Vector controlled splits/skills can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that will terminate the call to a vector controlled split/skill); however, the calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

The **Hunt Group** screen can vary according to system configuration and values populating particular fields.The following figures show several ways that page 1 of the **Hunt Group** screen might appear. The descriptions that follow the figures include all fields shown in all variations of Page 1.

Field descriptions for page 1

Figure 368: Hunt Group screen when Queue is y

```
change hunt-group n                                                  Page 1 of X
                                      HUNT GROUP

           Group Number: 4__                                            ACD? _
             Group Name: ____________________________                 Queue? y
                                                                Queue Limit:____
        Group Extension: ________                                    Vector? _
             Group Type: _______                              Coverage Path: ____
                     TN: ___                       Night Service Destination: ____
                    COR: _                                  MM Early Answer? _
          Security Code: ____                         Local Agent Preference? _
       ISDN Caller Disp: ________


Calls Warning Threshold: ___    Port: x____   Extension: ____
 Time Warning Threshold: ___    Port: x____   Extension: ____
```

Figure 369: Hunt Group screen when Queue and Vector are n

```
change hunt-group n                                                   Page 1 of X
                                      HUNT GROUP

            Group Number: ___                                           ACD? n
              Group Name: ____________________________                Queue? n
         Group Extension: ____                                       Vector? n
              Group Type: ________                            Coverage Path: ____
                      TN: ___                     Night Service Destination: _____
                     COR: _                                 MM Early Answer? _
           Security Code: ____                        Local Agent Preference? _
     ISDN Caller Display: ________
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are **x**.

Figure 370: Hunt Group screen when Queue and Vector are y

```
change hunt-group n                                            Page 1 of X
                                     HUNT GROUP

            Group Number: ___                                         ACD? n
              Group Name: ____________________________              Queue? y
         Group Extension: ________                                 Vector? y
              Group Type: ______
                      TN: ___
                     COR: _                             MM Early Answer?
           Security Code: ____                  Local Agent Preference? _
     ISDN Caller Display: ________


 Calls Warning Threshold: ___   Port: x___   Extension: ____
  Time Warning Threshold: ___   Port: x___   Extension: ____
```

The two **Extension** fields display only when the **Calls Warning Port** and the **Time Warning Port** fields are x.

Figure 371: Hunt Group screen when Queue is y and Vector is n

```
change hunt-group n                                             Page 1 of X
                                     HUNT GROUP

            Group Number: ___                                         ACD? n
              Group Name: ____________________________              Queue? n
         Group Extension: ________                                 Vector? n
              Group Type:                                   Coverage Path: ____
                      TN:                        Night Service Destination: _____
                     COR:                                 MM Early Answer?
           Security Code: ____                    Local Agent Preference? _
        ISDN Caller Disp: ________


Calls Warning Threshold: ___   Port: x___   Extension: ____
 Time Warning Threshold: ___   Port: x___   Extension: ____
```

ACD

Indicates whether Automatic Call distribution is used. This field cannot be set to **y** if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ACD** field is **n**.

Valid entries	Usage
y	The hunt group will function as an ACD split/skill. AUDIX hunt groups can function as ACD splits/skills.
n	This feature is not desired, even if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ACD** field is **y**. When the hunt group is assigned as an ACD split/skill, the hunt group members serve as ACD agents. The agents in this split/skill must log in to receive ACD split/skill calls. If this hunt group is on a remote server running Communication Manager and using the AUDIX in a DCS feature, then enter **n**.

Calls Warning Threshold

Appears if the **Queue** field is **y**. Enter the number of calls that can be queued before the System flashes the queue status (feature buttons assigned on agents telephones) and the optional Auxiliary Queue Call Warning Threshold lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and the threshold has not yet been reached

Valid entries	Usage
1 to **999** and must be less than or equal to the queue length or blank	This field must not be left blank if **Calls Warning Port** is assigned a port number.

(Calls Warning) Extension

Appears if the **Queue** field is **y** and when the **Calls Warning Port** and the **Time Warning Port** fields are **x**. An extension is needed when an X is placed in **Calls Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Calls Warning Port** is assigned a valid port (either via TTI or the `change hunt-group` command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field cannot be blank.

(Calls Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to connect the optional external Auxiliary Queue Call Warning Threshold lamp that will flash when the number of calls in queue has exceeded the queue warning threshold (assigned in **Calls Warning Threshold**).

This port is assigned to an Analog Line circuit pack or given an **x** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

Note:
For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

COR

Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group. If this is a hunt group supporting the AUDIX in a DCS feature, the CORs on the **Hunt Group** screen on each server running Communication Manager must be the same.

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group.

Coverage Path

Enter a coverage path number. This assigns a coverage path for the hunt group. The coverage path is assigned using the **Coverage Path** screen. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
1 to **999**	Enter a coverage path number.
t1 to **t999**	Time of day table
blank	

Group Extension

Enter an unused extension number to be assigned to the hunt group. The field cannot be blank.

Valid entries	Usage
1 to 7 digits	Unassigned extension

Group Name

This field identifies the hunt group.

Valid entries	Usage
27-character string	Enter a character string that uniquely identifies the group (for example, "parts dept," "purchasing," or "sales dept"). **Note:** The **Group Name** field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones. For more information on Unicode language display, see Administering Unicode display on page 203. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Group Number

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

The group types available depend on what is administered on your System Parameters Customer-Options (Optional Features) screen for **Automatic Call Distribution (ACD)**, **Expert Agent Selection (EAS)** and **Business Advocate**. The table below shows what group types are available depending on your configuration.

Each option uses a different method to select an extension or agent for a call when two or more extensions or agents are available. The second table shows how calls are handled for each group type.

	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	slm
ACD=n	x	x						
ACD, Split, Vector = n/y		x	x					
ACD, Skill, Vector = n/y			x	x				x
ACD, Skill, Vector = y Advocate or Elite			x	x	x	x		
ACD, Skill, Vector = y Dynamic Advocate			x	x	x	x	x	

Valid entries	Usage
circ	Enter **circ** (circular) when the call should be routed in a "round-robin" order. The order in which you administer the extensions determines the order that calls are directed. The server running Communication Manager keeps track of the last extension in the hunt group to which a call was connected. The next call to the hunt group is offered to the next extension in the circular list, independent of how long that extension has been idle. You cannot use circular hunting with automatic call distribution, queues, or vectors.
ddc	Enter **ddc** when the call should be routed to the first extension or ACD agent assigned in the ACD split. Group type ddc is also known as "hot seat" distribution. "ddc" distribution is not available when the group is administered as a skill.
ucd-mia **ucd-loa**	When **ucd-mia** or **ucd-loa** is entered, a call routes to the most-idle agent based on when the agent finished the most recent call (**ucd-mia**), or the least occupied agent based on agent occupancy (**ucd-loa**). Enter **ucd-mia** or **ucd-loa** if the hunt group has an AUDIX message. One of these entries is required when supporting the Outbound Call Management feature and when the **Controlling Adjunct** field is **asai**.
ead-mia **ead-loa**	When **ead-mia** or **ead-loa** is entered, a call routes to the available agent with the highest skill level for the call. If two or more agents with equal skill levels are available, Communication Manager routes the call to the most-idle agent based on when the agent finished the most recent call ("ead-mia"), or the least occupied agent based on agent occupancy ("ead-loa"). This allows a call to be distributed to an agent best able to handle it if multiple agents are available.
pad	Enter **pad** (percent allocation distribution) to select an agent from a group of available agents based on a comparison of the agent's work time in the skill and the agent's target allocation for the skill.
slm	Enter **slm** when you want to: ● Compare the current service level for each SLM-administered skill to a user-defined call service level target and identify the skills that are most in need of agent resources to meet their target service level. ● Identify available agents and assess their overall opportunity cost, and select only those agents whose other skills have the least need for their service at the current time.

ISDN Caller Disp

This field is required if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**.

Valid entries	Usage
grp-name **mbr-name**	Enter **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, will be sent to the originating user. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.
blank	If the **ISDN-PRI** or the **ISDN-BRI Trunks** field is **n**, this field must be blank.

Local Agent Preference

Use this field whether the routing of an incoming ACD call to an idle agent should be done by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. This field can only be set to **y** if:

- the **Call Center Release** field on the **System Parameters Customer-Options (Optional Features)** screen is set to 3.0 or later
- the **Expert Agent Selection** and the **Multiple Locations** fields on the **System Parameters Customer-Options (Optional Features)** screen are **y**
- the hunt group is defined as a skill hunt group (the **Skill?** field on page 2 of the **Hunt Group** screen is set to **y**)

Valid entries	Usage
y/n	Enter **y** to indicate that an incoming ACD call to an idle agent should be routed by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. Default is **n**.

MM Early Answer

This field applies for systems using Multimedia Call Handling only.

Valid entries	Usage
y/n	The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue.

Night Service Destination

Enter the destination where calls to this split will redirect when the split is in the night service mode. Not all features will work correctly if this is not a local extension. Does not appear if the **Vector** field is **y**.

Valid entries	Usage
An assigned extension number (can be a VDN extension)	Enter the destination where calls to this split will redirect when the split is in the night service mode.
attd	An attendant group code.
blank	

Queue

Specifies a queue for the hunt group.

Valid entries	Usage
y/n	Enter **y** so the hunt group will be served by a queue.

Queue Limit

This field appears when the **Queue** field is set to **y**.

Valid entries	Usage
1 to **999**	Enter a limit to the number of calls that will queue.
unlimited	The system dynamically allocates the queue slots from a common pool on an as-needed basis.

Security Code

Enter a security code (password) used for the Demand Print feature.

Valid entries	Usage
3 to 8 digits	Enter the password for the Demand Print feature.

Time Warning Threshold

Appears if the **Queue** field is **y** and when the **Calling Warning Port** and the **Time Warning Port** fields are **x**. Enter the time in seconds that a call can remain in the queue before the system flashes the Queue status lamps (feature buttons assigned members telephones) and the Auxiliary Queue Time Warning lamp assigned to this split/skill.

Valid entries	Usage
0 to **999** or blank	An entry of **0** provides a warning whenever a call is queued.

(Time Warning) Extension

Appears if the **Queue** field is **y**. An extension is needed when an **x** is placed in **Time Warning Port**. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Time Warning Port** is assigned a valid port (either via TTI or the `change hunt-group` command), then the extension is removed and considered unassigned.

Valid entries	Usage
Extension	Enter an unassigned extension. This field cannot be blank.

(Time Warning) Port

Appears if the **Queue** field is **y**. Enter the seven-character port number assigned to the Auxiliary Queue Time Warning lamp that flashes when the time entered in **Time Warning Threshold** has been reached by a call in queue.

Note:

This port is assigned to an Analog Line circuit pack or given an **X** designation if an extension is used. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Vector

See example screens for fields that display when this field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this hunt group will be vector controlled. On the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring-Basic** field must be **y** before **y** can be entered here.

Field descriptions for page 2

Page 2 of the **Hunt Group** screen can vary according to values for particular fields on Page 1. The screen shown in Figure 372 appears only when the **ACD** field on page 1 is **y**, and the **Queue** and **Vector** fields are **n**. If the **ACD** field is **n**, the **Message Center** page shown in Figure 374 becomes Page 2 and all subsequent page numbers are decreased by one. The **Timed ACW Interval** field shown below appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Timed ACW** field is **y**. The screen shown in Figure 373 appears when the **Queue** and **Vector** fields are **y**.

The descriptions that follow the figures include all fields in both variations of Page 2.

Figure 372: Hunt Group screen when ACD is y and Queue and Vector are n

```
change hunt-group n                                                 Page 2 of X
                                  HUNT GROUP
                  Skill? _        Expected Call Handling Time (sec): ___
                    AAS? _          Service Level Target (% in sec):80 in 20
               Measured: ____
    Supervisor Extension: ____
   Priority on Intraflow? _

     Controlling Adjunct: ____




Timed ACW Interval (sec): ___              Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                         Redirect on No Answer (rings): ___
                                                       Redirect to VDN: _____
                       Forced Entry of Stroke Counts or Call Work Codes? _
```

Figure 373: Hunt Group screen when Queue and Vector are y

```
change hunt-group n                       P                             Page 2 of X
                                  HUNT GROU
                   Skill?           Expected Call Handling Time (sec):___
                     AAS? _           Service Level Target (% in sec):80 in 20
                Measured: internal
     Supervisor Extension: ____



      Controlling Adjunct: ____



        VuStats Objective: ____
Timed ACW Interval (sec): ___ ___                Maximum Auto Reserve Agents: 0
  Multiple Call Handling: _____________
                                           Redirect on No Answer (rings): ___
                                                         Redirect to VDN: _____
                         Forced Entry of Stroke Counts or Call Work Codes? _
```

AAS

Appears when the **ACD** field is **y**

.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to serve as an Auto-Available Split.

Acceptable Service Level (sec)

Appears if the **ACD** field is **y** and the **Measured** field is **internal** or **both**. This allows BCMS and/or VuStats to report a percentage of calls that were answered within the specified time. This entry is also used by the Business Advocate Service Objective feature.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this hunt group should be answered.

Adjunct CTI Link

Appears when the **ACD** field is **y** and the **Controlling Adjunct** field is **asai** or **adjlk**. Enter the appropriate ASAI CTI Link. This field cannot be blank.

Controlling Adjunct

Appears only if the **ACD** field is **y**. If the controlling adjunct is a CONVERSANT voice system (requires an ASAI link), then enter **asai** in this field. (On the **System Parameters Customer-Options (Optional Features)** screen, the **ASAI Link Core Capabilities** and **Computer Telephony Adjunct Links** fields must be **y** for CallVisor ASAI capability and for an entry other than **none**.)

Valid entries	Usage
none	Indicates that members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can only use their data terminal keyboards to perform telephone functions (for example, change work state).
adjlk	Computer Telephony Adjunct Links
	1 of 2

Valid entries	Usage
asai-ip	Indicates ASAI links administered without hardware.
adj-ip	Indicates ASAI adjunct links administered without hardware.
	2 of 2

Dynamic Percentage Adjustment

Appears when **ACD** and **Group Type** fields on the **Hunt Group** screen are **pad** and the **Business Advocate** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to agents' target allocations as needed to help meet the administered service level targets.

Dynamic Queue Position

Appears when the **ACD** and **Skill** fields are **y** on the **Hunt Group** screen and the **Business Advocate** field is **y** on the **Feature-Related System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to apply the dynamic queue operation to the calls queued to the skill.

Dynamic Threshold Adjustment

Appears when the **ACD** and **Service Level Supervisor** fields on the **Hunt Group** screen are **y** and the **Business Advocate** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to enable automatic adjustments to overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels.

Expected Call Handling Time (sec)

Appears if, on the **System Parameters Customer-Options (Optional Features)** screen, either the **Vectoring (Advanced Routing)** or **Business Advocate** field is **y**. and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999** in increments of 1	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the Business Advocate Percent Allocation feature.

Forced Entry of Stroke Counts or Call Work Codes

Appears when the **ACD** field is **y** and **Controlling Adjunct** field is **none**.

Valid entries	Usage
y/n	Enter **y** so either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode.

Inflow Threshold (sec)

Appears only when the **ACD** and **Queue** fields are **y** and **Vector** field is **n**. Enter the number of seconds that a call can remain in the queue before no more calls will be accepted by the queue. If **0** is entered, a call is redirected to this split/skill only if there is an available agent.

Valid entries	Usage
0 to **999**	Enter the number of seconds that a call can remain in the queue before no more calls will be accepted by the queue.

Level 1 Threshold (sec)

Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 45 calls whose EWT exceeds 45 seconds will be classified as over threshold 1. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Level 2 Threshold (sec)

Appears if the **ACD** field is **y**. Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 60 calls whose EWT exceeds 60 seconds will be classified as over threshold 2. This field is used with Service Level Supervisor and only appears if the **Service Level Supervisor** field is **y**.

Maximum Auto Reserve Agents

Appears only if the **Group Type** field is **slm**. Set the maximum number of Auto Reserve Agents you want to be available for this skill (hunt group). Any time an auto-reserve skill is in danger of falling below its target service level percent, some of this skill's agents are auto-reserved (kept idle in other skills) so they will be available when a new call arrives for this skill. Valid values are **0** to **9**. Default is **0**.

Measured

Provides measurement data for the ACD split/skill collected (internal to the switch) for **VuStats** or **BCMS**. This measurement data is collected for **VuStats** and **BCMS** only if, on the **System Parameters Customer-Options (Optional Features)** screen, they are **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
internal	If you enter **internal** in this field and on the **System Parameters Customer-Options (Optional Features)** screen neither the **VuStats** or **BCMS** field is **y**, the system displays the following message: `<value> cannot be used; assign either BCMS or VuStats first` Contact your Avaya representative to assist with any changes you want to make on the **System Parameters Customer-Options (Optional Features)** screen.
external	Provides measurements made by the Call Management System (external to the server running Communication Manager).
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Multiple Call Handling

Appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Multiple Call Handling** field is **y** and the **ACD** field on this screen is **y**. This field defines whether the hunt group can have multiple call handling capabilities, and if so, what type.

Valid entries	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the telephone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/ skill or direct agent call then is routed to the agent.
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/ Manual-In (MI/AI) work mode, and un unrestricted line appearance is available. Valid in an EAS environment and only when the **Skill** field is **y**.

Priority On Intraflow

Appears if the **ACD** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
y/n	Enter **y** for calls intraflowing from this split to a covering split to be given priority over other calls waiting in the covering split queue.

Redirect on No Answer (rings)

Appears if the **ACD** field is **y**.

Valid entries	Usage
1 to **20**	Enter the maximum number of rings before a call will redirect back to the split/skill, or to the administered VDN.
blank	Deactivates Redirect on No Answer.

Redirect to VDN

Appears if the **ACD** field is **y**. To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that will in turn route to an off-premises VDN. You cannot enter an extension in this field if the **Redirection on No Answer (rings)** field is blank. Direct Agent calls go to the agent's coverage path if it is administered. If not, the calls go to a VDN.

Valid entries	Usage
Assigned VDN or blank	To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN.

Service Level Interval

This field displays only if **Actual** is assigned as the SLM algorithm on the **Feature-Related System Parameters** screen, and **Group Type** on the **Hunt Group** screen is set to **slm**. The interval can be set to the same interval used when specifying the target objectives for the application.

Valid entries	Usage
hourly	ASL algorithm calculations for accepted call and total call components are set to 0 at hourly intervals.
daily	ASL algorithm calculations for accepted call and total call components are set to 0 at daily intervals. This is the default.
weekly	ASL algorithm calculations for accepted call and total call components are set to 0 at weekly intervals. The weekly interval starts as 00:00 hours on Sunday.

Service Level Supervisor

Appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Business Advocate** field is **y** and, on the **Hunt Group** screen, the **ACD** and **Skill** fields are **y**. For information on Business Advocate, please contact your Avaya representative, or see the *Avaya Business Advocate User Guide*, 07-300653.

Valid entries	Usage
y/n	Enter **y** to use Service Level Supervisor for this skill.

Service Level Target (% in sec)

Appears when the **ACD** field and the **Dynamic Percentage Adjustment** or **Dynamic Threshold Adjustment** field on the **Hunt Group** screen is **y** and the **Business Advocate** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. Also appears when the **Group Type** field on the **Hunt Group** screen is **slm**, and on the **System Parameters Customer-Options (Optional Features)** screen, the **Service Level Maximizer** field is set to **y**, and the **Business Advocate** field is set to **n**.

Valid entries	Usage
1 to **99** (percentage) **1** to **9999** (time in seconds)	Enter the percentage and time components of the service level target. After 20 seconds, default service level target values are set at 80%.

Service Objective

Appears when the **Skill** and **Business Advocate** fields on the **Feature-Related System Parameters** screen are **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
1 to **9999**	Enter the per-skill service objective.

Skill

Appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Expert Agent Selection** field is **y** and, on the **Hunt Group** screen, the **ACD** field is **y**.

If this field is **y**, then the **Group Type** field must be **ucd** or **ead**.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to be an EAS skill.

SLM Count Abandoned Calls

Appears only if **Actual** is assigned as the SLM algorithm on the **Feature-Related System Parameters** screen, and **Group Type** on the **Hunt Group** screen is **slm**.

Valid entries	Usage
y	Abandoned calls are included in the ASL algorithm calculations for SLM.
n	Abandoned calls are *not* included in the ASL algorithm calculations for SLM. Use this option when reporting for this application does not account for calls that are abandoned while in skill queues.

Supervisor Extension

Appears if the **ACD** field is **y**.

Valid entries	Usage
Valid extension	Enter the extension number (cannot be a VDN number) of the ACD split/ skill supervisor that agents will reach when using the Assist feature

Timed ACW Interval (sec)

When a value is entered in this field, an agent in auto-in work mode who receives an ACD call from this hunt group is placed automatically into After Call Work (ACW) when the call drops. Enter the number of seconds the agent should remain in ACW following the call. When the administered time is over, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split. The **Timed ACW Interval** field appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Timed ACW** field is **y**, and, on the **Hunt Group** screen, the **ACD** field is **y**.

Note:

This field can be overridden by the settings on the VDN Timed ACW Interval field on the **Vector Directory Number** screen. Coordinate the settings for both fields in setting up delays.

Valid entries	Usage
1 to **9999** or blank	The number of seconds the agent should remain in ACW following the call.

VuStats Objective

Enter a numerical user-defined objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The objective appears on the VuStats display and allows agents and supervisors to compare the current performance against the value of the objective for the split or skill.

You can use this value in a customized VuStats display format if, on the **VuStats display format** screen, the **Object Type** field is either **agent**, **agent-extension**, or **split**.

This field appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **VuStats** field is **y** and the **Measured** field is either **internal** or **both** and, on the **Hunt Group** screen, the **ACD** field is **y**.

Valid entries	Usage
0 to **99999**	Enter a split or skill objective.

Field descriptions for Message Center page

The **Hunt Group** screen pages and fields can vary according to system configuration and values populating particular fields. The figure below is only an example, and is intended to show most of the fields that might appear on this page of the **Hunt Group** screen. This example might not show all fields, or might show fields that normally do not appear together. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and to include information on all fields that might appear.

Figure 374: Hunt Group Message Center screen

```
                                                            Page 2 of X
                              HUNT GROUP

                          Message Center: ________
                       Voice Mail Number: _________________
Routing Digits (e.g. AAR/ARS Access Code): ____
              TSC per MWI Interrogation? n
                   Send Reroute Request? y          Provide Ringback?
                        AUDIX Extension: ________
              Message Center AUDIX Name: ______
                               Primary? _
    Calling Party Number to INTUITY AUDIX? _
                          LWC Reception: _______
                             AUDIX Name: _______
                  Messaging Server Name: _______

           First Announcement Extension: 1234 ___ Delay (sec): 20
          Second Announcement Extension: ________ Delay (sec): __ Recurring? _
```

AUDIX Name

Enter the name of the AUDIX machine as it appears on the **IP Node Names** screen. Add the AUDIX name to the **IP Node Names** screen before entering it in this field. For more information on the **IP Node Names** screen, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Calling Party Number to INTUITY AUDIX

Appears when the **Message Center** field is **audix** or **rem-vm**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

First Announcement Delay (sec)

Appears only if the **Queue** field is **y** and the **Vector** field is **n**. Enter the number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call hasn't been answered after the announcement, the caller hears music (for first announcement only) if Music-on-Hold is provided or ringing for as long as the call remains in queue. Appears only if the **Queue** field is **y** and the **Vector** field is **n**.

Valid entries	Usage
0 to **99**	When **0** is entered, the first announcement is provided immediately to the caller. This value is set automatically to **0** if there is no queue.
blank	This field must be blank if there is no first announcement.

First Announcement Extension

Appears when the **ACD** and **Queue** fields are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Enter a recorded announcement extension number.	This is the announcement the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music (only after the first announcement) if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement; otherwise, the caller hears music (if provided).
blank	Leaving this field blank indicates there will be no announcement.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The **Audix Name** field must be filled in too.
msa	Messaging Server Adjunct
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the system can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, one as **rem-audix**, and as many as six as **qsig-mwi**.

Valid entries	Usage
audix	For AUDIX located on this server running Avaya Communication Manager
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the **ISDN Feature Plus** field on the System Parameters Customer-Options (Optional Features) screen is **y**.
msa	Messaging Server Adjunct
msa-vm	For voice-mail system integrated using Mode Codes or Digital Station Emulation
none	Indicates the hunt group does not serve as a message hunt group.
rem-vm	DCS feature allowing voice mail to be located on another server
qsig-mwi	QSIG network allowing voice mail to be located on another server
sip-adjunct	SIP message center server.

Message Center AUDIX Name

Enter the name of the Message Center AUDIX. Appears on Avaya S8300/S87XX Servers if the **Message Center** field is **audix** or **rem-vm**.

Message Center MSA Name

Note:
Administer the **IP Node Names** screen first.

Enter the name of the Message Center MSA. When it appears, it replaces the **Message Center AUDIX Name** field. Appears on S8300/S87XX Servers if the **Message Center** field is **msa**.

Primary

Appears on Avaya S8300/S87XX Servers if the **Message Center** field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to indicate that the specified AUDIX is the primary adjunct.

Provide Ringback

Appears only if **Message Center** on the **Hunt Group** screen is **fp-mwi** or **qsig-mwi**. Use this field if you are using an SBS trunk for the QSIG MWI hunt group. If set to **y**, a call covering to the message center provides ringback to the caller during the coverage interval.

Valid entries	Usage
y/n	When set to **y**, ringback is provided to the calling party until a Connect is received for the call to the Messaging system. Ringback is discontinued upon receipt of the Connect indication. Default is **n**.

Routing Digits (e.g. AAR/ARS Access Code)

Appears only if **Message Center** is **qsig-mwi** or **fp-mwi**. Shows the AAR (most likely for a Message Center type of **qsig-mwi**) or ARS (most likely for a Message Center type of **fp-mwi**) access code which when prepended to the **AUDIX Complete Number** field defines a route to the Message Center switch hunt group containing the line ports to the AUDIX.

Valid entries	Usage
0 to **9**, *****, or **#**	Enter 1 to 4 digits.

Second Announcement Extension

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Note:

If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

Valid entries	Usage
Valid extension	Enter the extension number assigned to a recorded announcement.
blank	Leaving the field blank indicates there is no second announcement.

Second Announcement Delay (sec)

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**. Enter the time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated.

Valid entries	Usage
1 to **99**	Avaya recommends that, if this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	Leave blank if there is no second announcement.

Second Announcement Recurring

Appears only when the **ACD** and **Queue** fields both are **y** and the **Vector** field is **n**.

Valid entries	Usage
y	The second announcement can be repeated.
blank	Leave blank if there is no second announcement.

Send Reroute Request

Appears only when the **Message Center** field is type **QSIG-MWI** and **Supplementary Services with Rerouting** is **y** on the System Parameters Customer-Options (Optional Features) screen.

Valid entries	Usage
y	Rerouting is invoked. This is the default.
n	Rerouting is not invoked when a call covers through a qsig-mwi hunt group.

TSC per MWI Interrogation

Appears when the **Message Center** field is type **QSIG-MWI**. Use this field to control Temporary Signaling Connections (TSCs) used for message waiting interrogations for users that are “local” to the system in which the hunt group is administered.

Valid entries	Usage
y	Communication Manager brings the TSC up, executes the Interrogate operation, and then tears the TSC down.
n	Communication Manager utilizes the existing TSC sending FACILITY messages to request MWI status if the TSC is already set up, or sets up a TSC, and when the interrogation operation is complete, leaves the TSC up, subject to the existing timer. This is the default.

Voice Mail Extension

Appears if the **Message Center** field is **rem-vm**.

Valid entries	Usage
extension	Enter the UDP extension of the voice-mail hunt group on the host server running Communication Manager.

Voice Mail Handle

This field indicates the SIP Enablement Services (SES) handle that can receive voice mail. This field can be left blank if you supply a Voice Mail Number.

Voice Mail Number

Appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Basic Call Setup** and **Basic Supplementary Services** fields are **y** and the **Message Center** field on this screen is **qsig-mwi** or **fp-mwi**. The **qsig-mwi** selection shows the complete number of the AUDIX hunt group on the Message Center server for QSIG MWI. The **fp-mwi** selection shows the public network number of the AUDIX hunt group on the Message Center server.

Valid entries	Usage
Up to 17 digits	Enter the complete AUDIX dial-up number.

Field descriptions for pages 4 through X

Figure 375: Hunt Group Group Member Assignments screen

```
add hunt-group next                                                   Page x of x
                                 HUNT GROUP

  Group Number: 3        Group Extension: 1234567890123        Group Type:
ucd-mia

   Member Range Allowed: 1 - 1500          Administered Members (min/max): 0 /0
                                               Total Administered Members: 0

GROUP MEMBER ASSIGNMENTS
      Extension     Name                           Extension     Name
      1234567890123 1234567890123456789            1234567890123 1234567890123456789
1457:                                   1470:
1458:                                   1471:
1459:                                   1472:
1460:                                   1473:
1461:                                   1474:
1462:                                   1475:
1463:                                   1476:
1464:                                   1477:
1465:                                   1478:
1466:                                   1479:
1467:                                   1480:
1468:                                   1481:
1469:                                   1482:
```

Administered Members (min/max)

Appears on all member pages. Indicates the minimum and maximum member number administered for this hunt group.

At End of Member List

This display-only field shows the current page is also the last page.

Group Extension

This display-only field shows the extension of the hunt group.

Group Number

This display-only field shows the number of a hunt group.

Group Type

This display-only field shows the type of the hunt group.

Member Range Allowed

The range of allowed members displays on all member pages. These values vary depending on the particular system and/or configuration.

More Members Exist

This display-only field shows there are more members than currently displayed (the current page is not the last page).

Total Administered Members

Appears on all member pages. Indicates the total number of members administered for this hunt group.

GROUP MEMBER ASSIGNMENTS

Ext

A display-only field if the **Controlling Adjunct** field is **asai**. Controlled Agent extensions must be entered on the **Adjunct Controlled Agent Table** screen. The extension cannot be a VDN. The data module cannot be a member of an ACD split/skill. Use this field when the **Controlling Adjunct** field is **none**.

Valid entries	Usage
Valid extension.	Enter the assigned station or attendant console extension.

Name

This display-only field shows the name assigned to the above extension number when it is administered in the system.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Incoming Call Handling Treatment

Use the **Incoming Call Handling Treatment** screen to specify call handling for ISDN and SIP Enablement Services (SES) trunk groups. For more information on ISDN trunk groups, see ISDN Trunk Group.

Note:

This page does not appear if, on the **ISDN Trunk Group** screen, the **Digit Handling (in/out)** field is **overlap** on the "in" side or if the **Direction** field is **outgoing**.

With the **Incoming Call Handling Treatment** screen, you can specify provide unique call treatment for different incoming calls on any ISDN trunk group. The number of unique treatments that can be specified in this table and the number of pages vary depending on whether the Usage Allocation Enhancements feature is enabled and on the amount of available system memory.

Unique digit manipulation capabilities, CPN/BN requests, and night service destinations are possible for different types of incoming calls. The unique digit manipulation capabilities can be particularly useful to accommodate different dial plans for different services on an ISDN trunk type with a **Service Type** (field entry) of **cbc** (Call-by-Call). The table can also be used for ISDN trunk groups that are not Call-by-Call. For example, an ISDN group with **Service Type** set to **mega800** could use the Incoming Call Handling Treatment table to distinguish treatment of calls to different 800 numbers based on the Dialed Number Identification Service (DNIS) number that is incoming to Avaya Communication Manager.

Each row in the table consists of seven columns. The first three columns (**Service/Feature**, **Called Len**, and **Called Number**) constitute a key that together select which row or unique treatment should apply for an incoming call on the group. The remaining four columns (**Del**, **Insert**, and so on) specify the treatment to be provided for a call that matches the key.

If an incoming call is for a service listed in a row on the table, then that row can specify the treatment for the call, depending on the other two columns of the key. The **Called Len** field is used to continue the row determination. If the number of digits received with the incoming call matches the number of digits in the **Called Len** field for calls to the matched service, then this row might apply. If no other row also contains a matching service and called length, then this row does apply. If another row does exist with the same service and number length, then the **Called Number** field will be used to continue the row determination.

If the leading digits received with the incoming call match the digits specified in the **Called Number** field, then this row applies to the call. Therefore, with this table, a unique treatment can be given to any incoming call, even if these calls are to the same service or have the same length of digits. The remaining four fields specify the unique treatment for the call once the row has been determined. Together, the **Del** and **Insert** fields can be used to manipulate the incoming number that will be used to route the call. The **Per Call CPN/BN** field appears only for ISDN trunk groups, and can be used to request CPN/BN from AT&T networks for specific calls incoming on the group. The **Night Serv** field also appears only for ISDN trunk groups, and is used to have calls of different types routed to different night destinations when night service is in effect.

The **Incoming Call Handling Treatment Table** always automatically rearranges to show the precedence order the system uses to pick an entry. Thus, you can easily predict the behavior of the **Incoming Call Handling Treatment Table** by noting the order in which the entries display. (The entries rearrange after submitting the **Trunk Group** screen. A subsequent **`change trunk-group`** or **`display trunk-group`** command then shows the rearranged order.)

Avaya Communication Manager traverses the table from top to bottom and picks the first entry that matches all the following criteria:

- The Service /Feature, if applicable, matches
- The Called/Length matches
- The Called Number matches

If the administered **Called Length** or **Called Number** is blank, that criterion is considered successful.

Incoming Call Handling Treatment Table entries with a predefined service/feature always appear before entries with a user-defined service/feature. To control the order in which certain entries appear, you must use user-defined services/features for those entries. For example, you can redefine the predefined mega800 service/feature as an identical user-defined entry with the name m800.

User-defined entries are always listed in the reverse order compared to the way they appear on the **Network Facilities** screen. Thus, given two user-defined services/features ABC and XYZ, you can force XYZ to appear before ABC in an **Incoming Call Handling Treatment Table** by putting **XYZ** after **ABC** on the **Network Facilities** screen.

Note:

DCS features that use the **remote-tgs** button (on the remote server/switch) do not work when the local trunk group deletes or inserts digits on the incoming call. These buttons try to dial a local TAC. Adding or deleting digits defeats this operation and renders the remote feature inoperable. If digit manipulation is needed, use it on the outgoing side, based on the routing pattern. One reason for digit manipulation is insertion of the AAR feature access code (FAC).

These fields are located on the **Incoming Call Handling Treatment Table** screen.

Field descriptions for page 1

Figure 376: Incoming Call Handling Treatment screen (ISDN)

```
change inc-call-handling-trmt trunk-group 1                        Page 1 of x
                      INCOMING CALL HANDLING TREATMENT

Service/      Called Called          Del Insert           Per Call Night
Feature        Len   Number                               CPN/BN   Service
123456789012345 12 1234567890123456 123 1234567890123456 12345678 1234567890123
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
tie
```

Note:

For SIP Enablement Services (SES) trunk groups, the **Per Call CPN/BN** and **Night Serv** fields do not appear, as shown in Figure 377.

Figure 377: Incoming Call Handling Treatment screen (SIP)

```
change inc-call-handling-trmt trunk-group1                Page 1 of X

                        INCOMING CALL HANDLING TREATMENT

     Service/   Called     Called         Del      Insert
     Feature     Len       Number
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
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   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
   _____________  ___  _______________  ___  ______________
```

Called Len

Specifies the number of digits received for an incoming call. A blank entry may be used only for the situation in which the Called Number field has been set to blank. When used with the blank entry, this means that any length of digits associated with the Called Party IE of the Incoming SETUP message will match this field. The use of the **0** entry is encouraged for the situation in which the PSTN provider does not provide any 'Number Digits' within the received Called Party IE (such as in Japan). Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are up to 16 digits, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type can be administered to be routed to a single destination by deleting all incoming digits and then administering the **Insert** field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, **#**, or leave blank.

Per Call CPN/BN

This field appears only for ISDN trunk groups. Specifies when and how to request Calling Party Number (CPN) or Billing Number (BN) for calls of this type. Leave blank when connected to another Avaya S8XXX Server, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that does not permit customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles "CPN/BN to Terminating End on a Per-Call Basis" and "ANI (BN) on Request." An entry of **none** indicates Communication Manager will not request either CPN or BN for any incoming calls of this type. Valid entries are **cpn-only**, **bn-only**, **bn-pref** (prefer BN, but accepts CPN), **cpn-pref** (prefer CPN, but accepts BN), **none**, or leave blank. Leave blank when connected to another server/switch or to a network other than the AT&T Switched Network.

Note:

A 4-second delay occurs in terminating the call to the far-end station if the connecting server or switch does not respond to the request.

Night Serv

This field appears only for ISDN trunk groups. Specifies a night service extension (can be a VDN extension) per Service/Feature. An entry other than blank overrides **Night Service** entry on page 1 of the screen. This entry can be overridden by the **Trunk/Member Night Service** entry when provided. Valid entries are an assigned extension, the attendant group access code (**attd**), or leave blank.

Service/Feature

This field is display-only. It is auto-populated with the value entered in the **Service Type** field on the **Trunk Group** screen.

Note:

An exception occurs when **cbc** is the value in the **Service Type** field on the **Trunk Group** screen. Because there are several possible values for the **Service/ Feature** field for cbc trunk groups, the field is not display-only, but is available for user entry. Valid **Service/Feature** values for cbc trunk groups can be viewed on the **Network Facilities** screen.

Note also that in addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature), **1** (service), or **2** (incoming) on the **Network Facilities** screen is allowed. For a Service/Feature defined as Type 2, it is this screen that determines which incoming calls are assigned to this Service/ Feature. See the description of the **Network Facilities** screen for details.

Integrated Announcement Boards

You can move integrated announcement boards that have been previously administered on the **Announcements/Audio Sources** screen to a new board location. You can also display a list of all administered integrated announcement circuit packs.

Field descriptions for page 1

Figure 378: Integrated Announcement Boards screen

```
display integrated-annc-boards                                      Page 1 of x

                         INTEGRATED ANNOUNCEMENT BOARDS
    Last Board Location Saved:

  Board                    Time                    Number of
  Location        Sfx      Remaining      Rate     Recordings          Checksum ID

1:01D09           C        206            32              7            047a
2:02817           C        248            32              1            00db
3:
4:
5:
```

Last Board Location Saved

Valid entries	Usage
Display-only	Applies to TN750 only; not applicable to VAL

Board Location

Valid entries	Usage
Display-only	The physical location of the integrated announcement circuit pack (UUCSS).

Sfx

Valid entries	Usage
Display-only	The circuit pack suffix letter(s).

Time Remaining

Valid entries	Usage
Display-only	The amount of recording time in seconds remaining on the circuit pack at the 64Kb rate.

Rate

Valid entries	Usage
Display-only	The announcement's compression rate.

Number of Recordings

Valid entries	Usage
Display-only	The number of nonzero-length announcement recordings or files on the circuit pack.

Checksum ID

Valid entries	Usage
Display-only	Applies to TN750 only; not applicable to VAL.

Integrated Announcement Translations

Use this screen to change board locations currently administered on the **Announcements/ Audio Sources** screen to a new board location.

Field descriptions for page 1

Figure 379: Change Integrated Announcement Translations screen

```
change integ-annc-brd-loc                                          Page 1 of x

                        CHANGE INTEGRATED ANNOUNCEMENT TRANSLATIONS
          Change all board location translations from board:     to board:


Changing board locations using this command will change
all currently administered "from" board locations on the
Announcements/Audio Sources screen to the "to" board location.
```

Change all board location translations from board

Valid entries	Usage
board; cabinet 1 to 3; carrier A-E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a VAL board that is currently administered on the **Announcements/Audio Sources** screen.

to board

Valid entries	Usage
board; cabinet 1 to 3; carrier A to E; slot 1 to 20; or gateway 1 to 10; module V1 to V9	Enter a VAL board to which you want to move announcement translations that are currently administered on the **Announcements/Audio Sources** screen.

Intercom Group

This screen assigns extensions to intercom groups.

Field descriptions for page 1

Figure 380: Intercom Group screen

```
change intercom-group n                                        Page 1 of x
                                 INTERCOM GROUP
                                               Group Number: n
                                        Length of Dial Code: _

GROUP MEMBER ASSIGNMENTS
              Ext      DC   Name
          1: ________ __
          2: ________ __
          3: ________ __
          4: ________ __
          5: ________ __
          6: ________ __
          7: ________ __
          8: ________ __
          9: ________ __
         10: ________ __
         11: ________ __
         12: ________ __
         13: ________ __
         14: ________ __
         15: ________ __
         16: ________ __
```

DC

This field assigns a dial code to an extension. The dial code is the code users must dial to make intercom calls to the corresponding extension.

Valid entries	Usage
1 or 2-digit code	The number of digits entered must exactly match the number assigned in the **Length of Dial Code** field. For example, if the **Length of Dial Code** field is set to 2, you must type 1 as 01 in the **DC** field. This field cannot be blank.

Ext

This field assigns an extension to the group.

Valid entries	Usage
an extension number	Enter a physical extension number. You cannot enter a VDN in this field.

Group Number

This display-only field shows the group's ID number.

Length of Dial Code

This field sets the number of digits that users must dial to access an extension in the group. (On Page 2, this is a display-only field.

Valid entries	Usage
1	Enter **1** if there are 9 or fewer members.
2	Enter **2** if there are 10 or more members.

Name

Display-only field. The server running Communication Manager fills in this field with the name from the **Station** screen.

Note:

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Related topics

See Using Telephones as Intercoms on page 527 for instructions.

See "Intercom" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Inter-Exchange Carrier (IXC) Codes

This screen allows identification of the IXC in the CDR record.

Field descriptions for page 1

Figure 381: Inter-Exchange Carrier Codes screen

```
change ixc-codes                                                    Page 1 of x
                              INTER-EXCHANGE CARRIER CODES
         IXC Codes Assignments (Enter up to 15)
         CDR   IXC                                CDR   IXC
         IXC   Access                             IXC   Access
         Code  Number     IXC Name                Code  Number     IXC Name
         1:    _______    _______________          9:   _______    _______________
         2:    _______    _______________         10:   _______    _______________
         3:    _______    _______________         11:   _______    _______________
         4:    _______    _______________         12:   _______    _______________
         5:    _______    _______________         13:   _______    _______________
         6:    _______    _______________         14:   _______    _______________
         7:    _______    _______________         15:   _______    _______________
         8:    _______    _______________
```

IXC Access Number

Valid entries	Usage
2 to 11 digits, **0** to **9** and *****	Enter the digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. No duplicate access numbers are allowed in the table.

IXC Name

Valid entries	Usage
0 to 15 characters	Description to identify the IXC

Field descriptions for page 2

Figure 382: Inter-Exchange Carrier Codes screen

```
change ixc-codes                                                    Page 2 of x
                                        IXC Prefix            IXC Code Format
                                   1.   ___                   ____
                                   2.   ___                   ____
                                   3.   ___                   ____
                                   4.   ___                   ____
                                   5.   ___                   ____
```

IXC Code Format

Valid entries	Usage
1 to 4 digit code format	
*****	
x	
X	
xxxx	For line 1
xxx	For line 2

IXC Prefix

Valid entries	Usage
1 to 3 digit prefix	
*****	
101	For line 1
10	For line 2

Intra-Switch CDR

This screen administers extensions for which Intra-Switch CDR is to be enabled.

Note:

Attendants are not allowed to be optioned for the Intra-Switch CDR feature.

When you enter the **`add`** command to add extensions, the system automatically begins after the last administered extensions. If you enter the **`change`** command, the system display begins with the first extension. If you enter the **`change`** command with an extension number, the system begins the display with that extension.

When you enter the command **`list intra-switch-cdr <extension> count x`**, the system lists "x" switch extensions administered for Intra-Switch CDR beginning with the extension specified by <extension>. For example, if you enter **`list intra-switch-cdr 81000 count 500`**, the system displays extension 81000 (if it is administered for Intra-Switch CDR) and the next 500 extensions that are administered for Intra-Switch CDR. The **`display`** command functions similarly to the **`change`** command.

Capacities

The Intra-Switch CDR extension capacities vary from server to server. For more information, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Field descriptions for page 1

Figure 383: Intra-Switch CDR screen

```
change intra-switch-cdr                                          Page   1 of   3
                           INTRA-SWITCH CDR

               Assigned Members: 0 of 5000 administered
      Extension          Extension          Extension           Extension
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________
    _____________      _____________      _____________      _____________

Use 'list intra-switch-cdr' to see all members.
Use 'change intra-switch-cdr <ext>' to add/change/remove other members.
```

Assigned Members

Displays the number of extensions currently administered for Intra-switch CDR.

Extension

Valid entries	Usage
Any valid extension	Enter the local extensions you want to track with Intra-Switch CDR. The number of extensions you can track might vary from one system to the next.

IP Address Mapping

This screen defines feature characteristics that depend on the IP address.

Note:
Enter data in either the **To IP Address** field or the **Subnet Mask** field.

Field descriptions for page 1

Figure 384: IP Address Mapping screen

```
change ip-network-map                                              Page 1 of X

                     IP ADDRESS MAPPING
                                                                     Emergency
                                   Subnet                 802.1Q     Location
FROM IP Address   (TO IP Address or Mask)   Region     VLAN      Extension
  1.__2.__3.__0    1.__2.__3.255    24       __1        ___3       ________
  1.__2.__4.__4    1.__2.__4.__4    32       __2        ___0       ________
  1.__2.__4.__5    1.__2.__4.__5    __       __3        ___0       ________
  1.__2.__4.__6    1.__2.__4.__9    __       __4        ___4       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
___.___.___.___  ___.___.___.___    __       ___        ____       ________
```

Emergency Location Extension

This field allows the system to properly identify the location of a caller who dials a 911 emergency call from this station. An entry in this field must be of an extension type included in the dial plan, but does not have to be an extension on the local system. It can be a UDP extension. The entry defaults to blank. A blank entry typically would be used for an IP softphone dialing in through PPP from somewhere outside your network.

If you populate the **IP Address Mapping** screen with emergency numbers, the feature functions as follows:

- If the **Emergency Location Extension** field in the **Station** screen is the same as the **Emergency Location Extension** field in the **IP Address Mapping** screen, the feature sends the extension to the Public Safety Answering Point (PSAP).
- If the **Emergency Location Extension** field in the **Station** screen is different from the **Emergency Location Extension** field in the **IP Address Mapping** screen, the feature sends the extension in the **IP Address Mapping** screen to the Public Safety Answering Point (PSAP).

Valid entries	Usage
0 to **9** (up to 7 digits)	Enter the emergency location extension for this station. Default is blank.

Note:
On the **ARS Digit Analysis Table** screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

From IP Address

Defines the starting IP address.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	See the *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

To IP Address

Defines the termination of a range of IP addresses.

If this field and the **Subnet Mask** fields are blank when submitted, the address in the **From IP Address** field is copied into this field.

The **Subnet Mask** field data is applied to the **From** field, creating the converted **To IP Address** field information.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	See the *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

To IP Address or Subnet Mask

The end of the IP address range can be specified by either entering the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

If the **Subnet Mask** field is used, then:

- The mask is applied to the **From IP Address** field, placing zeros in the non-masked rightmost bits. This becomes the stored "From" address.
- The mask is applied to the **To IP Address** field, placing 1's in the non-masked rightmost bits. This becomes the stored "To" address.

If this field and the **To IP Address** fields are blank when submitted, the address in the **From IP Address** field is copied into the **To IP Address** field.

Valid entries	Usage
0 to 32 or blank	Enter the last IP address in the range or the **From IP Address** and the number of bits of the subnet mask.

Region

Identifies the network region for the IP address range. For SIP, the value for this field must correlate with the configured network region for this range of addresses.

Valid entries	Usage
1 to **250**	Enter the network region number for this interface. This field must contain a non-blank value if the **From IP Address** field on the same row contains a non-blank value.

VLAN

Sends VLAN instructions to IP endpoints such as IP telephones and softphones. This field does not send VLAN instructions to the PROCR (S8300/S87XX Servers), CLAN, and Media Processor boards.

Valid entries	Usage
0 to **4094**	Specifies the virtual LAN value.
n	Disabled

IP Codec Set

The **IP Codec Set** screen allows you to specify the type of codec used for voice encoding and companding (compression/decompression). The main difference between codecs is in the compression algorithm used; some codecs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but might also introduce transmission delays and lower voice quality. Codecs are used for VoIP links between any two VoIP resources or endpoints, e.g., IP telephone to IP telephone, IP telephone to Media Gateway, Media Gateway to Media Gateway, etc. The order in which the codecs are listed on this screen is the order of your preference of usage. A trunk call between any two VoIP resources or endpoints is set up to use the first common codec listed.

The default codec is set for G711MU. The G711MU provides the highest voice quality because it does the least amount of compression, but it uses the most bandwidth. The G711MU default setting can be changed to one of two other codecs (and their "flavors") if the G711MU does not meet your desired voice-quality/bandwidth trade-off specification. For example, if a far-end server is not running Avaya Communication Manager, you might need to change the codec to match one that is supported by that server's software.

Field descriptions for page 1

This screen allows you to define the allowed codecs and packet sizes used between VoIP resources. You can also enable silence suppression on a per-codec basis. This screen will dynamically display the packet size in milliseconds for each codec in the set, based on the number of frames you administer per packet.

Figure 385: IP Codec Set screen - page 1

```
change ip-codec-set n                                           Page 1 of x

                              IP Codec Set

      Codec Set: 1

     Audio         Silence         Frames    Packet
     Codec         Suppression     Per Pkt   Size (ms)
1: G.711MU_______    y               3         30
2:_______________    _               _
3:_______________    _               _
4:_______________    _               _
5:_______________    _               _
6:_______________    _               _
7:_______________    _               _

Media Encryption:
1: aes
2: aea
3: srtp-aescm128-hmac80
```

Audio Codec

Specify the audio codec used for this codec set.

Valid entries	Usage
G.711A (a-law) **G.711MU** (mu-law) **G.722-64k** **G.722.1-24k** **G.722.1-32k** **G.723-5.3** **G.723-6.3** **G.726A-32K** **G.729** **G.729A** **G.729B** **G.729AB** **SIREN14-24k** **SIREN14-32k** **SIREN14-48k** **SIREN14-S48k** **SIREN14-S56k** **SIREN14-S64k** **SIREN14-S96k**	Enter the codec to be used for this codec set.

Important:

Avaya recommends that you include at least two codecs for every telephone in order to avoid incompatible codecs. Use the codecs specified in the following table for the telephones shown.

Telephone	Codec to use
All Avaya IP Telephones	**G.711**, **G.729B**
4601 4602 4602SW 4620SW 4621SW 4622SW	add **G.726A** (requires firmware R2.2)

Codec Set

Display only. Shows the number assigned to this Codec Set.

Frames Per Pkt

Specify the number of frames per packet up to a packet size of 60 milliseconds (ms).

Valid entries	Usage
1 to **6** or blank	Default frame sizes for codecs: • G.711 and G.729: 2 frames (20 ms) • G.723: 3 frames (30 ms) • G.726A: 1 frame (10 ms)

Media Encryption

This field appears only if the **Media Encryption over IP** feature is enabled in the license file. Use this field to specify a priority listing of the three possible options for the negotiation of encryption. Communication Manager attempts to provide bearer encryption per this administered priority order.The selected option for an IP codec set applies to all codecs defined in that set.

Valid entries	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: ● Server-to-gateway (H.248) ● Gateway-to-endpoint (H.323)
aea	Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: ● All endpoints within a network region using this codec set must be encrypted. ● All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
	1 of 2

Valid entries	Usage
	SRTP is a media encryption standard defined in RFC 3711 as a profile of RTP. Communication Manager 4.0 supports the following functionality as given in RFC 3711: ● Encryption of RTP (optional but recommended) ● Authentication of RTCP streams (mandatory) ● Authentication of RTP streams (optional but recommended) ● Protection against replay **Note:** In Communication Manager 4.0, SRTP encryption is supported by 96xx telephones only.
1-srtp-aescm128-hmac80	1-Encrypted/Authenticated RTP with 80-bit authentication tag
2-srtp-aescm128-hmac32	2-Encrypted/Authenticated RTP with 32-bit authentication tag
3-srtp-aescm128-hmac80-unauth	3-Encrypted RTP but not authenticated
4-srtp-aescm128-hmac32-unauth	4-Encrypted RTP but not authenticated
5-srtp-aescm128-hmac80-unenc	5-Authenticated RTP with 80-bit authentication tag but not encrypted
6-srtp-aescm128-hmac32-unenc	6-Authenticated RTP with 32-bit authentication tag but not encrypted
7-srtp-aescm128-hmac80-unenc-unauth	7-Unencrypted/Unauthenticated RTP
8-srtp-aescm128-hmac32-unenc-unauth	8-Unencrypted/Unauthenticated RTP **Note:** For stations, the only value supported is **srtp-aescm128-hmac80**. H.323 IP trunks support all eight of the listed algorithms.
none	Media stream is unencrypted. This is the default.
	2 of 2

Packet Size (ms)

A display-only field showing the packet size in milliseconds.

Silence Suppression

Enables RTP-level silence suppression on the audio stream.

Valid entries	Usage
y/n	Enter **y** to enable RTP-level silence suppression on the audio stream.

Field descriptions for page 2

Use this screen to assign the following characteristics to a codec set:

- Whether or not Direct-IP Multimedia is enabled for videophone transmissions
- Whether or not endpoints in the assigned network region can route fax, modem, or TTY calls over IP trunks

  Note:

  For more information on modem/fax/TTY over IP, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

- Which mode the system uses to route the fax, modem, or TTY calls
- Whether or not redundant packets will be added to the transmission for higher reliability and quality

These characteristics must be assigned to the codec set, and the codec set must be assigned to a network region for endpoints in that region to be able to use the capabilities established on this screen.

CAUTION:

If users are using Super G3 fax machines as well as modems, do *not* assign these fax machines to a network region with an IP Codec set that is modem-enabled as well as fax-enabled. If its Codec set is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and assign the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

Note:

Transporting modem tones over IP between Avaya Communication Manager systems is a proprietary implementation. Also, FAX transport implementations, other than T.38 are proprietary implementations.

Figure 386: IP Codec Set screen page 2

```
change ip-codec-set n                                         Page   2 of   x

                             IP Codec Set

                                 Allow Direct-IP Multimedia? y
        Maximum Bandwidth Per Call for Direct-IP Multimedia: 256:Kbits

                    Mode        Redundancy

               FAX  relay          0

             Modem  off            0

           TDD/TTY  us             0

     Clear-channel  n              0
```

Allow Direct-IP Multimedia

Valid entries	Usage
y/n	Enter **y** to allow direct multimedia via the following codecs: ● H.261 ● H.263 ● H.264 (video) ● H.224 ● H.224.1 (data, far-end camera control).

Clear-channel

For more information on Clear Channel, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Valid entries	Usage
y/n	Enter **y** to indicate that this codec set supports BRI data calls. **Note:** Clear Channel data transmission is supported on the TN2602AP IP Media Resource 320 circuit pack and the TN2302AP circuit pack.

FAX Mode

Valid entries	Usage
off	Turn off special fax handling when using this codec set. In this case, the fax is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax.
relay	For users in regions using this codec, use Avaya relay mode for fax transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
pass-through	For users in regions using this codec, use pass-through mode for fax transmissions over IP network facilities. This mode uses G.711-like encoding.
t.38-standard	For users in regions using this codec, use T.38 standard signaling for fax transmissions over IP network facilities.

Note:

If you have a telephone that is on an IP trunk too close to a fax machine, the handset can pick up the tones from the fax machine and change itself into the fax mode. To prevent this, set the **FAX** field to **off**, and put the FAX machines in an ARS partition that uses only circuit switched trunks, even for IGW FAX calls.

Maximum Bandwidth Per Call for Direct-IP Multimedia (value)

This field appears only when **Allow Direct-IP Multimedia** is **y**.

Valid entries	Usage
1 to **9999**	Enter the bandwidth limit for Direct-IP Multimedia transmissions on this codec set. Default is **256**.

Maximum Bandwidth Per Call for Direct-IP Multimedia (units)

This field displays only when **Allow Direct-IP Multimedia** is **y**.

Valid entries	Usage
kbits **mbits**	Enter the unit of measure corresponding to the value entered for bandwidth limitation. Default is **kbits**.

Modem Mode

Valid entries	Usage
off	Turn off special modem handling when using this codec set. In this case, the modem transmission is treated like an ordinary voice call. This is the default for new installations and upgrades to Communication Manager R2.1. With a codec set that uses G.711, this setting is required to send modem calls to non-Avaya systems.
relay	For users in regions using this codec, use relay mode for modem transmissions over IP network facilities. Avaya V.32/FNBDT Modem Relay is supported when using modem **relay** mode. **Note:** Modem over VoIP in relay mode is currently available only for use by specific analog telephones that serve as Secure Telephone Units (STUs). Contact your Avaya technical support representative for more information.
pass-through	For users in regions using this codec, use pass-through mode for modem transmissions over IP network facilities. Avaya V.8 Modem Pass-Thru is supported when using modem **pass-through** mode.

Redundancy

Valid entries	Usage
0 to **3**	Enter the number of duplicate or redundant packets that are sent in addition to the primary packet for all Modes except **pass-through** and **Clear-channel**. The default is **0**.

TDD/TTY Mode

Valid entries	Usage
off	Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send TTY calls to non-Avaya systems. However, there might be errors in character transmissions.
US	For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1.
	1 of 2

Valid entries	Usage
UK	For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities.
pass-through	For users in regions using this codec, use pass-through mode for TTY transmissions over IP network facilities.
	2 of 2

IP Interfaces

Use the **IP Interfaces** screen to assign a network region to an IP interface device, or to administer Ethernet options. The fields shown appear when the **`add`**, **`change`**, **`display`**, or **`remove`** command is used.

Note:

For information about Processor Ethernet interfaces, see Setting up Processor Ethernet on page 593.

The appearance of the **IP Interfaces** screen can vary according to the interface type you are administering, and your system's configuration. The figures shown are examples, intended to show most of the fields that might appear on this screen. Your own screen might vary from these examples. The list of field descriptions that follows the figures is intended to be comprehensive, and to include information on all fields that might appear. The field descriptions are in alphabetical order for quick reference.

Note:

When you start the process of administering the IP interface for the TN2602AP circuit pack, any active calls continue to use the TN2602AP circuit pack's physical IP address for the connection, not the virtual IP address you are setting in this procedure. Therefore, any of these calls, if they continue after you complete this procedure, will drop in the event of an interchange.

Field descriptions for page 1

Figure 387: IP Interfaces - Type board location screen

```
add ip-interface 1a03                                               Page 1 of 1
                                   IP INTERFACES

                           Critical Reliable Bearer? y
                   Type: MEDPRO
                   Slot: 01A03                              Slot: 01A03
            Code/Suffix: TN2602                      Code/Suffix: TN2602
              Node Name:                               Node Name:
             IP Address:                              IP Address:
            Subnet Mask: 255.255.255.0
        Gateway Address: . . .
   Enable Ethernet Port? y              Enable Ethernet Port? y
         Network Region: 20
                   VLAN: n                                  VLAN: n
          VOIP Channels: xxx
 Shared Virtual Address: 255.255.255.255
      Virtual MAC Table:                 Virtual MAC Address:

                                  ETHERNET OPTIONS
                  Auto? n                                  Auto? n
                 Speed: 100Mbps                           Speed: 100Mbps
                Duplex: Full                             Duplex: Full
```

Figure 388: IP Interfaces - Type procr screen

```
add ip-interface procr                                            Page 1 of x
                                   IP INTERFACES

                   Type: PROCR
             IP Address: 172.28.4.1
            Subnet Mask: 255.255.255.0

   Enable Ethernet Port? y                    Allow H.323 Endpoints?
         Network Region: 20                    Allow H.248 Gateways?
                                                Gatekeeper Priority?

Target socket load:
```

Figure 389: IP Interfaces - Type clan screen

```
add ip-interface 01a08                                          Page 1 of x
                                   IP INTERFACES

                   Type: CLAN
                   Slot: 01A08
            Code/Suffix: TN799
              Node Name: makita-clan1
             IP Address: 172.28.5.254
            Subnet Mask: 255.255.255.0                         Link?
        Gateway Address:
   Enable Ethernet Port? y                  Allow H.323 Endpoints?
         Network Region: 20                  Allow H.248 Gateways?
                   VLAN: n                    Gatekeeper Priority?

Target socket load and Warning level:
      Receive Buffer TCP Window Size:

                        ETHERNET OPTIONS
              Auto? y
```

Allow H.248 Gateways

This field controls whether or not H.248 media gateways (G7000, G350, G250) can register on the interface.

Valid entries	Usage
y/n	On a simplex main server, enter **y** to allow H.248 endpoint connectivity to the PE interface. Enter **n** if you do not want H.248 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.248 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**.

Allow H.323 Endpoints

This field controls whether or not IP endpoints can register on the interface.

Valid entries	Usage
y/n	On a simplex main server, enter **y** to allow H.323 endpoint connectivity to the PE interface. Enter **n** if you do not want H.323 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.323 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**. For more information, see Setting up Processor Ethernet on page 593.

Code/Sfx

Valid entries	Usage
y/n	Circuit pack TN code and suffix. Display-only for TN2602AP when **Critical Reliable Bearer** is **n**. The second (right-side) **Code/Sfx** field is automatically populated based on the corresponding **Slot** field information, when **Critical Reliable Bearer** is **y**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Critical Reliable Bearer

Appears when the board code of slot location is TN2602.

Valid entries	Usage
y/n	A **y** entry indicates that two TN2602AP circuit packs are duplicated in a port. If **y**, a second column of information appears, for administering the second shared circuit pack. Default is **n**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Enable Ethernet Pt

Allows use of the Ethernet port.

Valid entries	Usage
y/n	Enter **y** to indicate that the Ethernet port associated with the TN2602AP circuit pack is in service. If this is an active board, set to **n** only when there is no standby, or when the standby has been disabled. **Note:** Enter **n** in this field before you make changes to the screen.

Gateway Address

Valid entries	Usage
0 to **255**	The IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Gatekeeper Priority

Appears only if **Allow H.323 Endpoints** is **y** and the Communication Manager server is a main server or an LSP. This field does not display on an ESS server. This field allows a priority to be set on the interface. This affects where the interface appears on the gatekeeper list.

Valid entries	Usage
1 to **9**	Enter the desired priority number. The value in this field is used on the alternate gatekeeper list. The lower the number, the higher the priority. Default is **5**. For more information, see Setting Alternate Gatekeeper List (AGL) priorities on page 604.

IP Address

This field is display-only, taken from the **IP Node Names** screen, based on the node name entered.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range **0** to **255**)	Displays the IP address of the interface.

Link

This display-only field shows the administered link number for an Ethernet link.

Valid entries	Usage
y/n	Shows the unique number for the Ethernet link assigned on the **Data Module** screen.

Network Region

Identifies the network region for the specified interface.

Valid entries	Usage
1 to **250**	Enter the value of the Network Region where the TN2602AP resides. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Node Name

The unique node name for the IP interface administered on the **Node Names** screen.

Valid entries	Usage
Character string (up to 15 characters max.)	Enter the node name associated with the IP address of the TN2602AP circuit pack.

Receive Buffer TCP Window Size

Valid entries	Usage
512 to **8320**	The number of bytes allotted for the buffer that receives TCP data for a TN799 (CLAN) circuit pack. The default is **512**.

Shared Virtual Address

This field appears when **Critical Reliable Bearer** is **y**.

Valid entries	Usage
0 to **255**	The virtual IP address shared by the two TN2602AP circuit packs, when duplicated. This address enables Communication Manager to connect endpoints through the TN2602AP circuit packs to the same address, regardless of which one is actually active. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Slot

Displays the slot location entered in the command line. Enter the location of the second TN2602AP circuit pack for a non-duplicated board. The second (right-side) **Slot** field is automatically populated when **Critical Reliable Bearer** is **y**.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth character are the slot number.

Subnet Mask

The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
characters	Enter the Subnet Mask for TN2602AP. This entry also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Target socket load

This field appears when **Type** is **procr**. Use this field for load balancing endpoint traffic across multiple IP interfaces. The value that you enter in the **Target socket load** field controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number that you allocate, the system continues to add sockets until the interface is at its maximum capacity. For more information on load balancing, see Using load balancing on page 603.

Note:

The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid entries	Usage
1 to platform maximum as follows: ● S87XX series: **3500** ● S8500: **3500** ● S8400: **2500** ● CHAWK/BOXTER: **2000** ● VM/BLADE: **1700**	Enter the maximum number of sockets targeted for this interface. The default is 80% of the platform maximum.

Target socket load and Warning level

This field appears when **Type** is **clan**. The value that you enter in the **Target socket load and Warning level** field controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number that you allocate, the system continues to add sockets until the interface is at its maximum capacity. If the targeted percentage is exceeded on a CLAN, a warning alarm is generated.

If there is only one IP interface within a priority, the **Target socket load and Warning level** field is no longer used for load balancing. You can still enter a value in this field to receive an error or a warning alarm if the targeted value is exceeded.

Note:
The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid entries	Usage
1 to **499**	Enter the maximum number of sockets targeted for this interface. If the number of sockets exceeds the targeted number, a warning alarm is generated.The default is **400**.

Type

Identifies the type of IP interface.

Valid entries	Usage
clan **VAL** **medpro** **procr**	This field is auto-populated based on the slot location specified in the command line.

Virtual MAC Address

This field appears when **Critical Reliable Bearer** is **y**.

Valid entries	Usage
12 alpha-numeric characters	Display-only. The Virtual MAC address that is shared by duplicated TN2602AP circuit packs. Automatically populated based on the Virtual MAC address table. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Virtual MAC Address Table

Valid entries	Usage
1 through **4**	Enter the table number from which the virtual MAC address, shared by duplicated TN2602AP circuit packs, will be obtained. Default is **1**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

VLAN

This field sends VLAN instructions to the PROCR (S8300/S87XX Servers), CLAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. This field cannot be administered for VAL boards.

Valid entries	Usage
0 to **4095**	Specifies the virtual LAN value.
n	Disabled. This is the default.

VOIP Channels

Valid entries	Usage
0 (will not support voice calls) **80** (low density) **320** (standard)	This field indicates the number of VoIP channels that are allocated to the associated TN2602. This VoIP channel capacity is an administered value. Users will be blocked from administering 80 or 320 VoIP channels if there is no available capacity for the corresponding **Maximum TN2602 boards with 80 VoIP Channels** / **Maximum TN2602 boards with 320 VoIP Channels** license features. Default is **0**. This number also applies to the second TN2602AP circuit pack when **Critical Reliable Bearer** is **y**. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

ETHERNET OPTIONS

With each new system or IP board installation, one standard procedure should be to apply matching speed/duplex settings to each IP board and its corresponding Ethernet switch port. Then these fields can be used to verify the configured settings. You must set the **Enable Ethernet Pt?** field to **n** before you can make changes to these fields.

Auto

Valid entries	Usage
y/n	Enter **y** for auto-negotiation or **n** for manual speed and duplex settings. Default is **y**.

Duplex

Valid entries	Usage
Full **Half**	Enter the duplex settings for this IP board. When **Speed** is set to 100Mbps, this field defaults to **Full**. You still have the option of changing the value to **Half**. Default is **Half**.

Speed

Valid entries	Usage
10Mbps **100Mbps**	Enter the speed of the Ethernet connection. When **Auto** is set to **n**, the only speed option available for the TN2602AP circuit pack is 100Mbps. This is the default and cannot be changed.

IP Network Region

Use this screen to configure within-region and between-region connectivity settings for all VoIP resources and endpoints within a given IP region. The first page is used to modify the audio and QoS settings. The **Codec Set** field on this page reflects the CODEC set that must be used for connections between telephones within this region or between telephones and MedPro/Prowler boards and media gateways within this region. The ability to do NAT shuffling for direct IP-to-IP audio connections is also supported. Use the **IP Address Mapping** screen to administer network regions.

Field descriptions for page 1

Figure 390: IP Network Region screen

```
change ip-network-region n                                        Page 1 of x
                              IP NETWORK REGION
  Region: n
Location:                  Authoritative Domain:
    Name:
                                Intra-region IP-IP Direct Audio: n
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: n
   Codec Set: 1                            IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3028                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value:                      Server IP Address:  .   .   .
        Audio PHB Value:                            Server Port: 5005
        Video PHB Value:

802.1P/Q PARAMETERS                    RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 7
                                     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? y
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled? y
   Keep-Alive Interval (sec): 6                        RSVP Profile:
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 40
```

Note:

The **`display ip-network-region`** command displays the values that you assign on this screen.

Authoritative Domain

The name or IP address of the domain for which this network region is responsible (i.e., authoritative).

Valid entries	Usage
Up to 20 characters or blank.	Enter the name or IP address of the domain for which this network region is responsible. Note that this will appear in the "From" header of any SIP Enablement Services (SES) messages.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Region

A display-only field indicating the number of the network region being administered. Network regions are defined on the **IP Address Mapping** screen.

MEDIA PARAMETERS

Codec Set

Specifies the codec set assigned to the region

Valid entries	Usage
1 to **7**	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a network region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP telephones within the region are behind two or more firewalls.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone/softphone itself (without being translated by NAT). IP telephones must be configured behind a NAT device *before* this entry is enabled.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device *before* this entry is enabled.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. An **n** entry might be used if, for example, the IP telephones within the region are behind two or more firewalls.
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device *before* this entry is enabled.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP telephones must be configured behind a NAT device *before* this entry is enabled.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the Avaya S8XXX Server's IP circuit pack in IP format, without first going through the Avaya DEFINITY TDM bus. Default is **n**.

Location

Specifies the location of the IP network region, allowing correct date and time information, and trunk routing based on IP network region.

Note:

If the Multinational Locations feature is enabled, and IP telephones derive their network region from the IP Network Map, you must administer this field with a valid value (1 to 250). This allows the IP endpoint to obtain a VoIP resource.

Valid entries	Usage
1 to **44**	(For CSI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1 to **250**	(For Avaya S8300/S87XX Servers) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN or the media gateway that the endpoint registered with.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Enter **y** to send RTCP Reports to a special server, such as for the VMON tool. If this field is set to **y**, then the **RTCP Monitor Server Parameters** fields appear. **Note:** Regardless of how this field is administered, RTCP packets are always sent peer-to-peer.

UDP Port Range

UDP Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
1024 to **65534** defaults: **2048** to **3028**	Enter the lowest UDP port number to be used for audio transport.

UDP Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
1025 to **65535** defaults: **2048** to **3028**	Enter the highest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the **Use Default Server Parameters** field is set to **n** and the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
5 to **30**	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
0 to **255** in *nnn.nnn.nnn.nnn* format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the **Use Default Server Parameters** field is set to **n** and the and the **RTCP Enabled** field is set to **y**.

Valid entries	Usage
1 to **65535**	Enter the port for the RTCP Monitor server. Default is **5005**.

Use Default Server Parameters

This field only appears when the **RTCP Reporting Enabled** field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP-Options System Parameters screen. If set to **y**, you must complete the **Default Server IP Address** field on the **IP Options System Parameters** screen.
n	If you enter **n**, you need to complete the **Server IP Address**, **Server Port**, and **RTCP Report Period** fields that appear.

DIFFSERV/TOS PARAMETERS

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the Audio PHB codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the DiffServ Audio PHB value. Default is **46**.

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the **IP TOS** field to support the DiffServ codepoint.

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the Call Control PHB value. Default is **34**.

Video PHB Value

Valid entries	Usage
0 to **63**	Enter the decimal equivalent of the DiffServ Video PHB value. Default is **26**.

802.1P/Q PARAMETERS

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the Audio 802.1p priority value. Changes take effect after circuit pack reset, phone reboot, or system reset.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0 to **7**	Specifies the 802.1p priority value. Changes take effect after circuit pack reset, phone reboot, or system reset.

Video 802.1p Priority

Valid entries	Usage
0 to **7**	Specifies the Video 802.1p priority value. Changes take effect after circuit pack reset, phone reboot, or system reset.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

The entry in this field controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the **RSVP Enabled** field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	This limits end-to-end queuing delay from sender to receiver. This setting is best for VoIP applications.
controlled-load	This subset of **guaranteed-service** provides for a traffic specifier, but not end-to-end queuing delay.

RSVP Refresh Rate (secs)

This field only appears if the **RSVP Enabled** field is set to **y**.

Valid entries	Usage
1 to **99**	Enter the RSVP refresh rate in seconds.

RSVP unreserved (BBE) PHB Value

Valid entries	Usage
0 to **63**	The BBE codepoint is used whenever an RSVP reservation is being obtained (pending), or has failed in some way, to provide better-than-best service to the voice stream.

H.323 IP ENDPOINTS

H.323 Link Bounce Recovery

A **y** entry in this field enables the H.323 Link Bounce Recovery feature for this network region. An **n** disables the feature.

Valid entries	Usage
y/n	Specifies whether to enable H.323 Link Bounce Recovery feature for this network region. Default is **y**.

Idle Traffic Interval (seconds)

This field represents the maximum traffic idle time after which a TCP Keep-Alive (KA) signal is sent from the endpoint.

Valid entries	Usage
5 to **7200**	Enter the maximum traffic idle time in seconds. Default is **20**.

Keep-Alive Interval (seconds)

Use this field to set the interval between TCP Keep-Alive re-transmissions. When no ACK is received for all retry attempts, the local TCP stack ends the TCP session and the associated socket is closed.

Valid entries	Usage
1 to **120**	Specify the interval between KA retransmissions in seconds. Default is **5**.

Keep-Alive Count

Use this field to set the number of times the Keep-Alive message is transmitted if no ACK is received from the peer.

Valid entries	Usage
1 to **20**	Specify the number of retries when if no ACK is received. Default is **5**.

Field descriptions for Page 2

This page covers the information for Inter-Gateway Alternate Routing (IGAR), backup server names in priority order, and security procedures.

Figure 391: IP Network Region screen

```
 change ip-network-region n                                   Page 2 of x

                                IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY
Incoming LDN Extension:
Conversion to Full Public Number - Delete: ___Insert: ___
 Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n


BACKUP SERVERS IN PRIORITY ORDER            H.323 SECURITY PROCEDURES
1                                           1
2                                           2
3                                           3
4                                           4
5
6                                          Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY

If Inter-Gateway Alternate Routing (IGAR) is enabled for any row on pages 3 through 19, you must complete the following fields for each network region in order to route the bearer portion of an IGAR call. For more information on Inter-Gateway Alternate Routing, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Conversion to Full Public Number - Delete

Valid entries	Usage
0 to **7**	Enter the digits to delete.

Conversion to Full Public Number - Insert

Valid entries	Usage
up to 13 digits	Enter up to 13 digits to insert, or blank. International numbers should begin with '+'. **Note:** The optional "+" at the beginning of the inserted digits is an international convention indicating that the local international access code (e.g., 011 in North America and 00 in Europe) must be dialed before the number.

Dial Plan Transparency in Survivable Mode

Valid entries	Usage
y/n	Enter **y** to enable the Dial Plan Transparency feature when a media gateway registers with a local survivable processor (LSP), or when a port network registers with an Enterprise Survivable Server (ESS). Default is **n**.

Incoming LDN Extension

Valid entries	Usage
Valid unused extension	Assign an unused Listed Directory Number for incoming IGAR calls.

Maximum Number of Trunks to Use for IGAR

It is necessary to impose a limit on the trunk usage in a particular pot network in a network region when IGAR is active. The limit is required because if there is a major IP WAN network failure, it is possible to use all trunks in the network region(s) for IGAR calls.

Valid entries	Usage
1 to **999**, or blank	Enter the maximum number of trunks to be used for Inter-gateway alternate routing (IGAR).

Note:
The S8500 supports up to 800 IP trunks (via license file limitations), which is less than the S87XX limit, but the overall maximum number of trunk members is the same as on the S87XX: 8000.

BACKUP SERVERS IN PRIORITY ORDER

This section lists the backup server names in priority order. Backup server names should include LSP server names, but should not include ESS server names. The six fields under this label allow any valid node name as an entry. Valid node names can include names of Customer LANs, ICCs, and LSPs.

H.323 SECURITY PROCEDURES

Use this field to select the permitted security profile(s) for endpoint registration in this network region. At least one security procedure entry must be present when this screen is submitted; otherwise, no endpoint will be permitted to register from the region.

Valid entries	Usage
challenge	Includes the various methods of PIN-based challenge/response schemes in current use; relatively weak.
pin-eke	The H.235 Annex H SP1
strong	Permits use of any strong security profile; at present, only the **pin-eke** profile fits in this category.
all	Includes all of the above security profiles.
none	No security profile is required; permits use of an endpoint without user authentication (use with caution).

Allow SIP URI Conversion

Use this field to administer whether or not a SIP URI should be permitted to change. Degrading the URI from sips//: to sip//: may result in a less secure call. This is required when SIP SRTP endpoints are allowed to make and receive calls from endpoints that do not support SRTP.

Note:

If you enter **n** for no URI conversion, then calls from SIP endpoints that support SRTP made to other SIP endpoints that do *not* support SRTP will fail.

Valid entries	Usage
y/n	Enter **y** to allow conversion of SIP URIs. Default is **y**.

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS

Near End Establishes TCP Signaling Socket

Use this field to indicate whether Communication Manager (the near end) can establish the TCP socket for H.323 IP endpoints in this network region.

Valid entries	Usage
y	When set to **y**, Communication Manager determines when to establish the TCP socket with the IP endpoints, assuming the endpoints support this capability. This is the default.
n	When set to **n**, the IP endpoints always attempt to set up the TCP socket immediately after registration. This field should be set to **n** only in network regions where a non-standard H.323 proxy device or a non-supported network address translation (NAT) device would prevent the server from establishing TCP sockets with H.323 IP endpoints.

Near End TCP Port Min

Use the **Near End TCP Port Min** and **Near End TCP Port Max** fields to specify a range of port numbers to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1).

Valid entries	Usage
1024 to **65531**	Set the minimum port value to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. Default is **61440**.

Near End TCP Port Max

Use the **Near End TCP Port Min** and **Near End TCP Port Max** fields to specify a range of port numbers to be used by the Control Lan (C-LAN) or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1).

Valid entries	Usage
1028 to **65535**	Set the maximum port value to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. Default is **61444**.

Field descriptions for Page 3

Each page from page 3 on shows the inter-region connectivity for 15 region pairs. To accommodate the maximum of 250 regions for Linux platforms, up to 17 pages are available for this purpose.

Figure 392: Inter Network Region Connection Management screen

```
change ip-network-region n                                         Page   3 of x

                   Inter Network Region Connection Management

 src dst codec direct   Total          Video                          Dyn
 rgn rgn  set  WAN   WAN-BW limits Norm Prio Shr Intervening-regions   CAC IGAR
 3   1    1       y    256:Kbits             y                               n
 3   2    1       n       :NoLimit           n                               n
 3   3    1               :NoLimit                                           n
 3   4    1       n       :NoLimit           n                               n
 3   5    1       n       :NoLimit           n                               n
 3   6    1       y       :NoLimit           y                               n
 3   7    1       y     10:Calls             y                               n
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

Audio WAN-BW limits (units)

The entry in this field the unit of measure corresponding to the value entered for bandwidth limitation.

Valid entries	Usage
Calls **Dynamic** **Kbits/sec** **Mbits/sec** **NoLimit**	This field allows you to limit bandwidth by number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or it can be left blank. Default is blank.

codec-set

Indicates which codec set is to be used between the two regions.

Valid entries	Usage
1 to **7**, **pstn**	If the two regions are not connected at all, this field should be blank. When the codec set is blank, the **direct-WAN**, **WAN-BW-limits**, and **Intervening-regions** entry fields are not displayed. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions.

direct-WAN

The entry in this field indicates whether the two regions (source and destination) are directly connected by a WAN link.

Valid entries	Usage
y/n	The default value is **y(es)** if the **codec-set** field is not blank. If so, the **WAN-BW-limits** field displays, but the **Intervening-regions** fields do not. If the **direct-WAN** field is set to **n(o)**, then the **WAN-BW-limits** field does not display, but the **Intervening-regions** fields are displayed.

dst rgn

The entry in this field identifies the destination region for this inter-network connection.

Valid entries	Usage
1 to **250**	Display-only. Shows the destination region for this inter-network connection.

Dynamic CAC Gateway

This field only appears if the **Audio WAN-BW- limit** field is set to **dynamic**. The gateway must be configured to be a CAC (Call Admission Control) gateway.

Valid entries	Usage
1 to **250**	Set the gateway that reports the bandwidth-limit for this link. Default is blank.

IGAR

This field allows pair-wise configuration of Inter-Gateway Alternate Routing between network regions. If the field is set to **y**, the IGAR capability is enabled between the specific network region pair. If it is set to **n**, the IGAR capability is disabled between the network region pair. The (**f**)orced option moves all traffic onto the PSTN.

For more information on Inter-Gateway Alternate Routing, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Valid entries	Usage
y/	Enter **y** to enable IGAR capability between this network region pair.
n	IGAR capability between this network region pair is disabled. The default is **n**, except when **codec set** is **pstn**. When **codec set** is **pstn**, this field defaults to **y**.
f	Forced. This option can be used during initial installation to verify the alternative PSTN facility selected for a network region pair. This option can also be used to temporarily move traffic off of the IP WAN if an edge router is having problems or an edge router needs to be replaced between a network region pair.

Intervening-regions

The entry in this field allows entry of intervening region numbers between the two indirectly-connected regions.

Valid entries	Usage
1 to **250**	Enter up to four intervening region numbers between the two indirectly-connected regions. **Note**: Entry is not allowed for indirect region paths until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

src rgn

The entry in this field identifies the source region for this inter-network connection.

Valid entries	Usage
1 to **250**	Display-only. Shows the source region for this inter-network connection.

Video (Norm)

Valid entries	Usage
0 to **9999** for Kbits, **0** to **65** for Mbits, or blank for **NoLimit**	Set the amount of bandwidth that you want to allocate for the normal video pool to each IP network region.

Video (Prio)

Valid entries	Usage
0 to **9999** for Kbits, **0** to **65** for Mbits, or blank for **NoLimit**	Set the amount of bandwidth that you want to allocate for the priority video pool to each IP network region.

Video (Shr)

Valid entries	Usage
y/n	Specify whether the normal video pool can be shared for each link between IP network regions.

WAN-BW limits (value)

This field is used for entry of the bandwidth limits for direct WAN links.

Valid entries	Usage
1 to **9999**	Values for this field can be entered in the number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or left blank. Default is blank. **Note**: For Release 2.0, the number must be less than or equal to 65 when the units part of the field is set to Mbits/sec.

WAN-BW limits (units)

The entry in this field the unit of measure corresponding to the value entered for bandwidth limitation.

Valid entries	Usage
Calls **Kbits/sec** **Mbits/sec** **NoLimit**	This field allows you to limit bandwidth by number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or **NoLimit**. Default is **NoLimit**.

IP Node Names

Use this screen to administer node names and IP addresses for the switch and the terminal server media processors administered on the **IP Interfaces** screen.

Note:

The Processor Ethernet interface node name (**procr**) automatically appears on the **IP Node Names** screen. The PE interface node name cannot be added to the **IP Node Names** screen. The line containing the keyword **procr** displays the IP address. For more information on Processor Ethernet, see Setting up Processor Ethernet on page 593.

Field descriptions for page 1

Figure 393: IP Node Names screen

```
change node-names ip                                            Page 1 of X

                              IP NODE NAMES

Name                    IP Address                Name             IP Address
 1. ______________     ___.___.___.___  17. ______________     ___.___.___.___
 2. ______________     ___.___.___.___  18. ______________     ___.___.___.___
 3. ______________     ___.___.___.___  19. ______________     ___.___.___.___
 4. ______________     ___.___.___.___  20. ______________     ___.___.___.___
 5. ______________     ___.___.___.___  21. ______________     ___.___.___.___
 6. ______________     ___.___.___.___  22. ______________     ___.___.___.___
 7. ______________     ___.___.___.___  23. ______________     ___.___.___.___
 8. ______________     ___.___.___.___  24. ______________     ___.___.___.___
 9. ______________     ___.___.___.___  25. ______________     ___.___.___.___
10. ______________     ___.___.___.___  26. ______________     ___.___.___.___
11. ______________     ___.___.___.___  27. ______________     ___.___.___.___
12. ______________     ___.___.___.___  28. ______________     ___.___.___.___
13. ______________     ___.___.___.___  29. ______________     ___.___.___.___
14. ______________     ___.___.___.___  30. ______________     ___.___.___.___
15. ______________     ___.___.___.___  31. ______________     ___.___.___.___
16. ______________     ___.___.___.___  32. ______________     ___.___.___.___
```

Name

Identifies the name of the adjunct or server/switch node.

Valid entries	Usage
1 to 15 alpha-numeric characters	Used as a label for the associated IP address. The node names must be unique for each server/switch. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

IP Address

The IP address for the node named in the previous field.

Note:

If you are using the Converged Communications Server for SIP Enablement Services (SES) Instant Messaging, enter the IP address for the SIP Enablement Services (SES) Proxy Server for your network.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 0 to 255)	A unique IP address is assigned to each port on any IP device that is used for a connection. See the *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

IP-Options System Parameters

Field descriptions for page 1

Figure 394: IP-Options System Parameters screen

```
display system-parameters ip-options                           Page 1 of x

                        IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                    Packet Loss (%)     High: 40       Low: 15
                    Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10

 RTCP MONITOR SERVER
                   Default Server IP Address:
                         Default Server Port: 5005
            Default RTCP Report Period(secs): 5

AUTOMATIC TRACE ROUTE ON
            Link Failure? y


 H.248 MEDIA GATEWAY                H.323. IP ENDPOINT
 Link Loss Delay Timer (min):5      Link Loss Delay Timer (min):
                                      Primary Search Time (sec):
                              Periodic Registration Timer (min):
```

IP MEDIA PACKET PERFORMANCE THRESHOLDS

Number of Pings Per Measurement Interval

Specifies the number of test pings that comprise a measurement from which the performance values (delay and loss) are calculated.

Valid entries	Usage
10 to **100**	Enter the number. Default is **10**.

Packet Loss (%)

Specifies thresholds to be applied to packet loss rates (as measured by ping) for determining activation/deactivation of signaling group bypass.

High

Valid entries	Usage
0 to **100**	This value cannot be less than the minimum value. Default is **40**.

Low

Valid entries	Usage
0 to **100**	This value cannot be more than the maximum value. Default is **15**.

Ping Test Interval (sec)

Specifies the time between performance test pings for each testable signaling group.

Valid entries	Usage
10 to **999**	Enter the time. Default is **20**.

Roundtrip Propagation Delay (ms)

Specifies thresholds to be applied to roundtrip packet propagation delays (as measured by ping) for use in activating or clearing signaling group bypass.

High

Valid entries	Usage
10 to **9999**	This value cannot be less than the minimum value. Default is **800**.

Low

Valid entries	Usage
10 to **9999**	This value cannot be more than the maximum value. Default is **400**.

RTCP MONITOR SERVER

Default RTCP Report Period (secs)

In conjunction with the IP address and server port, this value tells the IP telephones, IP softphones and VoIP media modules how often to send the information (RTCP packets) to the RTCP server

.

Valid entries	Usage
5 to **99**	Enter the desired number of seconds.

Default Server IP Address

The default server IP address that can be utilized by the **IP Network Region** screen for each administered region.

Valid entries	Usage
0 to **255** in *nnn.nnn.nnn.nnn* format	A unique IP address is assigned to each port on any IP device that is used for a connection.

Default Server Port

The RTCP monitor is a separate computer that receives RTCP packets from many devices. Avaya Communication Manager pushes these values to IP telephones, IP softphones and VoIP media modules, such that they know where to send the data. The IP address is that of the

RTCP server. The server port is the TCP/IP port of that RTCP server where the information should be sent.

Valid entries	Usage
1 to **65535**	Enter the port being used as the RTCP monitor. Default is **5005**. **Note:** You can also change the RTCP monitor server port setting from the default of **5005** for an individual network region by entering **n** in the **Use Default Server Parameters** field in the RTCP Monitor Server section of the **IP Network Region** screen. When you enter **n**, additional fields appear for entering alternative server parameters.

Automatic Trace Route on Link Failure

In order to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its port networks, media gateways, or IP trunks is lost.

Valid entries	Usage
y	Enter **y** to turn the automatic trace route command feature on.
n	Enter **n** to turn the automatic trace route command feature off.

Note:

If you disable the feature, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are launched or logged (the link failure buffer is cleared).

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS

Announcement Server IP Address

Identifies the IP address of the Announcement Server.

Valid entries	Usage
0 to **255**	A unique IP address is assigned to each port on any IP device that is used for a connection.

Announcement Storage Path Name

Indicates the path name on the Announcement Server where the announcements are stored.

Valid entries	Usage
Up to 40 characters or blank	Enter the directory path name where announcements are stored.

Login

Indicates the login to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a login up to 10 characters.

Password

Indicates the password to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1 to 10 characters or blank	Enter a password up to 10 characters.

H.248 MEDIA GATEWAY

Link Loss Delay Timeout (minutes)

This field is to assist with the H.248 link bounce recovery mechanism of the Avaya G700 Media Gateway; specifically, to prevent the call controller from removing all boards and ports prematurely in response to a link bounce.

Valid entries	Usage
1 through **30**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**

H.323 IP ENDPOINT

Link Loss Delay Timer (minutes)

This timer specifies how long the Communication Manager server preserves registration and any stable calls that might exist on the endpoint after it has lost the call signaling channel to the endpoint. If the endpoint does not re-establish connection within this period, Communication Manager tears down the registration and calls (if any) of the endpoint. This timer does not apply to soft IP endpoints operating in telecommuter mode.

Valid entries	Usage
1 to **60**	Enter the number of minutes to delay the reaction of the call controller to a link bounce. Default is **5**.

Periodic Registration Timer (min)

This timer is started when an IP telephone registration is taken over by another IP endpoint. When the timer expires, the telephone tries to reregister with the server. Default timer value is dependent on the number of unsuccessful periodic registration attempts. As long as the RRJ error message continues to be “Extension in Use,” the endpoint continues to attempt registration with the current gatekeeper address. Sample field values apply unless the endpoint is interrupted, such as by power loss, or the user takes manual action to override this automatic process:

- 20 means once every 20 minutes for two hours, then once an hour for 24 hours, then once every 24 hours continually.
- 60 means once an hour for two hours, then once an hour for 24 hours, then once every 24 hours continually.

Valid entries	Usage
1 to **60**	Enter the number of minutes before an IP telephone registration is taken over by another IP endpoint attempts to reregister with the server. Default is **60**.

Primary Search Time (seconds)

While the telephone is hung-up, this is the maximum time period that the IP endpoint expends attempting to register with its current Communication Manager server. The need for this timer arises in situations where the current Communication Manager server might have a large number of Control Lan (C-LAN) circuit packs. this timer allows the customer to specify the maximum time that an IP endpoint spends on trying to connect to the Control Lan (C-LAN) circuit packs before going to an LSP.

While the IP telephone’s receiver is lifted, the endpoint continues trying to re-establish connection with the current server until the call ends.

Valid entries	Usage
5 to **3600**	Enter the number of seconds an IP endpoint spends on trying to connect to the C-LAN circuit packs before going to an LSP. Default is **75**.

Field descriptions for page 2

Figure 395: IP-Options System Parameters screen

```
change system-parameters ip-options                            Page 2 of x

                        IP-OPTIONS SYSTEM PARAMETERS

Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold?

IP DTMF TRANSMISSION MODE
     Intra-System IP DTMF Transmission Mode: in-band-g711
                       Inter-System IP DTMF: See Signaling Group Forms

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS
  Enable Detection and Alarms?
```

Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold

Valid entries	Usage
y/n	A **y** entry indicates that G.711 is used for intra-switch Music-On-Hold. Default is **n**.

IP DTMF TRANSMISSION MODE

Intra-System IP DTMF Transmission Mode

Enter the appropriate IP transmission mode.

Valid entries	Usage
in-band	DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band.
rtp-payload	Initially, support for SIP Enablement Services (SES) trunks requires the entry of **rtp-payload**.

Inter-System IP DTMF Transmission Mode

See the **DTMF over IP** field on the **Signaling Group** screen.

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS

Enable Detection and Alarms

Enable or disable the hyperactive media gateway registration feature. Default is **n**.

Valid entries	Usage
y/n	Enter **y** to enable the hyperactive media gateway registration feature.

Parameters for Media Gateway Alarms: Hyperactive Registration Window (minutes)

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the **IP-Options System Parameters** screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **15**	Time in minutes for checking hyperactive media gateway registrations. Default is 4 minutes.

Number of Registrations within the Window

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the **IP-Options System Parameters** screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **19**	Number of registrations that occur within the hyperactivity window for generating a Gateway alarm. Default is 3.

Parameters for Network Region Registration (NR-REG) Alarms: % of Gateways in Network Region with Hyperactive Registration Alarms

This field appears when, in the DETECTION AND ALARMING OF HYPERACTIVE MEDIA GATEWAY REGISTRATIONS section of the **IP-Options System Parameters** screen, **Feature Enabled** is **y**.

Valid entries	Usage
1 to **99**	Percent of Gateways within an ip-network region that should be alarmed before an IP-Registration alarm is generated. Default is 80%.

Field descriptions for page 3

Use this screen to administer SNMP station parameters and services dialpad parameters. Applicable terminal types include: 4601, 4602, 4610, 4620, 4621, 4622, 4625, 96xx, or 16xx.

Figure 396: IP-Options System Parameters screen

```
change system-parameters ip-options                          Page 3 of x

                       IP-OPTIONS SYSTEM PARAMETERS

SNMP STATION PARAMETERS
   Download Flag? n
Community String:

SOURCE ADDRESSES:
1.                                    4.
2.                                    5.
3.                                    6.

SERVICES DIALPAD PARAMETERS:
Download Flag? n
     Password: *
```

SNMP STATION PARAMETERS

Community String

The SNMP Community String is used by IP endpoints to determine whether the terminal allows receipt of SNMP queries, and if so, with what "password." If the SNMP community string is null, the terminal ignores all incoming SNMP messages. Otherwise, the community string must be present in the incoming SNMP message for the Terminal to act on that message (subject to other considerations, such as the SNMP Source Address).

Valid entries	Usage
1 to 32 ASCII characters, or blank	Default is NULL (string of zero length). If Community String is null, the terminal ignores all incoming 20 SNMP messages.

Download Flag

Valid entries	Usage
y/n	Determines whether the SNMP parameters are downloaded to the terminals or not. If set to **n**, the Community String and associated IP Addresses are NOT downloaded to terminals. If set to **y**, Community String and associated IP Addresses are downloaded to terminals.Default is **n**.

Source Addresses

The SNMP Source IP Address(es) are used to "validate" the source of an SNMP message. If the SNMP Source Address list is null, the Terminal will respond to any valid SNMP message (where "valid" means the appropriate SNMP community string is properly included). Otherwise, the Terminal will respond to valid SNMP messages only if the IP Source Address of the query matches an address in the SNMP Source Address list.

Valid entries	Usage
Valid Node Name	Enter up to 6 Node Names. Node Names map to proper IP Addresses on the **IP Node Names** screen. An IP address of 0.0.0.0 is a valid address. If you want to remove a node name from the list, you must make sure that the node name is not being used any place in the administration system.

SERVICES DIALPAD PARAMETERS

Download Flag

Valid entries	Usage
y/n	Determines whether the administered Password is downloaded to the terminals or not. If set to **n**, the administered Password is NOT downloaded to terminals. If set to **y**, the administered Password is downloaded to terminals. Default is **n**.

Password

Valid entries	Usage
up to 7 digits (1 to 9), or blank	Enter a password. The Craft Procedures Password is used as part of the Craft Procedures (also called "Local Procedures") that allow a technician to go to an IP Terminal, and modify individual parameters on that specific Terminal (such as the Terminal's IP address, Ethernet interface speed, etc.). The Craft Procedures Password must be entered on the dialpad in the applicable manner, for the technician to have access to the Craft Procedures. Default is 27238 (craft).

IP Routing

Figure 397: IP Routing screen

```
change ip-route n                                              Page 1 of x

                                IP ROUTING

    Route Number:
Destination Node:
    Network Bits:      Subnet Mask:
         Gateway:
           Board:
          Metric:
```

Field descriptions for page 1

Board

Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module

Destination Node

The node name of the final destination for this connection.

Valid entries	Usage
The name previously entered on the **IP Node Names** screen.	Enter the name of the final destination node of the IP route for this connection.

Gateway

The node name of the first intermediate node.

Valid entries	Usage
A name previously entered on the **IP Node Names** screen and is either a port on the CLAN circuit pack or is identified as a Destination Node on another IP route.	If there are one or more intermediate nodes, the first intermediate node is the Gateway. If there are no intermediate nodes between the local and remote CLAN ports for this connection, the Gateway is the local CLAN port.

Metric

Valid entries	Usage
0 or **1**	Enter **1** on a server that has more than one CLAN circuit pack installed.

Network Bits

This field is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
0 to **32**	Enter the number of Network Bits that corresponds to the Subnet Mask for the IP Route. If you skip this field and enter a Subnet Mask, the system automatically sets the corresponding Network Bits value. For example, if you set this field to **24**, then the system sets the **Subnet Mask** field to **255.255.255.0**.

Route Number

Identifies the IP route.

Valid entries	Usage
1 to **400**	Enter the number of the IP route you want to add or change, or enter **n** for the next available number.

Subnet Mask

Valid entries	Usage
255, 254, 252, 248, 240, 224, 192, 128, 0	Enter the subnet mask associated with this IP route. If you enter a value in the **Network Bits** field, the system automatically completes this field with the corresponding Subnet Mask. For example, if you set this field to **255.255.255.0**, then the system sets the **Network Bits** field to **24**.

There is one-to-one mapping between the **Network Bits** and the **Subnet Mask** fields; entering a value in one field uniquely determines the other field. A list of Subnet Mask addresses and their corresponding Network Bits are shown in Table 18.

Table 18: Network Bits and Subnet Mask fields

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	/ 0
1	128.0.0.0	2,147,483,646	/ 1
2	192.0.0.0	1,073,741,822	/ 2
3	224.0.0.0	536,870,910	/ 3
4	240.0.0.0	268,435,454	/ 4
5	248.0.0.0	134,217,726	/ 5
6	252.0.0.0	67,108,862	/ 6
7	254.0.0.0	33,554,430	/ 7
8	255.0.0.0	16,777,214	/ 8
9	255.128.0.0	8,388,606	/ 9
10	255.192.0.0	4,194,302	/ 10
11	255.224.0.0	2,097,150	/ 11
12	255.240.0.0	1,048,574	/ 12
13	255.248.0.0	524,286	/ 13
14	255.252.0.0	262,142	/ 14
15	255.254.0.0	131,070	/ 15
16	255.255.0.0	65,534	/ 16
17	255.255.128.0	32,766	/ 17
18	255.255.192.0	16,382	/ 18
19	255.255.224.0	8,190	/ 19
20	255.255.240.0	4,094	/ 20
21	255.255.248.0	2,046	/ 21
22	255.255.252.0	1,022	/ 22
			1 of 2

Table 18: Network Bits and Subnet Mask fields (continued)

Network Bits	Subnet Mask	Number of Hosts	Network Type
23	255.255.254.0	510	/ 23
24	255.255.255.0	254	/ 24
25	255.255.255.128	126	/ 25
26	255.255.255.192	62	/ 26
27	255.255.255.224	30	/ 27
28	255.255.255.240	14	/ 28
29	255.255.255.248	6	/ 29
30	255.255.255.252	2	/ 30
31	255.255.255.254	1	/31
	255.255.255.255	0	/32
			2 of 2

Note:

For the **Network Bits** and **Subnet Mask** fields, if you put a value into either field and then press **Enter** or **Tab** to move the cursor to another field, the other field gets populated automatically with a value corresponding to the one you just entered.

IP Server Interface (IPSI) Administration

Use this screen to add a TN2312 IPSI (IP Server Interface) circuit pack.The Avaya S8XXX Server uses the IP Server Interface (IPSI) to control port networks and provide tone, clock, and call classification services. The IPSI board connects to the control network by way of Ethernet.

Field descriptions for page 1

Figure 398: IP Server Interface (IPSI) Administration screen

```
add ipserver-interface n

        IP SERVER INTERFACE (IPSI) ADMINISTRATION - PORT NETWORK 2

                              IP Control? y            Socket Encryption? y
 Ignore Connectivity in Server Arbitration?                    Enable QoS? y
                          Administer secondary ip server interface board?


Primary IPSI                                                QoS Parameters
---------------                                            ---------------
Location: 1A02                                        Call Control 802.1p: 4
    Host: ipsi-A01a                                 Call Control DiffServ: 42
 DHCP ID: ipsi-A01a

Secondary IPSI
--------------
Location: 1B01
    Host: ipsi-A01b
 DHCP ID: ipsi-A01b
```

Administer secondary ip server interface board

Valid entries	Usage
y/n	Enter **y** to assign a secondary IPSI board.

Enable QoS

Valid entries	Usage
y/n	Enter **y** to turn on quality of service (QoS) from the Avaya S8XXX Server to the IPSI link. If you enable QoS for the control network, also enable it from the Web interface using **Server Configuration and Upgrades > Configure Server** and on each IPSI interface.

Ignore Connectivity in Server Arbitration

Valid entries	Usage
y/n	Default is **n**.

IP Control

Use this field to administer IP control of port networks.

Note:
In Phase 1 of the S8400, this field is display-only and is set to **y**. This is because, in phase 1, the S8400 is a single port network. The IPSI functionality must therefore be turned on to support the port network. In phase 2 of the S8400, when duplication will be supported, this restriction will be removed.

Valid entries	Usage
y	All port networks have an IPSI that provides control. ● display-only, if **IP-PNC** is **y** on the **System Parameters Customer-Options (Optional Features)** screen ● A DS1 Converter (DS1C) circuit pack cannot be added to a port network when **IP Control** is **y**
n	This IPSI is used only for Tone Clock/Tone Detector functions ● remaining fields on this screen do not appear when **IP Control** is **n** and **IP-PNC** is **n** on the **System Parameters Customer-Options (Optional Features)** screen ● **n** when the port network contains a DS1 Converter (DS1C) circuit pack

Socket Encryption

Valid entries	Usage
y/n	Enter **y** to turn on socket encryption for the Avaya S8XXX Server and IPSI link.

Primary IPSI

DHCP ID

Valid entries	Usage
characters/digits	Enter the DHCP client identifier.

Host

Valid entries	Usage
DHCP ID, alias name, or IP address	Enter the name of the host machine.

Location

Valid entries	Usage
cabinet (**1** to **64**); carrier (**A** to **E**), slot(**1** to **20**);OR gateway(**1** to **250**), module(**V1** to **V9**)	Enter the IPSI board location.

QoS Parameters

Call Control 802.1p

Valid entries	Usage
0 to **7**	Enter the call priority setting.

Call Control DiffServ

Valid entries	Usage
0 to **63**	Enter the DiffServ code point (DSCP).

Secondary IPSI

DHCP ID

Valid entries	Usage
characters/digits	Enter the DHCP client identifier for the secondary IPSI.

Host

Valid entries	Usage
DHCP ID, alias name, or IP address	Enter the name of the host machine for the secondary IPSI.

Location

Valid entries	Usage
cabinet (**1** to **64**); carrier (**A** to **E**), slot (**1** to **20**):OR gateway (**1** to **250**), module (**V1** to **V9**)	Enter the board location for the secondary IPSI.

IP Services

The **IP Services** screen is used to administer the connectivity for various adjuncts.

Note:

You cannot remove a service from this screen if that service has overrides defined on the **Survivable Processor** screen. For more information, see Setting up Processor Ethernet on page 593.

Field descriptions for page 1

Figure 399: IP Services screen

```
change ip-services                                              Page 1 of X

                                IP SERVICES
Service       Enabled     Local       Local          Remote          Remote
  Type                    Node        Port           Node            Port
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
___________     _       __________   ____    ___________________   _____
```

Enabled

This field appears when **Service Type** is **AESVCS** or **SAT**. Controls whether the IP Service specified under **Service Type** is enabled.

Valid entries	Usage
y	Enter **y** to enable this IP service.
n	This IP service is disabled.

Local Node

Specify the node name for the port.

Valid entries	Usage
Node names as defined on the **IP Node Names** screen.	If the link is administered for services over the Control Lan (C-LAN) circuit pack, enter a node name defined on the **IP Node Names** screen.
procr	Enter **procr** to use the Communication Manager's Processor Ethernet interface for adjunct connectivity.

Local Port

Specify the originating port number.

Valid entries	Usage
5000 to **9999**	**5111** to **5117** for SAT applications **5678** for ASAI
0	For client applications, defaults to **0**.

Remote Node

Specify the server/switch at the far end of the link for SAT. The remote node should not be defined as a link on the **IP Interface** or **Data Module** screens.

Valid entries	Usage
Node name as defined on the IP Node Names screen	For SAT, use a node name to provide added security.
any	Use any available node.

Remote Port

Specify the port number of the destination.

Valid entries	Usage
5000 to **64500**	Use if this service is a client application, such as CDR or PMS. This must match the port administered on the adjunct, PC or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

Defines the service provided.

Valid entries	Usage
AESVCS	AE Services.
CBC	Reserves the trunk for outgoing use only to enhance Network Call Redirection.
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.
PMS	Property Management System.
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.
	1 of 2

Valid entries	Usage
SAT	System administration terminal.
SYS_PRINT	Use this to connect the system printer over a TCP/IP link.
	2 of 2

IP Services screen - Session Layer Timers page

Use this screen to enable reliable protocol for TCP/IP links, and to establish other session-layer parameters. This screen only appears if you enter CDR1, CDR2, PMS_JOURNAL, or PMS_LOG in the **Service Type** field on page 1 or 2.

Figure 400: IP Services screen - Session Layer Timer page

```
change ip-services                                            Page   3 of   x

                                SESSION LAYER TIMERS
Service   Reliable   Packet Resp    Session Connect    SPDU    Connectivity
 Type     Protocol      Timer        Message Cntr      Cntr       Timer
CDR1         y         3                   1             1           1
```

Connectivity Timer

Valid entries	Usage
1 to **255**	Indicates the amount of time (in seconds) that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid entries	Usage
1 to **255**	Determines the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Indicates whether you want to use reliable protocol over this link.

Valid entries	Usage
y/n	Use reliable protocol if the adjunct on the far end of the link supports it.

Service Type

A display-only field that identifies the service type for which you are establishing parameters.

Valid entries	Usage
CDR1, CDR2	Used to connect either the primary or secondary CDR device over a TCP/IP link.
PMS_JOURNAL	Used to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Used to connect the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid entries	Usage
1 to **5**	The Session Connect Message counter indicates the number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid entries	Usage
1 to **5**	The Session Protocol Data Unit counter indicates the number of times Communication Manager transmits a unit of protocol data before generating an error.

IP Services screen - AE Services Administration page

This screen appears when the **Service Type** is **AESVCS**. Use this screen to create symbolic name and password pairs for all AE Services servers that are allowed to connect to Communication Manager.

Figure 401: IP Services screen - AE Services Administration page

```
change ip-services                                            Page   4 of   x

                          AE Services Administration

Server ID        AE Services       Password     Enabled      Status
                   Server
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
___________      ____________   ______________   _____
```

AE Services Server

Valid entries	Usage
characters	Enter a valid AE Services Server name. The name must match the AE Services server machine name. Each name must be unique on this screen.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the AE Services server

Password

Valid entries	Usage
12-16 alphanumeric characters; must contain at least one alpha character and one numeric character	Enter a password for future access to this screen.

Server ID

This field is display only.

Valid entries	Usage
1 to **16**	Displays the number assigned to this server.

Status

This field is display only.

Valid entries	Usage
idle	The AE Services server is connected to Communication Manager.
in-use	The AE Services server is not connected to Communication Manager.
blank	No AE Server is administered.

ISDN Network Facilities

See **Network Facilities** screen.

ISDN Numbering Calling Party Number Conversion for Tandem Calls

Tandem calls that route to the public network cannot always provide the correct calling party information, resulting in loss of caller ID information. Communication Manager provides a way of modifying the calling party number on a tandem call that lands in the public network.

Use the **Calling Party Number Conversion for Tandem Calls** screen to administer calling party number formats for tandem calls. To generate a calling party number for the public network, the system compares the incoming calling party number to the sets of calling party lengths, calling party prefixes, and trunk groups. When a match is found, the calling party number is constructed by deleting digits identified in the **Delete** field on the **Calling Party Number Conversion for Tandem Calls** screen, and then inserting the digits specified in the **Insert** field. The numbering format specified in the **Format** field is used to determine the encoding of the NPI and TON fields for the calling party number.

Entries on this screen are only exercised if the **Modify Tandem Calling Number** field on the **ISDN Trunk Group** screen is set to **y**. To access the **Calling Party Number Conversion for Tandem Calls** screen, type `change tandem-calling-party-number`. Press **Enter**.

Field descriptions for page 1

Figure 402: Calling Party Number Conversion for Tandem Calls screen

```
change tandem-calling-party-number                         Page   1 of   x

                    CALLING PARTY NUMBER CONVERSION
                         FOR TANDEM CALLS

        CPN              Trk
  Len   Prefix           Grp(s)   Delete    Insert             Format

   5    22               12-99    1         732852             natl-pub
   7    5381234          2        0         303                lev0-pvt
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
   __   _______________  _____    __        _______________    ________
```

CPN Len

Use the **Calling Party Length** field to enter the total number of digits in the calling party number.

Valid entries	Usage
1 to **15**	Enter a number between 1 and 15 to indicate calling party number length.
blank	This is the default. Leave blank when deleting an entry.

CPN Prefix

Use the **Calling Party Prefix** field to enter the prefix of the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter up to 15 digits to indicate the calling party prefix.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. A specific calling party number digit string match is not required, provided other matching criteria for tandem calling party number modification are met. This is the default.

Trk Grp(s)

Use the **Trunk Groups** field to enter the ISDN trunk group number.

Valid entries	Usage
Valid trunk group or range of group numbers	Enter an ISDN trunk group number, or a range (x to y) of group numbers.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Number** field on the **ISDN Trunk Group** screen is set to **y**. This is the default.

Delete

Use the **Delete Digits** field to enter the digits to delete in modifying the tandem calling party number.

Valid entries	Usage
1 to **15**	Enter a valid number of deleted digits up to 15.
all	Enter all to indicate that all digits are deleted.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. No digits are deleted from the received calling party number. This is the default.

Insert

Use the **Insert Digits** field to enter the digits to insert in modifying the tandem calling party number.

Valid entries	Usage
any combination of digits 0 to 9, up to 15 digits	Enter a valid number of between 1 and 15 to indicate the number of digits to insert.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. The received calling party number is not prefixed with any digits. This is the default.

Number Format

Use the **Number Format** field to enter the numbering format to use in modifying the tandem calling party number.

Valid entries	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	Enter the appropriate format for the tandem calling number.
blank	Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the **ISDN Trunk Group** screen is set to **y**. The numbering format information is not modified.

Note:

The following end validation checks should be performed for this screen:

- The length of the calling party number (the combination of CPN length, deleted digits and inserted digits) cannot exceed 15 digits.
- The number of deleted digits cannot be greater than the CPN length.
- The number of digits entered for the CPN prefix cannot be greater than the CPN length.

If any of the above are true, an error message displays.

ISDN Trunk Group

This screen assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group can contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they can also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

For more information on ISDN trunk groups, see Administering ISDN trunk groups on page 505.

For descriptions of the screens and fields used with non-ISDN trunks, see Trunk Group on page 1669.

Field descriptions for page 1

The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 403: ISDN Trunk Group screen

```
add trunk-group next                                            Page   1 of   xx
                                TRUNK GROUP

Group Number: 1                    Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1        TAC:
   Direction: outgoing        Outgoing Display? n        Carrier Medium:
 Dial Access? n                 Busy Threshold: 255
Queue Length: 0
Service Type:                       Auth Code:            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC:                                 Member Assignment Method:
                                                       Signaling Group:
                                                     Number of Members:
```

Carrier Medium

This field lets you to specify the type of transport medium interface used for the ISDN trunk group. Appears only when the **Group Type** field is **isdn** and, on the **System Parameters Customer-Options (Optional Features)** screen, either the **Async. Transfer Mode (ATM) Trunking** field or the **H.323** field is set to **y**.

Valid entries	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack.
H.323	The trunk is implemented as an H.323 trunk group.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Charge Advice

Use this field to accumulate and access charge information about a call. You already must have set the **CDR Reports** field to **y** or **r** (ring-intvl) before changing this field from its default of **none**. Remember that receiving Advice of Charge during the call (administered as "automatic" or "during-on-request") affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid entries	Usage
none	Enter **none** if you do not want the system to collect Advice of Charge information for this trunk group.
automatic	Enter **automatic** only if your public network sends Advice of Charge information automatically.
end-on-request	Enter **end-on-request** if Avaya Communication Manager must request charge information with each call, and you want to receive only the final call charge.
during-on-request	Enter **during-on-request** if Avaya Communication Manager must request charge information with each call, and you want charges to display during and at the end of a call.

Far End Test Line No.

Specifies the number sent to the far-end's ISDN test line extension. When the `test trunk long` command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. For example, for an ISDN tandem trunk, the far-end test number should be a 7-digit ETN (Electronic Tandem Network) number.

Valid entries	Usage
Up to 15 digits	Enter a code to test signaling channel
blank	

Incoming Calling Number Insert

Valid entries	Usage
Enter up to 15 characters (**0** to **9**), **all**, or blank	Enter the number of digits to insert in the calling party number for all incoming calls on this trunk group.

Incoming Calling Number Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Avaya Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to **pub-unk**.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to **unk-unk** encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:

The values for this field map to the **Type of Numbering (TON)** and **Numbering Plan Identifier (NPI)** values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Member Assignment Method

Appears when **Carrier Medium** on the **Trunk Group** screen is **H.323**.

Valid entries	Usage
manual	Default. Users manually assign trunk members to a signaling group.
auto	The system automatically generates members to a specific signaling group. Entering **Auto** causes the **Signaling Group** and **Number of Members** fields to appear.

Number of Members

Appears when **Carrier Medium** on the **Trunk Group** screen is **H.323** and **Member Assignment Method** is **auto**. Indicates the number of virtual trunk members to be automatically assigned to the signaling group number entered in the **Signaling Group** field.

Valid entries	Usage
0 to **255**	Enter the number of trunks assigned to this signaling group. Default is **0**.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined **Facility Type** of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 (for Avaya DEFINITY Server CSI) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is **public-ntwrk**, Dial Access can be set to **y**.
	1 of 2

Valid entries	Usage
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.
	2 of 2

Signaling Group

Appears when **Carrier Medium** on the **Trunk Group** screen is **H.323** and **Member Assignment Method** is **auto**.

Valid entries	Usage
1 to **650** or blank	Enter assigned h.323 or SIP Enablement Services (SES) signaling group number between **1** and **650**, or blank.

TestCall BCC

Indicates the Bearer Capability Code (BCC) used for the ISDN test call.

Valid entries	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability Information Element (IE) in the SETUP message when generating an ISDN test call. Allowed values are **rest** (restricted) and **unre** (unrestricted).

Note:

ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Testcall Service

Specifies the call-by-call selection for an ISDN test call. Only appears if the **Service Type** field is **cbc**. Valid entries are all of the services listed in Service Type on page 1246, excluding **sddn** or any new **Facility Type** of **0** (feature), **1** (service), or **3** (outgoing) that is defined by users on the **Network Specific Facility Encoding** screen.

Usage Alloc

Appears when the **Service Type** field is **cbc**.

Valid entries	Usage
y/n	Enter **y** to allocate service provided by the trunk group. Use **y** to enhance Network Call Redirection. When you enter **y**, the **CBC Trunk Group Usage Allocation Plans** screen and the **CBC Trunk Group Usage Allocation Plan Assignment Schedule** appear.

Field descriptions for page 2

The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 404: ISDN Trunk Group screen

```
add trunk-group next                                                Page   2 of  x
      Group Type: isdn                  Trunk Type:

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13

              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision -       Out? n
 Answer Supervision Timeout: 0
           Administer Timers? n
```

Administer Timers

This field is displayed for all trunk group types except **cpe**, **h.323**, and **sip**. When this field is **y**, the **Administrable Timers** page is available to administer timer values.

Valid entries	Usage
y/n	Enter **y** to allow administration of timers on this trunk group. For **Group Type isdn**, the default value is **n**. For all other trunk group types, the default is **y**.

Codeset to Send Display

This field defines the codeset for sending the information element for display. The value depends on the type of server/switch to which the user is connected.

Valid entries	Usage
0	CCITT (non-Communication Manager equipment).
6	Any other than CCITT or System 85 R2V4, 4E11.
7	System 85 R2V4, 4E11.

Codeset to Send National IEs

This field defines the codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of server/switch the user is connected to.

Valid entries	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer Avaya S8XXX Server types.

Note:

A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the **Service Type** field is **tandem**. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

Digit Handling (in/out)

This field defines whether overlap receiving and overlap sending features are enabled.

Valid entries	Usage
enbloc/enbloc **enbloc/overlap** **overlap/enbloc** **overlap/overlap**	Set the field to **overlap** when you want overlap receiving or overlap sending. Set to **enbloc** when you do not want these features enabled. The first field value indicates digit receiving and the second value indicates digit sending.

Without overlap receiving or sending enabled, the digits on incoming and outgoing calls are sent enbloc. If the **Digit Handling** field is **overlap/enbloc** or **overlap/overlap**, the following results:

- Incoming Call Handling Treatment table does not appear
- The **Digit Treatment** and **Digits** fields appear
- Warning message indicates that all **Incoming Call Handling** entries are removed when screen is submitted
- When screen is submitted with these values, all **Incoming Call Handling** entries are removed

Group Type

Displays the type of trunk group selected for this field on page 1 of the **Trunk Group** screen. For details, see the field description for the page 1 **Group Type** field.

Max Message Size to Send

Defines Communication Manager's maximum ISDN message size. Currently, the system can receive 260-byte messages. Valid entries are **128**, **244**, **256**, and **260**.

The following table indicates the expected ISDN-PRI message size from several Lucent Technologies and Avaya Inc. products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

Supplementary Service Protocol

Indicates which supplementary service protocol to use for services over this trunk group. Supplementary Service protocols are mutually exclusive.

Valid entries	Usage
a	National
b	ISO/ETSI QSIG Private Network. Also used for SBS signaling trunks.
c	ETSI public network
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC))
e	DCS with Rerouting ● Do not use the **Service Type** field entry of **dmi-mos** or **sddn** with this option. ● Set the Used for DCS field (on page 2) to **y**.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI. Available only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

Trunk Hunt

Avaya Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed.

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:

The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Trunk Type

Displays the type of trunk selected for this field on page 1 of the **Trunk Group** screen. For details, see the field description for the page 1 **Trunk Type (in/out)** field.

Field descriptions for page 3

The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other **Trunk Group** fields, see Trunk Group on page 1669.

Figure 405: ISDN Trunk Group screen

```
add trunk-group next                                          Page 3 of x
                                 TRUNK FEATURES
           ACA Assignment? n            Measured: none____
                                                        Maintenance Tests? y
                                Data Restriction? n
                                       Send Name:

    Abandoned Call Search? n
    Suppress # Outpulsing? n

        Charge Conversion: 1
            Decimal Point: none
          Currency Symbol:
              Charge Type: units
                                       Per Call CPN Blocking Code:
                                     Per Call CPN Unblocking Code:

              Outgoing ANI:                    DS1 Echo Cancellation? n

Apply Local Ringback? n                   US NI Delayed Calling Name Update? _
Show ANSWERED BY on Display? y
                              Network (Japan) Needs Connect Before Disconnect? _
DSN Term?
```

Apply Local Ringback

This field appears for ISDN and H.323 trunk groups when the **Carrier Medium** field is **PRI_BRI**.

Valid entries	Usage
y/n	Enter **y** to provide a local ringback tone to the caller. The local ringback is removed when the call is connected. Default is **n**.

BSR Reply-best DISC Cause Value

Servers running Avaya Communication Manager that are polled as resources in a Best Service Routing application return data to the polling serve in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that your server will return in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group will be dropped before any data is returned to the polling server or switch. This field only appears if the **UUI IE Treatment** field is set to **shared**.

Valid entries	Usage
31 (normal-unspecified)	Enter **31** unless otherwise instructed by Avaya or your network service provider.
17 (user-busy)	
16 (normal-call-clearing)	

CAUTION:
In most cases, this field will be set to the appropriate value during installation. If you need to change it, your network service provider should be able to help you choose an appropriate value. Don't change this field without the assistance of Avaya or your network service provider.

DCS Signaling

Specifies the means used to send the DCS message. This field only appears if the **Used for DCS** field entry is **y** and the **Service Type** field is anything except **dmi-mos** or **sddn**.

Valid entries	Usage
d-chan	Enter for the DCS over ISDN-PRI D-channel feature.

DCS over D-channel is not supported on trunk groups containing ISDN-BRI interfaces.

- Hop Dgt — The Tandem Hop Limitation and QSIG Additional Network Feature Transit Counter (ANF-TC) features provide a counter that reflects the number of switches (that is, the number of hops) that a call has gone through. The counter increments as a call leaves Avaya Communication Manager using tandem facilities. Valid values are **y** and **n**. One or both of the features can be applied to the trunk group depending on the following:
  - If you enter **y** and the **Group Type** field is **tandem** or the **Group Type** field is **isdn** and the **Service Type** field is **tandem**, the Tandem Hop Limitation feature is applied to the trunk group.
  - If you enter **y** and you set the **Group Type** field to **isdn**, set the **Service Type** field to **access**, **dmi-mos**, **public-ntwrk**, **tandem**, **tie**, or any of the craft-defined services allowed in the field. Set the **Supplementary Service Protocol** field to **b** or **d**, then the ANF-TC feature is applied to calls on the trunk group.

  Note:

  The above conditions overlap. If the **Group Type** field is **isdn**, the **Service Type** field is **tandem**, and the **Supplementary Service Protocol** field is **b** or **d**, then both the Tandem Hop Limitation and ANF-TC features are applied to calls on the trunk group.

  - If both features are applied to calls on the trunk group, ANF-TC takes precedence. In situations where Avaya Communication Manager serves as an Incoming or Outgoing Gateway, either feature uses the hop count and transit information provided by the other.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**. Chose the appropriate representation for a decimal point as it will appear on telephone displays. Entering **comma** or **period** in this field divides the charge value by 100.

Note:

On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field does not drive whether a decimal point appears on the calling display. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier. If the received charge contains no decimals, no decimal point is displayed (i.e., the administered decimal point is ignored for charge information received with no decimals). On an upgrade from a QSIG trunk group with the **Decimal Point** field administered as **none**, the field defaults to **period**.

Valid entries	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point.
none	No decimal point is displayed.

Maximum Size of UUI IE Contents

This field appears when the **UUI IE Treatment** field is **shared**.

Valid entries	Usage
32 to **128**	Enter the maximum number of bytes of user information that the network supports.

Modify Tandem Calling Number

This field appears when **Trunk Group Type** is **ISDN**, Direction is either **Outgoing** or **Two-way**, **Carrier Medium** is **PRI/BRI** or **IP**, and **Send Calling Number** is either **y** or **r**. It is used to control whether call processing processes the entries on the **ISDN -Tandem Calling Party Number** screen.

Valid entries	Usage
y/n	Enter **y** to modify the calling party number IE in the format specified on the **ISDN-Tandem Calling Party Number** screen. Default is **n**.

NCA-TSC Trunk Member

Identifies the trunk member whose D-channel will be used to route tandem NCA-TSCs or QSIG CISCs. Value range for this field is from 1 to the maximum number of members per trunk group supported on the server/switch, or blank.

Network Call Redirection

This field is administrable if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** field is **y**, the **ISDN Network Call Redirection** field is **y**, and on the **ISDN Trunk Group** screen, the **Supplementary Service Protocol** field is **a, c,** or **g**.

Whenever the **Supplementary Service Protocol** field is changed, this field resets to **none** to prevent an inadvertent incorrect value.

Following are the allowed settings for TBCT or MCI/Verizon NCT:

	Telcordia TBCT	ANSI-1998 ECT
G3 Version	V11 or later	V11 or later
Customer Options	ISDN-PRI ISDN Network Call Redirection	ISDN PRI ISDN Network Call Redirection
DS1 Country Protocol	1b or 1d	Any, but typically 1a
Trunk Group Supplementary Service Protocol	a	g
Network Call Redirection keyword	Telcordia-TBCT	ANSI Transfer (MCI/Verizon) Enhanced ANSI Transfer Nortel-Transfer (Nortel DMS)
Trunk Group member's signaling group is Network Call Transfer	y	y

Following are the allowed settings for ETSI protocol:

G3 Version	V12 or later
Customer Options	ISDN PRI ISDN Network Call Redirection
Trunk Group Supplementary Service Protocol	c
Network Call Redirection keyword	implicit-etsi-ect explicit-etsi-ect deflect
Trunk Group member's signaling group is Network Call Transfer	y
Call Center Release	8.3 or later

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the Disconnect message.

Numbering Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the Type of Number is national. **Unknown** indicates the Numbering Plan Indicator is unknown and that the Type of Number is unknown. **Private** indicates the Numbering Plan Indicator is PNP and the Type of Number is determined from the **Numbering - Private Format** screen. An entry of **unk-pvt** also determines the **Type of Number** from the **Numbering - Private Format** screen, but the **Numbering Plan Indicator** is unknown.

Outgoing Channel ID Encoding

Appears only if the **Group Type** field is **isdn** and the **Service Type** field is anything except **dmi-mos** or **sddn**. Determines whether to encode the Channel ID IE as preferred or exclusive. Blank is not a valid entry. Defaults are determined as follows:

If the **Group Type** field is **isdn** and the **Used for DCS** field is **y**, default is **exclusive**.

If the **Group Type** field is **isdn** and the **Used for DCS** field is **n**, default is **preferred**.

If the **Group Type** field is not **isdn** or it is **isdn**, but the **Used for DCS** field does not appear, default is **preferred**.

Path Replacement Method

Appears when either the **ISDN-PRI trunk** or the **ISDN-BRI trunk** fields and the **Basic Call Setup** and **Supplementary Services with Rerouting** fields are set to **y** on the **System Parameters Customer-Options (Optional Features)** screen and when the **Supplementary Service Protocol** is either **b** or **e** and the **Group Type** field is **isdn** on the **ISDN Trunk Group** screen.

Valid entries	Usage
better-route	Uses the most economical route, for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Replace Restricted Numbers

Appears when the **Group Type** field is **isdn**. Indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI, PRI, H.323 and SIP trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI, PRI, and H.323 trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

SBS

Appears when the **Local Country Code and International Access Code** fields are administered on the **Feature-Related System-Parameters** screen and when the **Supplementary Service Protocol** is **b** and the **Group Type** field is **isdn** and **Carrier Medium** is **IP** and **Dial Access** is **n** on page 1 of the **ISDN Trunk Group** screen.

Valid entries	Usage
y/n	Enter **y** to enable Separation of Bearer and Signaling (SBS) for the trunk group. The default is **n** (SBS is not enabled).

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are y, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number to be sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted."

When **Send Calling Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Calling Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The **ISDN Numbering - Public/Unknown Format** screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Codeset 6/7 LAI IE

Specifies whether the ISDN trunk should transmit information in Codeset 6/7. If the UUI IE Treatment field is **shared**, then this field should be **n**. Otherwise, the same information will be sent twice and might exceed the message size. Default is **y** for pre-DEFINITY 6.3 compatibility.

Send Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering-Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

When **Send Connected Number** is **n**, an incoming number is not tandemed out again. Similarly, when **Send Connected Number** is **r** (restricted), an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1 Circuit Pack** screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the **ISDN Trunk Group** screen, as this causes display problems.

The **ISDN Numbering - Public/Unknown Format** screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Send Name

Specifies whether the calling/connected/called/busy party's administered name, or the name on a redirected call, is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Avaya Communication Manager, but will be marked "presentation restricted." This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/ busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted. For redirected calls, the value **y** indicates that the name is displayed, while for **n** and **r**, the redirected caller name is not displayed.

When the **Send Name** field in **n**, an incoming name is not tandemed out again if the **Supplementary Service Protocol** field is any value other than **b** (QSIG). Similarly, when **Send Name** is **r** (restricted), an incoming name is marked restricted when it is tandemed out again. However, if the **Supplementary Service Protocol** field is **b** (QSIG), then an incoming name is passed on unchanged and the **Send Name** field is ignored.

Note:

If name information is not administered for the calling station or the connected/ called/busy station, the system sends the extension number in place of the name.

Send UCID

Specifies whether or not the trunk should transmit Universal Call IDs. Valid entries are **y** and **n**.

Send UUI IE

Specifies whether to block sending UUI information on a per trunk group basis. The valid entries are y and **n**.

Show ANSWERED BY on Display

This field appears when the **Group Type** field is **isdn pri/bri** or **sip**. Use this field to administer whether or not the words "ANSWERED BY" are displayed in addition to the connected telephone number on calls over this trunk.

Note:

Based on display language settings for stations, "ANSWERED BY" is translated into and displayed in the appropriate language.

Valid entries	Usage
y	When set to **y**, the words "ANSWERED BY" are displayed in addition to the connected telephone number. This is the default.
n	When set to **n**, only the connected telephone number is displayed. This might be preferred when outgoing calls are over a trunk that might be redirected.

US NI Delayed Calling Name Update

Administrable if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** field is **y**, and on the **Trunk Group** screen, the **Carrier Medium** field is either **PRI/ BRI** or **ATM**, and the **Supplementary Service Protocol** field is **a**. This field provides display updates to the terminating telephone for delayed calling party name provided by the network.

Valid entries	Usage
y	If calling name information is received after the incoming call has been delivered to the terminating telephone, there is a display update. **Note:** BRI trunks do not support this value.
n	If calling name information is received after the incoming call has been delivered to the called telephone, there is no display update to the terminating telephone.

UUI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid entries	Usage
shared	If the trunk is connected to an Avaya DEFINITY 6.3 (or later) server, or an Avaya S8XXX Server.
service-provider	If the trunk is connected to a pre-DEFINITY 6.3 switch, or service provider functionality is desired.

Wideband Support

Note:
This feature is not supported on the DS1 interfaces on H.248 gateways.

Specifies whether Wideband Switching is supported by this trunk group. Valid entries are **y** or **n**. For ISDN trunk groups containing ISDN-BRI interfaces, the only valid entry is **n**. Otherwise you can administer this field only if the **Wideband Switching** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. If set to **y**, the **Wideband Support Options** page appears. All trunk members must be from TN464C (or later) circuit packs. The trunk members that are supported using DS1 modules on the H.248 gateways (G700/G350) do not provide for "n" X 64kbps wideband channel support. Those interfaces only provide for call connections based on a single B-channel.

Note:
Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Field descriptions for QSIG Trunk Group Options page

This fields on this screen appear only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**. The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 406: QSIG Trunk Group Options screen

```
add trunk-group next                                              Page x of y

                        QSIG Trunk Group Options

   TSC Method for Auto Callback? n
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                           SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: iso8859-1
               QSIG Value-Added? n
   QSIG-Value Coverage Encoding: proprietary
```

Character Set for QSIG Name

Use this field to set the character set for transmission of QSIG name data for display. This field appears only when **Group Type** is **isdn**, **Supplementary Service Protocol** is **b**, and **Display Character Set** on the **System Parameters Country-Options** screen is **Roman**. For more information, see Administering displays for QSIG trunks on page 210.

Valid entries	Usage
eurofont	The Roman Eurofont character set. This is the default. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont, Kanafont, or Optrex. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.
iso-8859-1	All data (i.e., characters) in the Name value transmitted over QSIG are converted from Eurofont (Avaya proprietary encoding) to ISO 8859-1. **Note:** ISO 8859-1, more formally known as ISO/IEC 8859-1, or less formally as Latin-1, is part 1 of ISO/IEC 8859, a standard character encoding defined by ISO. It encodes what it refers to as Latin alphabet no. 1, consisting of 191 characters from the Latin script, each encoded as a single 8-bit code value.

Diversion by Reroute

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y	The Diversion by Reroute feature is enabled. Default is **y**.
n	The Diversion by Reroute feature is disabled. Communication Manager will not originate a Diversion/Reroute request over that trunk group, and will reject any Diversion/Reroute request it receives over that trunk group.

Display Forwarding Party Name

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y/n	Enter **y** to display the name of the party who is forwarding the call. Default is **y**.

QSIG Value-Added

Valid entries are **y** and **n**. Provides QSIG-VALU services. This field appears only if the **Value-Added (VALU)** field on the System Parameters Customer-Options (Optional Features) screen is **y**. This field can be set to **y** only if the **Supplementary Service Protocol** field on the **System Parameters Customer-Options (Optional Features)** screen is **b**.

QSIG-Value Coverage Encoding

Use this field to indicate the encoding method to use to encode DL1, DL2, and DL3 extensions. This field appears only when **Group Type** is **isdn**, **Supplementary Service Protocol** is **b**, and **QSIG Value-Added** is **y**.

Valid entries	Usage
proprietary	Communication Manager sends extension information in the normal manner. This is the default.
standard	In addition to normal extension information, Communication Manager also sends the data part (as null) of the extension.

Path Replacement

This field appears only when **Group Type** is **isdn** and **Supplementary Service Protocol** is **b**.

Valid entries	Usage
y	The Path Replacement feature is enabled. Default is **y**.
n	The Path Replacement feature is disabled. Communication Manager will not originate a Path Replacement request over that trunk group, and will reject any Path Replacement request it receives over that trunk group.

Path Replacement Method

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, the **Path Replacement with Retention** is **n**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the **Path Replacement with Retention** field is **y**. Path replacement fails (and the original trunk group is retained) if the **Path Replacement with Retention** field is **n**.

Path Replacement with Retention

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the **Path Replacement Method** field.

TSC Method for Auto Callback

Use this field to control the signaling connection method for the QSIG-TSC when Communication Manager is the terminating or the outgoing gateway PINX. When this field is set to **always-retain**, QSIG Temporary Signaling Connections are always retained for successful call completion activation.

Valid entries	Usage
drop-if-possible	signaling connection is released
always-retain	signaling connection is retained

Field Descriptions for Administrable Timers page

This screen displays only when the **Administer Timers** field on page 2 of the **Trunk Group** screen is **y**. This screen does not display when for trunks of **Group Type cpe** or **sip**.

Note:

If the ISDN trunk group has a **Carrier Medium** value of **H.323**, or it the trunk group has BRI members, then the **Administrable Timers** page is not administrable. In these cases, an error message displays when you attempt to submit the screen.

Figure 407: Administrable Timers for ISDN Trunk Group screen

```
add trunk-group next                                              Page 5 of x
                           ADMINISTRABLE TIMERS







 Programmed Dial Pause(msec): _____



END TO END SIGNALING
    Tone (msec): ____    Pause (msec): 150
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

END TO END SIGNALING

Pause (msec)

THis field is administrable only if the **Trunk Type** field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid telephone.

Tone (msec)

This field appears only if the **Trunk Type** field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid telephone is pressed.

Field descriptions for the Shared UUI Feature Priorities page

The fields in this page show the priorities for each type of information to be forwarded in the Shared UUI. This page appears only on the **ISDN trunk group** screen when all of the following conditions are met:

- The **UUI IE Treatment** field is **shared**.
- The **Supplementary Service Protocol** field is set to anything except **b**.

The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other **Trunk Group** fields, see Trunk Group on page 1669.

Figure 408: Shared UUI Feature Priorities screen

```
add trunk-group next                                             Page y of x
                       SHARED UUI FEATURE PRIORITIES

                     ASAI: 1

        Universal Call ID: 2

MULTI SITE ROUTING (MSR)

              In-VDN Time: 3
                 VDN Name: 4
         Collected Digits: 5
    Other LAI Information: 6
```

Changing the priorities in this screen might affect whether certain information will be sent.These fields are unique to the **ISDN Trunk Group** screen.

ASAI

User information from ASAI. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

In-VDN TIme

Number of seconds the call has spent in vector processing. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call's priority level in its current queue, and the type of interflow. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Universal Call ID

Unique tag to identify each call. Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS). Valid entries are **1** to **6** (**1** is high) and blank. If blank, that field's information is not forwarded.

Field Descriptions for the CBC Trunk Group Usage Allocation page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**. The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 409: CBC Trunk Group Usage Allocation screen

```
add trunk-group next                                               Page y of x
                         CBC TRUNK GROUP USAGE ALLOCATION
Usage Allocation Plan 1    Usage Allocation Plan 2    Usage Allocation Plan 3
                Min# Max#                      Min# Max#                      Min# Max#
Service/Feature Chan Chan  Service/Feature Chan Chan Service/Feature  Chan Chan

_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
```

The **CBC Trunk Group Usage Allocation** screen sets a minimum and maximum number of members for up to ten different Services/Features for up to three different Usage Allocation Plans (1 to 3). See "Call-by-call Service Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for a detailed description of Usage Allocation Plans.

Max# Chan

Indicates the maximum number of members of a ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. This field must be completed if a Service/Feature has been entered in the **Incoming Call Handling Treatment Table** screen. Valid values are **0** to **99** or blank.

Min# Chan

Indicates the minimum number of members of an ISDN trunk group with a **Service Type** field of **cbc** that a particular Service/Feature can use at any given time. The sum of the minimum number of members for all Service/Features must not exceed the total number of members of the trunk group. Valid values are **0** to **99** or blank.

Service/Feature

Specifies the ISDN Services/Features that can be requested at call setup time when using this trunk group. See the Service Type field description for a list of predefined Services/Features that can be received on a call by call basis. In addition, user-defined service types can be used. Any user-defined **Facility Type** of **0** (feature) or **1** (service), **2** (incoming), or **3** (outgoing) on the **Network Facilities** screen is allowed.See the description of the **Network Facilities** screen for details. The identifier **other** is used for all Services/Features not explicitly specified.

Field descriptions for the CBC Service Trunk Group Allocation Plan Assignment Schedule page

Appears when the **Service Type** field is **cbc** and the **Usage Alloc** field is **y**. The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 410: CBC Service Trunk Group Allocation Plan Assignment Schedule screen

```
add trunk-group next                                          Page y of x
            CBC SERVICE TRUNK GROUP ALLOCATION PLAN ASSIGNMENT SCHEDULE

  Usage Method:

         Fixed? y      Allocation Plan Number: 1
     Scheduled? n


Usage Allocation Plan Activation Schedule:

       Act  Plan    Act  Plan    Act  Plan    Act  Plan    Act  Plan    Act  Plan
        Time  #      Time  #      Time  #      Time  #      Time  #      Time  #
Sun     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Mon     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Tue     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Wed     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Thu     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Fri     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Sat     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
```

The **CBC Service Trunk Group Allocation Plan Assignment Schedule** screen provides for administering a fixed schedule or administering a schedule that can change up to 6 times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan will be in use at any given time.

Act Time

Indicates the time the usage allocation plan administered in the next field (**Plan #**) will become effective. Enter the time in military time. There must be at least one entry per day. Valid entries are **00:00** through **23:59**.

Allocation Plan Number

Specifies the CBC Trunk Allocation Plan (1 through 3) that is in effect if a fixed usage method has been selected. This field must be assigned if the **Fixed** field is **y**. Valid entries are **1** to **3** or blank.

Fixed

Indicates whether the allocation plan will be fixed. If **y** is entered, the plan number entered in the **Allocation Plan Number** field will be enabled.

Plan

Specifies the number of the usage allocation plan that will be in effect from the activation time until the activation time of the next scheduled plan change. Valid entries are **1** to **3** or blank.

Scheduled

Indicates whether or not the allocation plans will be in effect according to the schedule found on this page. If **y** is entered in this field then there must be at least one entry in the schedule.

Field descriptions for the Wideband Support Options page

The **Wideband Support Options** screen appears immediately before the trunk group member pages. The actual page number will vary.The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other Trunk Group fields, see Trunk Group on page 1669.

Figure 411: Wideband Support Options screen

```
add trunk-group next                                              Page y of x
                      Wideband Support Options

                              H0? n
                             H11? n
                             H12? n
                           NxDS0? y     Contiguous? n
```

Note:

All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility. Also, all trunk members in an ISDN trunk group with the **Wideband Support** field set to **y** must be from a TN464C (or later) circuit pack.

H0

Enter **y** to specify the ISDN information transfer rate for 384-kbps of data, which is comprised of six B-channels. When a trunk group is administered to support H0, the trunk/hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

Enter **y** to specify the ISDN information transfer rate for 1536-kbps of data, which is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk/hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

Enter **y** to specify the ISDN information transfer rate for 1920-kbps of data, which is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk/hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

Contiguous

Specifies whether or not to hunt contiguous NXDS0 channels. This field only appears if the **NxDS0** field is **y**.

The trunk/hunt algorithm to satisfy an NXDS0 call is as follows:

- Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).

Note:
H0 and NXDS0 "floating" scheme cannot both be **y**.

- Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints such as contiguity of B-channels or fixed starting points.

NxDS0

Enter **y** to specify the "N by DS-zero" multi-rate service.

Field Descriptions for the Group Member Assignments page

The field descriptions which follow are for fields that are unique to the **ISDN Trunk Group** screen. For descriptions of other **Trunk Group** fields, see Trunk Group on page 1669.

Figure 412: ISDN Group Member Assignments screen

```
add trunk-group next                                                 Page y of x
                               TRUNK GROUP
                               Administered Members(min/max): xxx/yyy
GROUP MEMBER ASSIGNMENTS          Total Administered Members: xxx

       Port     Code     Sfx  Name         Night    Sig Grp
    1: _______  _____     _   __________   _____      ___
    2: _______  _____     _   __________   _____      ___
    3: _______  _____     _   __________   _____      ___
    4: _______  _____     _   __________   _____      ___
    5: _______  _____     _   __________   _____      ___
    6: _______  _____     _   __________   _____      ___
    7: _______  _____     _   __________   _____      ___
    8: _______  _____     _   __________   _____      ___
    9: _______  _____     _   __________   _____      ___
   10: _______  _____     _   __________   _____      ___
   11: _______  _____     _   __________   _____      ___
   12: _______  _____     _   __________   _____      ___
   13: _______  _____     _   __________   _____      ___
   14: _______  _____     _   __________   _____      ___
   15: _______  _____     _   __________   _____      ___
```

The total number of pages that make up the **Trunk Group** screen, and the first page of Group Member Assignments, will vary depending on whether the CBC and Wideband Support pages display.

Note:

When supporting DCS, Member Number Assignments must be the same between nodes (Member #1 must be Member #1 at the far-end trunk group).

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1's port number is 01A1002, then B channel 2's port number is 01A1018.

When using ISDN-PRI interfaces, the port number will be the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

Note:

When administering analog trunks connected to a TIM518, physical ports 17-24 are administered as ports 9 to 16 in Communication Manager.

Sig Grp

This field appears when the **Group Type** field is **isdn-pri**. Enter the signaling group of this trunk group member. Valid entries are from **1** to **650**, and must be configured for IP group members. If you administer a port that resides on a DS1 board and that DS1 board belongs to one and only one signaling group, you can leave the **Signaling Group** column blank. Then, when you submit the screen, the appropriate default signaling group number is inserted by Communication Manager. If a DS1 board is assigned to more than one signaling group, then you must enter a signaling group number. You must enter a signaling group if the port is entered as **IP**. A trunk group can contain members from different signaling groups.

Related topics

See "ISDN Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information Integrated Services Digital Network trunks.

See "DS1 Trunk Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

ISDN-BRI Trunk Circuit Pack

This screen administers an ISDN-BRI circuit pack. See *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207, for information on the maximum number of ISDN-BRI circuit packs that you can administer.

Figure 413: BRI Trunk screen (using a TN2185 circuit pack)

```
change bri-trunk-board                                          Page 1 of x
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                              Name: ______________
  Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
 T3 Timer Length (sec): 15_                 Termination Type: TE


Port  Interface   Side  Cntry/Peer TEI               Synch   Layer 1  Detect
                         Protocol                    Source? Stable?  Slips?
  1:  user_______         12__     0__                  n       n       n
  2:  network____         etsi     0__                  y       y       y
  3:  user_______         2___     auto                 n       y       n
  4:  peer-slave_  b      QSIG     0__                  y       y       n
  5:  peer-master  a      QSIG     auto                 n       n       n
  6:  ___________         ____     0__                  n       y       n
  7.  ___________         ____     0__                  n       y       n
  8:  ___________         ____     0__                  n       y       n
```

Field descriptions for page 1 (with a TN2185 circuit pack)

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25** **etsi**	When this field is **10**, **12**, **13**, or **etsi**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen. When the **Cntry/Peer Protocol** field is set **10**, **12**, **13**, or **etsi**, set the **Protocol Version** field to **b**. For all other administered values, the **Protocol Version** sets to **a**.
QSIG	When the **Interface** field is **peer-slave** or **peer-master**, this field must be **QSIG**. The choice **QSIG** is valid only when the Interface field is **peer-slave**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value

DCP/Analog Bearer Capability

Valid entries	Usage
3.1kHz **speech**	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.

Detect Slips

Valid entries	Usage
y/n	Tells maintenance software whether slips reported by the BRI port should be logged.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN2185. You can enter **peer-slave** only if the QSIG Basic Call Setup feature is enabled
user	
peer-master	
peer-slave	

Interface Companding

Valid entries	Usage
a-law	Indicates the companding algorithm expected by the system at the far end.
mu-law	

Layer 1 Stable

Valid entries	Usage
y	Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. Only the TN2185 can be set to **n**.
n	

Location

This is a display-only field. It shows the TN2185 circuit pack location (PPCSS)

Name

Valid entries	Usage
1 to 15 alpha-numeric characters	This name is used to identify the circuit pack. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Side

Valid entries	Usage
a	Determines how glare conditions are handled when Interface field is **peer-slave**.
b	

Synch Source

The **Synch Source** field applies only for TN2185 boards.

Note:
For MM720 and MM722 bri media modules, the **Synch Source** field does not appear. For the MM720 and MM722, this parameter is configured using the gateway CLI.

Valid entries	Usage
y	When set to **y**, allows a TN2185 board displayed on the **Synchronization Plan** screen to be entered as the Primary or Secondary synchronization source, if at least one of the ports on that board has **Synch Source?** enabled. Only those ports marked **y** are candidates for clock synchronization with the far-end network.
n	

T3 Timer Length (sec)

Valid entries	Usage
1 to **127**	Tells the TE side how long to wait, in seconds, for an inactive Layer 1 to become active.

TEI

Valid entries	Usage
auto	TEI is assigned automatically by the network.
0	TEI is fixed.

Termination Type

When a MM720 media module is used as a trunk interface, and the MM720 supports both Line side and Trunk side of BRI, use this field to indicate whether the media module is to operate in Terminal or Network termination mode.

Note:

On a MM720 that can function only as a BRI Trunk Media Module (i.e., MM720 without the firmware upgrade that supports both line side and trunk side of BRI), this field defaults to **TE** and is display-only.

Valid entries	Usage
TE	Terminal Endpoint termination. The MM720 provides the TE side of the BRI interface. This is the default.
NT	Network Termination. The MM720 provides the NT side of the BRI interface.

Field descriptions for page 1 (TN556B or TN2198 circuit pack)

Figure 414: BRI Trunk screen (with a TN556B or TN2198 circuit pack)

```
change bri-trunk-board                                         Page 1 of x
                        ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                   Name:  ______________
   Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
                                             Termination Type: TE

Port  Interface   Side  Cntry/Peer TEI
                        Protocol
  1:  network__           12__      0__
  2:  network__           etsi      0__
  3:  network__           2___      auto
  4:  peer-master  b      QSIG      0__
  5:  peer-master  a      QSIG      auto
  6:  __________          ____      0__
  7.  __________          ____      0__
  8:  __________          ____      0__
  9:  __________          ____      0__
 10:  __________          ____      0__
 11:  __________          ____      0__
 12:  __________          ____      0__
```

The following field descriptions are unique to the **ISDN-BRI Circuit Pack** screen with a TN556B or TN2198 circuit pack. The following fields do not display with a TN556B or TN2198 circuit pack:

- **T3 Timer Length (sec)**
- **Synch Source**
- **Layer 1 Stable**
- **Detect Slips**

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, or **13**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen.
etsi	When this field is **etsi**, the **Protocol Version** field is equivalent to **b** on the **DS1 Circuit Pack** screen.
QSIG	When the **Interface** field is **peer-master**, this field must be **QSIG**.
blank	You cannot leave this field blank if the **Interface** field is set to a valid, non-blank value.

Interface

Valid entries	Usage
network	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN556B. You can enter **peer-master** only if the QSIG Basic Call Setup feature is enabled
peer-master	

Side

Valid entries	Usage
a	Determines how glare conditions are handled when **Interface** field is peer-slave. This field is not administrable when the Interface field is **network**.
b	

Field descriptions for page 2

Note:

If administering a TN2185 circuit pack, 8 ports appear; otherwise, 12 ports appear.

Figure 415: BRI Trunk screen - Page 2 (using a TN2185 circuit pack)

```
change bri-trunk-board                                    Page 2 of x
                         ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt    SPID      Endpt  Max
       Message   Test? Init?               ID                 ID   NCA TSC
   1:  PROGress   y      n    ____________ __  ____________  __     0___
   2:  ALERTing   y      y    908957200000 __  ____________  __     0___
   3:  PROGress   y      y    0001________ __  ____________  __     0___
   4:  PROGress   n      n    ____________ __  ____________  __     0___
   5:  PROGress   n      y    625761449___ 01  ____________  __     0___
   6:  PROGress   n      n    ____________ __  ____________  __     0___
   7.  PROGress   n      n    ____________ __  ____________  __     0___
   8:  PROGress   n      n    ____________ __  ____________  __     0___

   Port Directory   Directory          Port Directory   Directory
          Number      Number                  Number      Number
     1:                                  5:
     2:                                  6:
     3:                                  7:
     4:                                  8:
```

Figure 416: BRI Trunk screen - Page 2 (using a TN2198/TN556B circuit pack)

```
change bri-trunk-board                                             Page 2 of x
                         ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt                          Max
       Message   Test? Init?               ID                          NCA TSC
   1:  PROGress   n      y    ____________  __                          0___
   2:  ALERTing   n      y    ____________  __                          0___
   3:  PROGress   n      y    ____________  __                          0___
   4:  PROGress   n      y    ____________  __                          0___
   5:  PROGress   n      y    ____________  __                          0___
   6:  PROGress   n      y    ____________  __                          0___
   7.  PROGress   n      y    ____________  __                          0___
   8:  PROGress   n      y    ____________  __                          0___
   9:  PROGress   n      y    ____________  __                          0___
  10:  PROGress   n      y    ____________  __                          0___
  11:  PROGress   n      y    ____________  __                          0___
  12:  PROGress   n      y    ____________  __                          0___
```

Note:

You cannot change the **Endpt Init**, **SPID**, or **Endpt ID** port parameters unless that port is busied out or unadministered. It is possible to change all other fields on this page even if the corresponding port is active.

If the **Interface** field on page 1 contains a valid value when the screen is submitted, the contents of the fields on page 2 for that port are validated. If the **Interface** field is blank when the screen is submitted, the fields on this page for that port reset to their default values.

Directory Number

These 10-digit fields contain the directory numbers assigned to the interface, which it views as being allocated to 2 separate endpoints.

Valid entries	Usage
Any string of 1 to 10 digits	These fields must be administered in pairs. If you enter a value in one field, you must enter a value in the other.

Endpt ID

A 2-digit field containing the Endpoint Identifier expected by the far end. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
00 to **62**	Leading zeroes considered significant and not ignored.

Endpt Init

Indicates whether the far end supports endpoint initialization. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
y	If set to **y**, the **SPID** field must *not* be blank. Avaya Communication Manager blocks you from changing this field and the **SPID** field unless that port is busied out or unadministered.
n	If set to **n**, the **SPID** and **Endpt ID** fields must be blank.

Interworking Message

This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Max NCA TSC

Valid entries	Usage
0 to **63**	This 2-digit field gives the maximum number of Non-Call-Associated Temporary Signaling Connections allowed on this BRI D-channel. This field's function is the same as the field with the same name on the **Signaling Group** screen.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

SPID

A 12-digit field containing the SPID expected by the far end. Avaya Communication Manager blocks you from changing this field unless the port is busied out or unadministered. The only protocol supported for SPID initialization is Bellcore (Country Code 1). Trunks will not be put in service if SPID installation is not successful.

Valid entries	Usage
Any string of 1 to 12 digits	Leading zeroes considered significant and not ignored.

XID Test

Valid entries	Usage
y/n	Indicates whether the far end supports the Layer 2 XID test.

ISDN-BRI Trunk Member Administration

Administer BRI trunk members using the following scheme to address the individual B-channels:

- B-channel 1 uses the port address of the BRI Trunk Port.
- B-channel 2 uses the port address of B-channel 1 incremented by 16.

When adding a BRI trunk to an ISDN trunk-group, Avaya Communication Manager blocks you from administering a Signaling Group for that trunk member.

Avaya Communication Manager blocks you from administering a BRI trunk member if the port has not yet been administered on the **BRI Trunk** screen.

For example, administer the B-channels on a TN2185 circuit pack inserted in slot 01A10 as follows:

Port	B-channel 1	B-channel 2
1	01A1001	01A1017
2	01A1002	01A1018
3	01A1003	01A1019
4	01A1004	01A1020
5	01A1005	01A1021
6	01A1006	01A1022
7	01A1007	01A1023
8	01A1008	01A1024

Interactions

The **`add bri-trunk board PPCSS`** command is rejected if PPCSS identifies a TN556B circuit pack, and a port on that circuit pack has already been assigned to a station or data-module. If a TN556B circuit pack has been administered as a BRI trunk circuit pack, any port on that circuit pack is prevented from being assigned to a station or data-module.

Language Translations

Pre-translated messages are available in English, French, Italian, and Spanish to display on your system telephones. Translations for many Communication Manager messages can be assigned using the **Language Translations** screens. As of July 1, 2005, however, new messages are no longer added to the **Language Translations** screens, so these screens might not show all available Communication Manager messages.

As a preferred method for entering translations for user-defined phone messages, Avaya recommends using the *Avaya Message Editing Tool (AMET)*. This tool is available for download from http://www.avaya.com. For more information, see *Avaya Message Editing Tool Job Aid*.

All button names can be assigned a user-defined name. For more information, see Telephone Feature Buttons Table on page 134.

Note:

If "user-defined" is entered for the display language on the **Station** screen or **Attendant Console** screen, and no messages are defined on these screens, a string of asterisks appears on all display messages. For information on administering Unicode languages, see Administering Unicode display on page 203.

In this section, the field descriptions are listed before the screens.

Field descriptions for Language Translation pages

English

This is a display-only field. It contains the English version of the message on the display.

Meaning of English term

This is a display-only field. It explains the abbreviated English message.

Translation

Enter the message you want to appear on the telephone display in place of the English message. Remember that a long message might be shortened on telephones that display fewer than 32 characters.

Figure 417: Language Translations screen — AD programming

```
change display-messages ad-programming                          Page  1 of  x
                              LANGUAGE TRANSLATIONS

     English                            Translation

  1. Press button to program.        1. ****************************************
  2. Change program?                 2. ***********************
  3. Yes = 1 No = 2                  3. ****************
  4. Enter number:                   4. ****************************************
  5. Press # to save.                5. ****************
  6. Number saved.                   6. ****************************************
  7. Change label?                   7. ***********************
  8. Enter label:                    8. ****************************************
  9. Press * to advance; # to save.  9. ******************************
 10. Press * to reenter; # to save. 10. ******************************
 11. Label saved. Hang up to update. 11. ****************************************
```

Figure 418: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 1)

```
change display-messages auto-wakeup-dn-dst                      Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:    AUTO WAKEUP - Ext:
      Translation: ********************:

   2.     English: WAKEUP ENTRY DENIED
      Translation: ************************

   3.     English: WAKEUP REQUEST CANCELED
      Translation: ****************************************

   4.     English: WAKEUP REQUEST CONFIRMED
      Translation: ****************************************

   5.     English: Wakeup Call
      Translation: **************************************

   6.     English:  Time:
      Translation: *****:
```

Figure 419: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 2)

```
change display-messages auto-wakeup-dn-dst                     Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English:    DO NOT DIST - Ext:
      Translation: *******************:

   8.     English:     DO NOT DIST - Group:
      Translation: *********************:

   9.     English: DO NOT DIST ENTRY DENIED
      Translation: ************************

  10.     English: THANK YOU - DO NOT DIST ENTRY CONFIRMED
      Translation: ****************************************

  11.     English: THANK YOU - DO NOT DIST REQUEST CANCELED
      Translation: ****************************************
```

Figure 420: Language Translations screen — Auto-Wakeup-Dn-Dst (Page 3)

```
change display-messages auto-wakeup-dn-dst                     Page  3 of  x
                              LANGUAGE TRANSLATIONS

  12.     English: INTERVAL FULL
      Translation: *************

  13.     English: NO PERMISSION
      Translation: *************

  14.     English: SYSTEM FULL
      Translation: *************

  15.     English: TOO SOON
      Translation: *************

  16.     English: INVALID EXTENSION - TRY AGAIN
      Translation: ****************************************

  17.     English: INVALID GROUP - TRY AGAIN
      Translation: ****************************************
```

Figure 421: Language Translations screen — Button Labels (page 1)

```
change display-messages button-labels                            Page 1 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Abr Mark                  1. *************
  2. Abr Pause                 2. *************
  3. Abr Program               3. *************
  4. Abr Spec Char             4. *************
  5. Abr Stop                  5. *************
  6. Abr Suppress              6. *************
  7. AbRing                    7. *****
  8. Abr Wait                  8. *************
  9. Account                   9. *************
 10. AD                       10. **********
 11. AddBusyInd               11. *************
 12. AdmConnAlarm             12. *************
 13. AfterCall                13. **********
 14. Alert Agent              14. *************
```

Figure 422: Language Translations screen — Button Labels (page 2)

```
change display-messages button-labels                            Page 2 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Alternate FRL             1. *************
  2. ANI Request               2. *************
  3. Assist                    3. **********
  4. ASVN Halt                 4. *************
  5. AttQueueCall              5. *************
  6. AttQueueTime              6. *************
  7. Audix Record              7. *************
  8. Auto Callback             8. *************
  9. Auto Ckt Halt             9. *************
 10. AutoIC                   10. *******
 11. Auto In                  11. **********
 12. AutoWakeAlarm            12. *************
 13. Auto Wakeup              13. *************
 14. AuxWork                  14. ********
 15. Busy                     15. *****
```

Figure 423: Language Translations screen — Button Labels (page 3)

```
change display-messages button-labels                              Page 3 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Button Ring               1. *************
  2. Button View               2. *************
  3. Caller Info               3. *************
  4. CFrwd                     4. *****
  5. Call Park                 5. *************
  6. Call Pickup               6. *************
  7. Call Time                 7. *************
  8. CAS Backup                8. *************
  9. Cancel LWC                9. *************
 10. CDR1 Fail                10. *************
 11. CDR2 Fail                11. *************
 12. CFBDA                    12. *****
 13. Check In                 13. *************
 14. Check Out                14. *************
```

Figure 424: Language Translations screen — Button Labels (page 4)

```
change display-messages button-labels                             Page 4 of x
                               LANGUAGE TRANSLATIONS

 English                          Translation

  1. ClkOverride               1. ************
  2. CO Line                   2. ***********
  3. Conf Display              3. *************
  4. Consult                   4. *************
  5. Cover Msg Ret             5. *************
  6. CovrCallBack              6. *************
  7. CPN Block                 7. *************
  8. CPN Unblock               8. *************
  9. Crisis Alert              9. *************
 10. Data                     10. *****
 11. Delete Msg               11. *************
 12. Dial Icom                12. **********
 13. DID Remove               13. *************
 14. DID View                 14. *************
```

Figure 425: Language Translations screen — Button Labels (page 5)

```
change display-messages button-labels                        Page 5 of x
                              LANGUAGE TRANSLATIONS

 English                           Translation

  1. Directory                  1. *************
  2. Dir Pickup                 2. *************
  3. Disp Charges               3. *************
  4. DoNotDisturb               4. *************
  5. EC500                      5. *************
  6. Exclusion                  6. *************
  7. ExtDoNotDstrb              7. *************
  8. Extend Call                8. *************
  9. Far End Mute               9. *************
 10. Flash                     10. *************
 11. FTC Alarm                 11. *************
 12. Goto Cover                12. *************
 13. GrpPg                     13. *****
 14. GrpDoNotDstrb             14. *************
 15. Hunt NS                   15. **********
 16. InCalID                   16. ********
```

Figure 426: Language Translation screen - Button Labels (page 6)

```
change display-messages button-labels                        Page 6 of x
                              LANGUAGE TRANSLATIONS

 English                           Translation

  1. Inspect                    1. *************
  2. IntAutoAnswer              2. *************
  3. License Error              3. *************
  4. Link Fail                  4. ***********
  5. Lock LWC                   5. *************
  6. LSVN Halt                  6. *************
  7. Major Alarm                7. *************
  8. Make Call                  8. *************
  9. ManOverid                  9. ************
 10. Manual In                 10. **********
 11. MCT Activate              11. *************
 12. MCT Control               12. *************
 13. Mj/Mn Alarm               13. *************
```

Figure 427: Language Translations screen — Button Labels (page 7)

```
change display-messages button-labels                          Page 7 of x
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. MM Basic                    1. *************
  2. MM Call                     2. *************
  3. MM Call Fwd                 3. *************
  4. MM Data Conf                4. *************
  5. MM Mult Nbr                 5. *************
  6. MM PC Audio                 6. *************
  7. Msg                         7. *****
  8. Msg Retrieve                8. *************
  9. MsgW                        9. *****
 10. MsgWaitAct                 10. *************
 11. MsgWaitDeact               11. *************
 12. MST Debug                  12. *************
 13. Next                       13. *************
 14. Night Service              14. *************
```

Figure 428: Language Translations screen — Button Labels (page 8)

```
change display-messages button-labels                           Page 8 of x
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. NoAnsAltr                   1. **********
  2. OffBoardAlarm               2. *************
  3. PAGE1 Alarm                 3. *************
  4. PAGE2 Alarm                 4. *************
  5. PMS Failure                 5. *************
  6. PMS Ptr Alarm               6. *************
  7. Posted MSGs                 7. *************
  8. Priority Call               8. *************
  9. QueueCall                   9. **********
 10. QueueTime                  10. **********
 11. Release                    11. *************
 12. RemBusyInd                 12. *************
 13. ResetAlert                 13. *************
 14. Ringer Off                 14. *************
 15. Ring Stat                  15. *************
```

Figure 429: Language Translations screen — Button Labels (page 9)

```
change display-messages button-labels                          Page 9 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. RSVN Halt                 1. *************
  2. SD                        2. **
  3. SendAllCalls              3. *************
  4. Send TEG                  4. *************
  5. Service Obsrv             5. *****
  6. Sgnl                      6. *************
  7. SSVN Halt                 7. *************
  8. Start Bill                8. *************
  9. Station Lock              9. *************
 10. Stored Number            10. *************
 11. Store LWC                11. *************
 12. Stroke Count             12. ************
 13. System Alarm             13. *************
 14. TermGroup                14. **********
 15. Time/Date                15. *************
 16. Timer                    16. *************
```

Figure 430: Language Translations screen — Button Labels (page 10)

```
change display-messages button-labels                         Page 10 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Toggle Swap  :            1. ************
  2. Trunk ID                  2. ***********
  3. Trunk Name                3. *************
  4. Trunk NS                  4. *************
  5. UUI Info                  5. *************
  6. Verify                    6. *************
  7. VIP Check In              7. *************
  8. VIP retry                 8. *************
  9. VIP Wakeup                9. *************
 10. VOA Repeat               10. *****
 11. VU Display               11. *************
 12. WhisperAct               12. **********
 13. WhisperAnbk              13. *************
 14. WhisperOff               14. *************
 15. Work Code                15. *************
```

Figure 431: Language Translations screen — Call-Identifiers (Page 1)

```
change display-messages call identifiers                         Page  1 of  x
                              LANGUAGE TRANSLATIONS

     English               Meaning of English term                   Translated
     Term                                                            Term

  1. sa           ACD Supervisor Assistance                       1: **
  2. ac           Attendant Assistance Call                       2: **
  3. tc           Attendant Control Of A Trunk Group              3: **
  4. an           Attendant No Answer                             4: **
  5. pc           Attendant Personal Call                         5: **
  6. rc           Attendant Recall Call                           6: **
  7. rt           Attendant Return Call                           7: **
  8. sc           Attendant Serial Call                           8: **
  9. co           Controlled Outward Restriction                  9: **
 10. cs           Controlled Station To Station Restriction      10: **
 11. ct           Controlled Termination Restriction             11: **
 12. db           DID Find Busy Station With CO Tones            12: **
 13. da           DID Recall Go To Attendant                     13: **
 14. qf           Emergency Queue Full Redirection               14: **
 15. hc           Held Call Timed Reminder                       15: **
```

Figure 432: Language Translations screen — Call-Identifiers (Page 2)

```
change display-messages call identifiers                         Page  2 of  x
                              LANGUAGE TRANSLATIONS

     English               Meaning of English term                   Translated
     Term                                                            Term

 16. ic           Intercept                                      16: **
 17. ip           Interposition Call                             17: **
 18. ld           LDN Calls On DID Trunks                        18: **
 19. so           Service Observing                              19: **
 20. na           Unanswered Or Incomplete DID Call              20: **
 21. ACB          Automatic Callback                             21: ********
 22. callback     Callback Call                                  22: ********
 23. park         Call Park                                      23: ********
 24. control      Control                                        24: ********
 25. ICOM         Intercom Call                                  25: ********
 26. OTQ          Outgoing Trunk Queuing                         26: ********
 27. priority     Priority Call                                  27: ********
 28. recall       Recall Call                                    28: ********
 29. return       Return Call                                    29: ********
 30. ARS          Automatic Route Selection                      30: ********
```

Figure 433: Language Translations screen — Call-Identifiers (Page 3)

```
change display-messages call identifiers                       Page  3 of  x
                              LANGUAGE TRANSLATIONS

     English               Meaning of English term                  Translated
     Term                                                           Term

 31. forward       Call Forwarding                              31: ********
 32. cover         Cover                                        32: ********
 33. DND           Do Not Disturb                               33: ********
 34. p             Call Pickup                                  34: *
 35. c             Cover All Calls                              35: *
 36. n             Night Station Service, Including No Answer   36: *
 37. B             All Calls Busy                               37: *
 38. f             Call Forwarding                              38: *
 39. b             Cover Busy                                   39: *
 40. d             Cover Don't Answer                           40: *
 41. s             Send All Calls                               41: *
 42. to            <calling party> to <called party>            42: **
 43. VDN           Vector Directory Number                      43: ***
 44. hunt          Station Hunting, Origination                 44: ********
 45. h             Station Hunting, Termination                 45: *
```

Figure 434: Language Translations screen — Call-Identifiers (Page 4)

```
change display-messages call identifiers                       Page  4 of  x
                              LANGUAGE TRANSLATIONS

     English         Meaning of English term                 Translated
     Term                                                    Term

 46. OPERATOR        Operator                              46: ***************
 47. EXT             Extension                             47: ***************
 48. OUTSIDE CALL    Outside Call                          48: ***************
 49. UNKNOWN NAME    Unknown Name                          49: ***************
 50. CONFERENCE      Conference                            50: ************
 51. ringing         Ringing                               51: ********
 52. busy            Busy                                  52: ********
 53. busy(I)         Busy With Intrusion Allowed           53: ********
 54. wait            Wait                                  54: *****
 55. (I)             Intrusion                             55: ***
 56. Sta             Station                               56: *********
 57. Trk             Trunk                                 57: *********
 58: offered         QSIG call offered to remote endpoint  58: ********
 59: cl              Controlled Toll Restriction           59: **
 60: vm              Call to Attendant Out of Voicemail    60: **
```

Figure 435: Language Translations screen — Date-Time (Page 1)

```
change display-messages date-time                               Page  1 of  x
                          LANGUAGE TRANSLATIONS

    English          Translation            English          Translation

  1. SUNDAY         1: ***********        11. APRIL         11: ***********
  2. MONDAY         2: ***********        12. MAY           12: ***********
  3. TUESDAY        3: ***********        13. JUNE          13: ***********
  4. WEDNESDAY      4: ***********        14. JULY          14: ***********
  5. THURSDAY       5: ***********        15. AUGUST        15: ***********
  6. FRIDAY         6: ***********        16. SEPTEMBER     16: ***********
  7. SATURDAY       7: ***********        17. OCTOBER       17: ***********
  8. JANUARY        8: ***********        18. NOVEMBER      18: ***********
  9. FEBRUARY       9: ***********        19. DECEMBER      19: ***********
 10. MARCH         10: ***********

 20.     English: SORRY, TIME UNAVAILABLE NOW
     Translation: ****************************************
```

Figure 436: Language Translations screen — Leave-Word-Calling (Page 2)

```
change display-messages leave-word-calling                       Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:             MESSAGES FOR
      Translation: ************************

   2.     English: WHOSE MESSAGES? (DIAL EXTENSION NUMBER)
      Translation: ****************************************

   3.     English: END OF MESSAGES (NEXT TO REPEAT)
      Translation: ****************************************

   4.     English: MESSAGES UNAVAILABLE - TRY LATER
      Translation: ****************************************

   5.     English: MESSAGE RETRIEVAL DENIED
      Translation: ****************************************

   6.     English: MESSAGE RETRIEVAL LOCKED
      Translation: ****************************************
```

Figure 437: Language Translations screen — Leave-Word-Calling (Page 2)

```
change display-messages leave-word-calling                       Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English: NO MESSAGES
      Translation: ****************************************

   8.     English: IN PROGRESS
      Translation: ****************************************

   9.     English: DELETED
      Translation: ****************************************

  10.     English: GET DIAL TONE, PUSH Cover Msg Retrieval
      Translation: ****************************************

  11.     English: Message Center (AUDIX) CALL
      Translation: ****************************************

  12.     English: CANNOT BE DELETED - CALL MESSAGE CENTER
      Translation: ****************************************
```

Figure 438: Language Translations screen — Malicious-Call-Trace (Page 1)

```
change display-messages mailcious-call-trace                     Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English: MALICIOUS CALL TRACE REQUEST
      Translation: ****************************************

   2.     English: END OF TRACE INFORMATION
      Translation: ****************************************

  3.     English: original call redirected from:
     Translation: ****************************:

  4.     English:           voice recorder port:
     Translation: ****************************:

  5.     English:  MCT activated by:        for:
     Translation: *****************:       ****:
```

Figure 439: Language Translations screen — Malicious-Call-Trace (Page 2)

```
change display-messages mailcious-call-trace                    Page  2 of  x
                             LANGUAGE TRANSLATIONS


  6.     English:    party :                   (EXTENSION)
     Translation: ******** :       ***********************

  7.     English:    party :                (ISDN SID/CNI)
     Translation: ******** :              ****************

  8.     English:    party :                     (PORT ID)
     Translation: ******** :           *******************

  9.     English:    party :                (ISDN PORT ID)
     Translation: ******** :           *******************
```

Figure 440: Language Translations screen — Miscellaneous-Features (Page 1)

```
change display-messages miscellaneous-features                   Page  1 of x
                              LANGUAGE TRANSLATIONS

     English                                 Translation

  1. ALL MADE BUSY                       1: ***************
  2. BRIDGED                             2: ***************
  3. DENIED                              3: ***************
  4. INVALID                             4: ***************
  5. NO MEMBER                           5: ***************
  6. OUT OF SERVICE                      6: ***************
  7. RESTRICTED                          7: ***************
  8. TERMINATED                          8: ***************
  9. TRUNK SEIZED                        9: ***************
 10. VERIFIED                           10: ***************
 11. CDR OVERLOAD                       11: ***************
 12. ANSWERED BY                        12: ***************
 13. CALL FROM                          13: ***************
 14. Skills                             14: ***************
```

Figure 441: Language Translations screen — Miscellaneous Features (Page 2)

```
change display-messages miscellaneous-features                  Page  2 of x
                              LANGUAGE TRANSLATIONS
     English               Meaning of English term              Translated
     Term                                                       Term

 15. TOLL          Toll                                     15: ****
 16. FULL          Full                                     16: ****
 17. NONE          None                                     17: ****
 18. ORIG          Origination                              18: ****
 19. OTWD          Outward                                  19: ****
 20. CALL          <call> This Number                       20: ****
 21. INTL          International                            21: ****
 22. Info          Information                              22: *****
 23. p             Primary                                  23: *
 24. s             Secondary                                24: *
 25. m             Mark                                     25: *
 26. p             Pause                                    26: *
 27. s             Suppress                                 27: *
 28. w             Wait For A Specified Time                28: *
 29. W             Wait For Off-Premise Dial Tone           29: *
```

Figure 442: Language Translations screen — Miscellaneous-Features (Page 3)

```
change display-messages miscellaneous-features                  Page 3 of x

                              LANGUAGE TRANSLATIONS

  30.     English: You have adjunct messages
      Translation: ****************************************

  31.     English: Login Violation
      Translation: ****************************************

  32.     English: Barrier Code Violation
      Translation: ****************************************

  33.     English: Authorization Code Violation
      Translation: ****************************************

  34.     English: DIRECTORY - PLEASE ENTER NAME
      Translation: ****************************************

  35.     English: DIRECTORY UNAVAILABLE - TRY LATER
      Translation: ****************************************
```

Figure 443: Language Translations screen — Miscellaneous-Features (Page 4)

```
change display-messages miscellaneous-features                   Page 4 of x

                              LANGUAGE TRANSLATIONS

  36.     English: NO MATCH - TRY AGAIN
      Translation: ****************************************

  37.     English: NO NUMBER STORED
      Translation: ****************************************

  38.     English: TRY AGAIN
      Translation: ****************************************

  39.     English: Ext           in EMRG Q
      Translation: ***           **********

  40.     English:       HUNT GROUP   NOT ADMINISTERED
      Translation: ****************   *********************

  41.     English: Q-time        calls
      Translation: ******        *****
```

Figure 444: Language Translations screen — Miscellaneous-Features (Page 5)

```
change display-messages miscellaneous-features                   Page 5 of x
                             LANGUAGE TRANSLATIONS

   42.     English: Add Skill: Enter number, then # sign
       Translation: ****************************************

   43.     English: Remove Skill: Enter number, then # sign
       Translation: ****************************************

   44.     English: Enter Skill Level, then # sign
       Translation: ****************************************

   45.     English: Enter Agent LoginID
       Translation: ****************************************

   46.     English: Call Type
       Translation: ***************

   47.     English: Call Charge
       Translation: ********************
```

Figure 445: Language Translations screen — Miscellaneous-Features (Page 6)

```
change display-messages miscellaneous-features                    Page 6 of x
                               LANGUAGE TRANSLATIONS

   48.     English: Station Security Code Violation
       Translation: ***************************************
   49.     English: ENTER REASON CODE
       Translation: ***************************************
   50.     English: Whisper From
       Translation: ********************
   51.     English: Whisper To
       Translation: ********************
   52.     English: Press button to remove.
       Translation: ***************************************
   53.     English: Press # to remove.
       Translation: ********************
```

Figure 446: Language Translations screen — Miscellaneous-Features (Page 7)

```
change display-messages miscellaneous-features                     Page 7 of x
                               LANGUAGE TRANSLATIONS

   54.     English: Button removed.
       Translation: ***************************************
   55.     English: Ringer On
       Translation: ***************************************
   56.     English: Ringer Off
       Translation: ***************************************
   57.     English: Ringer Abbreviated
       Translation: ***************************************
   58.     English: Ringer Delayed
       Translation: ***************************************
   59.     English: Select a held party's line to talk.
       Translation: ***************************************
```

Figure 447: Language Translations screen — Property-Management (Page 1)

```
change display-messages property-management                     Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English:                    CHECK IN - Ext:
      Translation: *********************************:

   2.     English: CHECK-IN: ROOM ALREADY OCCUPIED
      Translation: ****************************************

   3.     English: CHECK IN COMPLETE
      Translation: ****************************************

   4.     English: CHECK IN FAILED
      Translation: ****************************************

   5.     English:                   CHECK OUT - Ext:
      Translation: *********************************:

   6.     English: CHECK OUT: ROOM ALREADY VACANT
      Translation: ****************************************
```

Figure 448: Language Translations screen — Property-Management (Page 2)

```
change display-messages property-management                    Page  2 of  x
                              LANGUAGE TRANSLATIONS

   7.     English: CHECK OUT FAILED
      Translation: ****************************************

   8.     English: MESSAGE NOTIFICATION FAILED
      Translation: ****************************************

   9.     English:    MESSAGE NOTIFICATION ON - Ext:
      Translation: ********************************:

  10.     English:   MESSAGE NOTIFICATION OFF - Ext:
      Translation: ********************************:
  11.     English: CHECK OUT COMPLETE: MESSAGE LAMP OFF
      Translation: ****************************************

  12.     English: CHECK OUT COMPLETE: MESSAGE LAMP ON
      Translation: ****************************************
```

Figure 449: Language Translations screen — Property-Management (Page 3)

```
change display-messages property-management                    Page  3 of  x
                              LANGUAGE TRANSLATIONS


 13.     English: MESSAGE LAMP ON
     Translation: ****************************************

 14.     English: MESSAGE LAMP OFF
     Translation: ****************************************
 15.     English: Occupied Rooms
     Translation: ****************************************
 16.     English: Enter Room Status (1-6)
     Translation: ****************************************
 17.     English: State, Enter number from 1 - 6
     Translation: ****************************************
 18.     English: DID
     Translation: *********************
```

Figure 450: Language Translations screen — Property-Management (Page 4)

```
change display-messages property-management                    Page 4 of x
                              LANGUAGE TRANSLATIONS
   19.     English: DID VIEW: EXT?
       Translation: ****************************************
   20.     English: DID=                   CHANGE?
       Translation: ****                   **********************
   21.     English: DID VIEW DONE
       Translation: ****************************************
   22.     English: NO DID AVAILABLE
       Translation: ****************************************
   23.     English: CHECK IN COMPLETE, DID=
       Translation: *********************************
```

Figure 451: Language Translations screen — Property-Management (Page 5)

```
change display-messages property-management                     Page 5 of x

                               LANGUAGE TRANSLATIONS
   24.     English: REMOVE(1), REPLACE(2)?
       Translation: ****************************************
   25.     English: DID REMOVED
       Translation: ****************************************
   26.     English: VIP CHECK IN - Ext:
       Translation: ****************************************
   27.     English: SPECIFY VID DID:
       Translation: ****************************************
   28.     English: CHECK IN COMPLETE, INVALID DID
       Translation: ****************************************
   29.     English: CHECK IN COMPLETE, DID UNAVAILABLE
       Translation: ****************************************
```

Figure 452: Language Translations screen — Property-Management (Page 6)

```
change display-messages property-management                     Page 6 of x

                               LANGUAGE TRANSLATIONS
   30.     English: DID REMOVE - Ext:
       Translation: ****************************************
   31.     English: DID CHANGED
       Translation: ****************************************
   32.     English: AUTOMATIC ASSIGN(1), SELECT(2)?
       Translation: ****************************************
   33.     English: DID UNAVAILABLE
       Translation: ****************************************
```

Figure 453: Language Translations screen — Self-Administration (page 1)

```
hange display-messages self-administration                    Page 1 of x

                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. SECURITY CODE:            1. **************
  2. INCORRECT SECURITY CODE   2. ************************
  3. SELECT FEATURE            3. ************************
  4. EXTENSION:                4. ******************
  5. OPTIONAL EXTENSION:       5. ******************
  6. TEL NUM:                  6. ********
  7. PRESS BUTTON TO PROGRAM   7. ************************
  8. BUTTON PROGRAMMED!        8. ************************
  9. INCORRECT BUTTON          9. ************************
 10. BUTTON LIMIT REACHED     10. ************************
 11. BUSY, TRY AGAIN LATER    11. ************************
 12. YOUR PHONE IS BUSY       12. ************************
 13. SECURITY CODE NEEDED     13. ************************
 14. BUTTON NOT CHANGED       14. ************************
```

Figure 454: Language Translations screen — Self-Administration (page 2)

```
change display-messages self-administration                     Page 2 of x

                              LANGUAGE TRANSLATIONS

 English     Translation    English                Translation

  1. Acct    *****          CDR Account Code       ************************
  2. AutoD   *****          Automatic Dialing      ************************
  3. CFrwd   *****          Call Forwarding        ************************
  4. CPark   *****          Call Park              ************************
  5. CPkUp   *****          Call Pickup            ************************
  6. DPkUp   *****          Directed Call Pickup   ************************
  7. GrpPg   *****          Group Paging           ************************
  8. SAC     *****          Send All Calls         ************************
  9. Swap    *****          Conf/Trans Toggle-Swap ************************
 10. WspPg   *****          Activate whisper Page  ************************
 11. WspAn   *****          Answerback for Whisper ************************
 12. WsOff   *****          Whisper Page Off       ************************
 13. Blank   *****          Blank Button           ************************
```

Figure 455: Language Translations screen — Self-Administration (page 3)

```
change display-messages self-administration                     Page 3 of x

                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Whisper Page Off 1. *****
  2. Blank Button     2. *****
  3. Done             3. *****
  4. Cont             4. **********
  5. Expl?            5. *****
  6. ShortMode?       6. *****
  7. Next             7. *****
  8. Selct            8. *****
  9. Clear            9. *****
 10. Cancel          10. **********
 11. Delete          11. *****
 12. Replace         12. *****
 13. Bksp            13. *****
```

Figure 456: Language Translations screen — View-buttons (page 1)

```
change display-messages view-buttons                           Page 1 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Analog Bridge Appearance       1. ************************
  2. Abbreviated Dial Program       2. ************************
  3. Abbrev Dial Character          3. ************************
  4. Abbreviated Dial               4. ************************
  5. Abrv/Delayed Ring Change       5. ************************
  6. Auto Circuit Assurance         6. ************************
  7. Admin Connection Alarm         7. ************************
  8. CDR Account Code               8. ************************
  9. TSA Administration Mode        9. ************************
 10. ACD After Call Work           10. ************************
 11. ACD Change Alert              11. ************************
 12. Alternate FRL                 12. ************************
 13. Supervisor Assist             13. ************************
 14. SVN Auth Code Halt            14. ************************
 15. Attendant Queue Calls         15. ************************
 16. Attendant Queue Time          16. ************************
```

Figure 457: Language Translations screen — View-buttons (page 2)

```
change display-messages view-buttons                              Page 2 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Auto Message Waiting          1. ************************
  2. Automatic Call Back           2. ************************
  3. Automatic Dialing             3. ************************
  4. Automatic Intercom            4. ************************
  5. Auto-In Work Mode             5. ************************
  6. Automatic Wakeup              6. ************************
  7. Auxiliary Work Mode           7. ************************
  8. Bridged Appearance            8. ************************
  9. Busy Indicator                9. ************************
 10. Call Appearance              10. ************************
 11. Call Displayed Number        11. ************************
 12. Call Forwarding              12. ************************
 13. Call Park                    13. ************************
 14. Call Pickup                  14. ************************
 15. Caller Information           15. ************************
 16. CAS (Branch) Backup          16. ************************
```

Figure 458: Language Translations screen — View-buttons (page 3)

```
change display-messages view-buttons                             Page 3 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. CDR 1st Printer Alarm         1. ************************
  2. CDR 2nd Printer Alarm         2. ************************
  3. Check In Hotel/Motel          3. ************************
  4. Check Out Hotel/Motel         4. ************************
  5. Call Forwarding Busy/DA       5. ************************
  6. Clocked Override              6. ************************
  7. Consult/Return                7. ************************
  8. Coverage Callback             8. ************************
  9. Cover Message Retrieve        9. ************************
 10. Data Extension               10. ************************
 11. Time of Day/Date Display     11. ************************
 12. Delete LWC Message           12. ************************
 13. Dial Intercom                13. ************************
 14. Integrated Directory         14. ************************
 15. Directed Call Pickup         15. ************************
 16. Normal/Local Mode Toggle     16. ************************
```

Figure 459: Language Translations screen — View-buttons (page 4)

```
change display-messages view-buttons                              Page 4 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Do Not Disturb                 1. ************************
  2. Drop                           2. ************************
  3. Off-Board DS1 Alarm            3. ************************
  4. Manual Exclusion               4. ************************
  5. Do Not Disturb Extension       5. ************************
  6. Flash                          6. ************************
  7. Go To Cover                    7. ************************
  8. Do Not Disturb Group           8. ************************
  9. Group Paging                   9. ************************
 10. Headset On/Off                10. ************************
 11. Hunt Night Service            11. ************************
 12. Coverage Call Identify        12. ************************
 13. Inspect Call Appearance       13. ************************
 14. Internal Auto Answer          14. ************************
 15. Last Number Dialed            15. ************************
 16. Link Failure Alarm            16. ************************
```

Figure 460: Language Translations screen — View-buttons (page 5)

```
change display-messages view-buttons                               Page 5 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Login Security Violation       1. ************************
  2. Cancel Leave Word Call         2. ************************
  3. Lockout Leave Word Call        3. ************************
  4. Leave Word Call Message        4. ************************
  5. Major Hardware Alarm           5. ************************
  6. Manual Message Waiting         6. ************************
  7. Manual Override                7. ************************
  8. ACD Manual-In                  8. ************************
  9. MCT Call Trace Activate        9. ************************
 10. MCT Call Trace Control        10. ************************
 11. Major/Minor Alarm             11. ************************
 12. Message Retrieve              12. ************************
 13. Message Waiting On            13. ************************
 14. Message Waiting Off           14. ************************
 15. Next                          15. ************************
 16. Night Service Activate        16. ************************
```

Figure 461: Language Translations screen — View-buttons (page 6)

```
change display-messages view-buttons                              Page 6 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Redirect On No Answer          1. ************************
  2. Normal Display Mode            2. ************************
  3. Personal CO Line               3. ************************
  4. Property Management Fail       4. ************************
  5. Wakeup Printer Alarm           5. ************************
  6. Print Messages                 6. ************************
  7. Priority Calling               7. ************************
  8. PMS Printer Alarm              8. ************************
  9. System Printer Alarm           9. ************************
 10. Number of Queued Calls        10. ************************
 11. Oldest Queued Time            11. ************************
 12. ACD Release                   12. ************************
 13. Ringer Cut-Off                13. ************************
 14. System Reset Alert            14. ************************
 15. Remote Access Violation       15. ************************
 16. Scroll Mode                   16. ************************
```

Figure 462: Language Translations screen — View-buttons (page 7)

```
change display-messages view-buttons                              Page 7 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Send All Calls                 1. ************************
  2. TEG Send All Calls             2. ************************
  3. Service Observing              3. ************************
  4. Manual Signalling              4. ************************
  5. Station Security Call          5. ************************
  6. Stored Number Display          6. ************************
  7. Stroke Counts                  7. ************************
  8. Term Extension Group           8. ************************
  9. Timer                          9. ************************
 10. Facility Test Call Alarm      10. ************************
 11. Trunk ID                      11. ************************
 12. Trunk Name                    12. ************************
 13. Trunk Night Service           13. ************************
 14. UUI Info                      14. ************************
 15. Busy Verification             15. ************************
 16. VDN of Origin Announce        16. ************************
```

Figure 463: Language Translations screen — View-buttons (page 8)

```
change display-messages view-buttons                             Page 8 of x
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Vu-Stats Displays             1. ************************
  2. Activate Whisper Page         2. ************************
  3. Whisper Page Answerback       3. ************************
  4. Whisper Page Off              4. ************************
  5. Call Work Code                5. ************************
  6. Unassigned Button             6. ************************
  7. View Button                   7. ************************
  8. Call Timer                    8. ************************
  9. Add Busy Indicator            9. ************************
 10. Remove Busy Indicator        10. ************************
 11. VIP Wakeup                   11. ************************
 12. VIP Retry                    12. ************************
 13. Crisis Alert                 13. ************************
 14. DID View                     14. ************************
 15. DID Remove                   15. ************************
 16. VIP Check In                 16. ************************
```

Figure 464: Language Translations screen — View-buttons (page 9)

```
change display-messages view-buttons                            Page   9 of x
                               LANGUAGE TRANSLATIONS

 English                           Translation

  1.  Station Lock                  1. ************************
  2.  License Error                 2. ************************
  3.  Conference Display            3. ************************
  4.  Conf/Trans Toggle-Swap        4. ************************
  5.  Far End Mute                  5. ************************
  6.  Paging 1st Link Alarm         6. ************************
  7.  Paging 2nd Link Alarm         7. ************************
  8.  EC500                         8. ************************
  9.  No Hold Conference            9. ************************
 10.  Posted Messages              10. ************************
 11.  Audix Recording              11. ************************
 12.  Extend Call                  12. ************************
```

Figure 465: Language Translations screen — Vustats

```
change display-messages vustats                          Page   1 of x

                              LANGUAGE TRANSLATIONS

 English                           Translation

  1.  FORMAT                        1. *******

  2.  NOT DEFINED                   2. *****************************
  3.  DOES NOT ALLOW OR REQUIRE ID  3. *****************************

  4.  AGENT                         4. ************
  5.  SPLIT/SKILL                   5. ************
  6.  TRUNK GROUP                   6. ************
  7.  VDN                           7. ************

  8. NOT ADMINISTERED               8. *****************
  9. NOT MEASURED                   9. *****************

 10. AGENT NOT LOGGED IN           10. ****************************************
```

Figure 466: Language Translations screen — Softkey-Labels

```
change display-messages softkey-labels                 Page   1 of   x
                        LANGUAGE TRANSLATIONS
English      Translation   English      Translation   English      Translation
 1. Acct      1. *****     17. Drop     17. *****     33. RmBsy    33. *****
 2.  AD       2. *****     18. Excl     18. *****     34. RngOf    34. *****
 3. AdBsy     3. *****     19. FMute    19. *****     35.  SAC     35. *****
 4. Admin     4. *****     20. GrpPg    20. *****     36. SFunc    36. *****
 5. AutCB     5. *****     21. HFAns    21. *****     37. Spres    37. *****
 6. BtnVu     6. *****     22. IAuto    22. *****     38. Stats    38. *****
 7. CFrwd     7. *****     23. IDial    23. *****     39. Stop     39. *****
 8. CnfDs     8. *****     24. Inspt    24. *****     40. Swap     40. *****
 9. CnLWC     9. *****     25. Last     25. *****     41. Timer    41. *****
10. Cnslt    10. *****     26.  LWC     26. *****     42. TmDay    42. *****
11. Count    11. *****     27. Mark     27. *****     43. View     43. *****
12. CPark    12. *****     28. NHCnf    28. *****     44. Wait     44. *****
13. CPkUp    13. *****     29. Pause    29. *****     45. WspAn    45. *****
14. CTime    14. *****     30. PCall    30. *****     46. WspPg    46. *****
15.  Dir     15. *****     31. PoMSG    31. *****
16. DPkUp    16. *****     32. Prog     32. *****
```

In order to provide unique labeling for abbreviated dialing button types for softkey-labels, Communication Manager replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D telephones.

On the softkey label **Language Translation** screen, the digits following the "**AD**" are derived from the button position. If the first button is an AD button, then it is **AD1** and the fifteenth button is **AD15**. All the AD buttons between 1 and 15 have the position number appended to "**AD**."

Figure 467: Language Translations screen — Time-Of-Day-Routing

```
change display-messages time-of-day-routing                     Page  1 of  x
                              LANGUAGE TRANSLATIONS

   1.     English: ENTER ACTIVATION ROUTE PLAN, DAY & TIME
      Translation: ****************************************

   2.     English: ENTER DEACTIVATION DAY AND TIME
      Translation: ****************************************

   3.     English:    OLD ROUTE PLAN:      ENTER NEW PLAN:
      Translation: *****************:   *****************:

   4.     English:    OLD ROUTE PLAN:         NEW PLAN:
      Translation: *****************:   **************:

   5.     English: ROUTE PLAN:    FOR       ACT-TIME:
      Translation: **********:   ****     **********:

   6.     English: ROUTE PLAN:    FOR     DEACT-TIME:
      Translation: **********:   ****     **********:
```

Figure 468: Language Translations Transfer-conference screen (page 1)

```
change display-messages transfer-conference                    Page 1 of x
                         LANGUAGE TRANSLATIONS

 1.     English: Transfer completed.
    Translation: ********************************

 2.     English: Call next party.
    Translation: ****************************************

 3.     English: Press conference to add party.
    Translation: ****************************************

 4.     English: ^-party conference in progress.
    Translation: ****************************************

 5.     English: Conference canceled.
    Translation: ****************************************

 6.     English: Select line ^ to cancel or another line.
    Translation: ****************************************
```

Note:

For Messages 4, 6, 12, you manually must change "~" to "^" in your user-defined language. The software will not update automatically.

Message 4

The character "^" is a place holder.

English Text	Replacement Info
^-party conference in progress	"^" is replaced with the number of parties currently on the conference call.

Message 6

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to cancel or another line.	"^" is replaced with the letter of the line that is on soft hold.

Figure 469: Language Translations Transfer-conference screen (page 2)

```
change display-messages transfer-conference           Page 2 of x
                         LANGUAGE TRANSLATIONS

 7.     English: Dial number.
    Translation: ****************************************

 8.     English: Press transfer to complete.
    Translation: ****************************************

 9.     English: Hang-up to complete transfer.
    Translation: ****************************************

10.     English: Dial number or select held party.
    Translation: ****************************************

11.     English: Select held party to conference.
    Translation: ****************************************

12.     English: Select line ^ to add party.
    Translation: ****************************************
```

Figure 470: Language Translations Transfer-conference screen (page 3)

```
change display-messages transfer-conference           Page 3 of x
                         LANGUAGE TRANSLATIONS

13.     English: Select alerting line to answer call.
    Translation: ****************************************

14.     English: Transfer canceled.
    Translation: ****************************************

15.     English: Connecting to ^.
    Translation: ****************************************

16.     English: Called party ^is busy.
    Translation: ****************************************

17.     English: Invalid number dialed -
    Translation: ***************************

18.     English: Party ^ is not available.
    Translation: ***************************
```

Message 15, 16, 18

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Figure 471: Language Translations Transfer- Conference screen (page 4)

```
change display-messages transfer-conference                        Page 4 of x
LANGUAGE TRANSLATIONS
19. English: Mute
Translation: ****
20. English: ^-party conference:
Translation: *********************************
```

Message 12

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Figure 472: Language Translations screen — VuStats

```
change display-messages vustats                                    Page  1 of  x
                              LANGUAGE TRANSLATIONS

        English                          Translations

   1.   FORMAT                           1.   *******
   2.   NOT DEFINED                      2.   *******
   3.   DOES NOT ALLOW OR REQUIRE ID     3.   *******

   4.   AGENT                            4.   *****************************
   5.   SPLIT/SKILL                      5.   *****************************
   6.   TRUNK GROUP                      6.   *****************************
   7.   VND                              7.   *****************************

   8.   NOT ADMINISTERED                 8.   *****************
   9.   NOT MEASURED                     9.   *****************

  10.   AGENT NOT LOGGED IN             10.    *************************
```

Listed Directory Numbers

Allows Direct Inward Dialing (DID) numbers to be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

The number of Listed Directory Numbers that can be assigned varies depending on system configuration. See the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207, for maximum values.

Field descriptions for page 1

Figure 473: Listed Directory Numbers screen

```
change listed-directory-number                                Page   1 of  x
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext     Name                         TN
      1:
      2:                                      1
      3:                                      1
      4:                                      1
      5:                                      1
      6:                                      1
      7:                                      1
      8:                                      1
```

Ext

Valid entries	Usage
1 to **8** digits	Enter the extension number.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter a name used to identify the Listed Directory Number

Night Destination

Enter the valid assigned extension number that will receive calls to these numbers when Night Service is active.

Valid entries	Usage
0 to **9** (1 to 8 digits)	For DEFINITY CSI, S87XX Series IP-PNC. Enter a night service extension, a recorded announcement extension, a Vector Directory Number, an individual attendant extension, or a hunt group extension.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

Local Survivable Processor

See **Survivable Processor**.

Locations

Use the **Locations** screen to provide daylight savings time displays to users, to set the area code for each location, and to administer different location information for each location. If the Multiple Locations feature is enabled, you can administer up to 250 location specifications, depending on the configuration of the server that is running Communication Manager. Otherwise, information for Location No.1 applies to all your locations.

Field descriptions for page 1

Figure 474: Locations screen

```
display locations                                                    Page 1 of x
                                 LOCATIONS

               ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc  Name            Timezone Rule   NPA  ARS  Atd Loc  Disp Prefix     Proxy Sel
 No                   Offset              FAC  FAC Parm Parm              Rte Pat
 1:  _______________ _ __:__   __    ___  ____  __  __   __  ___________   ___
 2:  _______________ _ __:__   __    ___  ____  __  __   __  ___________   ___
 3:  _______________ _ __:__   __    ___  ____  __  __   __  ___________   ___
```

ARS FAC

This field is controlled by the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen (use the **`system-parameters customer-options`** command). Administration of this field must follow the same rules that exist for administering an ARS code on the **Feature Access Code (FAC)** screen.

Valid entries	Usage
0 to **9**	Any valid FAC format is acceptable, up to four digits. Characters ***** or **#** are permitted, but only in the first position. Many locations are expected to share the same access code.

ARS Prefix 1 Required for 10-Digit NANP Calls?

Valid entries	Usage
y/n	Enter **y** when a 1 must be dialed before all 10-digit NANP calls.

Attd FAC

The **Attd FAC** field is controlled by the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen (use the **`system-parameters customer-options`** command).

A user cannot administer an Attendant FAC unless an Attendant Access Code has first been administered on either the **Dial Plan Analysis Table** screen or the **Feature Access Code (FAC)** screen.

Note:

Within a dial plan, **FAC/DAC** codes and extensions cannot both start with the same first digits.Either the **FAC/DAC** entries or the block of extensions must be changed to have a different first digit.

Valid entries	Usage
0 to **9**	.Values up to two digits are permitted. Characters ***** or **#** are not permitted. Many locations are expected to share the same access code.

Disp Parm

This field is an index to the corresponding location on the Display Parameters screen. It shows the display parameters for the location.

Loc Parm

This field is an index to the corresponding **Location Parameters n** screens for a specific location. If Multinational Locations is activated, and you enter information into any other field on a location row, you must make an entry in the **Loc. Parms** field. If you don't, an error message displays, and your IP telephones might not be usable.

Valid entries	Usage
1 to **25** or blank	Enter the number of the corresponding Location Parameter set for this location. Default is blank.

Name

Identifies the server running Communication Manager associated with each location number.

Valid entries	Usage
up to 15 alphanumeric characters	A name you use for the location. Names are easier to remember than location numbers. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

NPA

Valid entries	Usage
0 to **9**	Enter the 3-digit numbering plan area code for each location.

Prefix

This field is used for prepending the leading digits for **Uniform Dial Plan Table** screen entries for calls that have, in the Insert Digits field, an **L*x*** value, where ***x*** is the number of leading digits of the Prefix digits to prepend for the location of an originating call. This field is controlled by the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen (use the `system-parameters customer-options` command).

Valid entries	Usage
0 to **9**	Values from one to five digits (0 to 99999) are permitted.

Proxy Sel Rte Pat

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the proxy server. This is the route pattern assigned on the **Route Pattern** screen.

Valid entries	Usage
1 to **999** or blank	Enter the number of the routing pattern to be used to get to the proxy server.

Rule

This field must be filled in for each administered location.

Valid entries	Usage
0	No Daylight Savings
1 to **15** or blank	Specifies the number for each Daylight-Savings Rule (set up on the **Daylight Savings Rule** screen) that is applied to this location.

Timezone Offset

Timezone Offset is actually 3 fields (+/-, hour, and minute) that specify how much time each location differs from the system time. This field must be completed for each administered location. Use +00:00 for the time zone offset for a single location Avaya S8XXX Server.

Valid entries	Usage
+	Shows that the time set on this location is a certain amount of time ahead of the system time.
-	Shows that the time set on this location is a certain amount of time behind the system time.

Valid entries	Usage
0 to **23**	Shows the number of hours difference between this location and system time.

Valid entries	Usage
0 to **59**	Shows the number of minutes difference between this location and system time.

Location Parameters

The **Location Parameters** screen allows you to set or change certain administrable characteristics that determine part of a location's behavior. These include recall timing intervals and loss plans for 2-party and conference calls.

Multiple instances of the **Location Parameters** screen are accessible if the **Multiple Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. If the **Multinational Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**, Location Parameters 2-25 contain the same fields as for Location Parameters 1 (see Figure 475). If the **Multinational Locations** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **n**, the system does not display the following fields for Location Parameters 1:

Tone Generation Plan

DCP Terminal-parameters Plan

Country Code for CDR

Field descriptions for page 1

Figure 475: Location Parameters screen

```
change location-parameters 1                                  Page   1 of   x

                    LOCATION PARAMETERS 1

               Tone Generation Plan: 1
             Analog Ringing Cadence: 1        International Access Code:
           Analog Line Transmission: 1         Local E.164 Country Code:
       DCP Terminal-parameters Plan: 1
               Country code for CDR: 1
                    Companding Mode: Mu-law

RECALL TIMING
                 Flashhook Interval? _                Upper Bound (msec): 1000
           Disconnect Timing (msec): 150              Lower Bound (msec): 200


            Forward Disconnect Timer (msec): 600
                  MF Interdigit Timer (sec): 10
Outgoing Shuttle Exchange Cycle Timer (sec): 4
```

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog telephones in the system

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Options** screen description. For more information, see "Audible Ringing Patterns" in *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207. **Note:** This field must be set to **1** (US) in order for the **Message Waiting Indicator** field on the **Station** screen to be set to **neon**.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Options** screen description.

Companding Mode

Identifies the companding algorithm to be used by system hardware.

Valid entries	Usage
A-Law	Generally used outside the U.S.
Mu-law	Generally used in the U.S.

Country code for CDR

Appears only when the Multinational Locations feature is enabled in the license file.

Valid entries	Usage
1 to **999**	The value in this field corresponds to the country code to be used for Call Detail Recording information for a location. Default is **1**. For a list of country codes, see the Country code table on page 1579.

DCP Terminal-parameters Plan

Appears only when the Multinational Locations feature is enabled in the license file. The value in this field corresponds to the DCP terminal transmission parameters administered for location *n* on the **Terminal Parameters n** screens.

Valid entries	Usage
1 to **25**	Enter terminal-parameters plan number 1 to 25. Default is **1**.

International Access Code

Valid entries	Usage
up to 5 digits (**0** to **9**), or blank	Enter up to 5 digits for the International Access Code. Default is blank.

Local E.164 Country Code

Valid entries	Usage
up to 3 digits (**0** to **9**), or blank	Enter up to 3 digits for the E.164 Country Code. Default is blank. For a list of country codes, see the International Telecommunications Union "List of ITU-T Recommendation E.164 Assigned Country Codes .

Tone Generation Plan

Appears only when the Multinational Locations feature is enabled in the license file. The value in this field corresponds to the tone generation characteristics administered for location *n* on the **Tone Generation *n*** screens.

Valid entries	Usage
1 to **25**	Enter tone-generation plan number 1 to 25. Default is **1**.

RECALL TIMING

Disconnect Timing (msec)

Appears when the **Flashhook Interval** field is **n**.

Valid entries	Usage
80 to **1250** (in increments of 10).	An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect.

Flashhook Interval

Valid entries	Usage
y	Enter **y** to indicate that a flashhook interval (recall window) is required. If a **y** is entered, **Upper Bound** and **Lower Bound** appear.
n	If **n** is entered, **Disconnect Timing** appears.

Forward Disconnect Timer (msec)

Valid entries	Usage
25 to **1500** (in increments of 25).	Specify the duration of a momentary disconnect sent by the server/switch to an analog station user when that user is the last party still off-hook on a call.

Lower Bound (msec)

The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	Specify the lower bound of the flashhook interval.

MF Interdigit Timer (sec)

Applies only to multifrequency signaling trunks. Specify the maximum number of seconds Communication Manager will wait for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires.

Valid entries	Usage
1 to **255**	This number must be less than the number of seconds entered in the short interdigit timer.

Outgoing Shuttle Exchange Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none** on the **Multifrequency-Signaling-Related System Parameters** screen. This field applies only to multifrequency signaling calls made from Avaya Communication Manager.

Valid entries	Usage
1 to **25**	Enter the number of seconds to time an exchange cycle (starts when the far end requests a digit until Avaya Communication Manager sends the requested digit).

Upper Bound (msec)

Specifies the upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. Appears when the **Flashhook Interval** field is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	A flash of 50 msec to 130 msec is always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and will be treated as the digit one.

Field descriptions for page 2

Figure 476: Location Parameters screen - page 2

```
display location-parameters                                        Page  2 of  x
                             LOCATION PARAMETERS

 LOSS PLANS

                    2 Party Loss Plan: 1             Customize? n
                       Tone Loss Plan: 1             Customize? n

 End-to-End total loss (dB) in a n-party conference:
   3: 15   4: 15   5: 15   6: 15                     Customize? n
```

LOSS PLANS

2-Party Loss Plan/Tone Loss Plan

Provides the default values for digital loss plan and for n-party conference loss.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System Parameters Country-Options** screen description. Note that different codes might have similar plans.

Customize

This field appears when the **Digital Loss Plan Modification** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. This setting is controlled by your license file. It enables customization on the corresponding loss plan table. For the **End-to-End total loss (dB) in a n-party conference** field, when **Customize** is set to **y** (yes), the fields can be changed by the administrator. When set to **n**, the **End-to-End total loss (dB) in a n-party conference** fields are reset to the values that they would have had under the **2 Party Loss Plan** administered on page 3 of this screen. They also become display only.

Valid entries	Usage
y/n	Enables customization on the corresponding loss plan table.

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Note:

The End-to-End total loss for multi-party conference calls that is administered on this screen is not always applied to a specific call. For more information on how loss is applied to a multi-party conference call, see the "Loss Plans" feature description in the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Valid entries	Usage
0 to **99**	The higher the number listed for a call with a fixed display number of parties, the more loss Communication Manager adds into a conference call with that number of parties; therefore, the conference call is quieter.

Inter-location Loss Group

Appears only when the Multinational Locations feature is enabled in the license file. When inserting loss for a call, the server treats parties on the call who are in separate locations as if the location with the most parties were connected by an equal number of IP tie trunks as there are parties at other locations. The **Inter-location Loss Group** field specifies the digital loss group number that is used by these "virtual" IP tie trunks.

Valid entries	Usage
1 to **19**	Enter the digital loss group number to use on inter-location calls involving this location. Default is **18**.

Field descriptions for page 3

Figure 477: 2 Party Loss Plan screen

```
change location-parameters                                     Page 3 of x


                              2 PARTY LOSS PLAN

     TO:   1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17  18  19
      1:   0   0   0   0   3   0   0   0   0   3   0   6   6   6   0   3   3   0   0
      2:   0   0   0   0   0   3   3   2   2   3   0   6   6   6   2   3   3   0   0
      3:   0   0   0   0   0   3   3   3   2   3   0   6   6   6   0   3   3   6   6
      4:  15   0   0   0   6   0   0   0   0   3   0   6   6   6   0   3   3   0   0
      5:   0  -3  -3   0   0  -3  -3  -3  -3   0  -3   3   0   0  -3   3   3   0   0
      6:   0   3   3   0   0   6   8   6   5   5   5   9   9   9   5   3   3   0   0
   F  7:   0   3   3   0   0   8   8   6   5   5   5   9   9   9   5   3   3   0   0
   R  8:   0   3   3   0   0   6   6   6   3   5   3   9   6   6   3   3   3   0   0
   O  9:   0   2   2   0   0   5   5   3   0   0   2   3   3   3   9   3   3   0   0
   M 10:   3   3   3   3   3   5   5   5   0   0   3   3   3   3   3   3   3   3   3
     11:   0   0   0   0   0   5   5   3   2   3   0   6   6   3   0   3   3   0   0
     12:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3   6   6
     13:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3   6   6
     14:   0   0   0   0   0   3   3   3  -3  -3  -3   0   0   0   0   3   3   6   6
     15:   0   2   0   0   0   5   5   3   0   3   0   6   6   6   0   3   3   0   0
     16:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
     17:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
     18:   0   0   0   0   3   0   0   0   0   3   0   6   6   6   0   3   3   0   0
     19:   0   0   0   0   0   3   3   2   2   3   0   6   6   6   2   3   3   0   0
```

FROM / TO

Display-only fields that identify the variable digital loss values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Field descriptions for page 4

Figure 478: Tone Loss Plan screen

```
change location-parameters                                            Page 4 of x


                                   TONE LOSS PLAN

                                    TO
               1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17   18  19
        Dial: 0   3   3   0   0   6   6   6   5  0   6   5  5   5   5   0  0    0  0
     Confirm: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
     Reorder: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
        Busy: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
     Ringing: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
   Spec Ring: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
   Intercept: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
     Waiting: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
      Verify: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
     Intrude: 0   0   0   0   0   0   0   0   0   0  0   0  0   0   0   0  0    0  0
         Zip: 3  -3  -3   3   3  -3  -3  -3  -3  -3  3  -3 -3  -3  -3   0  0   -3 -3
       Music: 0   3   3   0   0   6   6   6   3   0  6   3  3   3   3   0  0    0  0
```

FROM / TO

Display-only fields that identify the variable digital tone values.

Valid entries	Usage
-3 through **15**	An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Login Administration

Beginning with Communication Manager 4.0, there is no longer a **Login Administration** screen. For details on screens used for login administration, see *Maintenance Commands for Avaya Communication Manager*, 03-300431, and "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Logging Levels

Use the **Logging Levels** screen to administer logging of SAT activities. You can specify that commands associated with specific actions shown on this screen will be logged by the system. The amount of detail to be logged is the same for all enabled actions and is specified by the **Log Data Values** field on page 1 of this screen.

Note:

The defaults on this screen provide the same amount and type of logging as in Communication Manager releases prior to 4.0.

Field descriptions for page 1

Figure 479: Logging Levels screen - page 1

```
change logging-levels                                       Page   1 of   x

                              LOGGING LEVELS

Enable Command Logging? y
       Log Data Values: none

When enabled, log commands associated with the following values:

           add? y                    export? y                    refresh? y
       busyout? y                       get? n                    release? y
campon-busyout? y                        go? y                     remove? y
        cancel? n                    import? y                      reset? y
        change? y                      list? n                       save? y
         clear? y                      mark? n                        set? y
       disable? y                   monitor? y                     status? n
       display? n                   netstat? n                       test? y
     duplicate? y                    notify? n                 traceroute? n
        enable? y                      ping? n                     upload? n
         erase? y                   recycle? y
```

Enable Command Logging

Valid entries	Usage
y(es)	SAT activity is logged based on the actions selected on the remainder of the **Logging Level** screen.
n(o)	SAT activity is not logged.

Log Data Values

Valid entries	Usage
none	Only the object, the qualifier, and the command action are logged.
new	The new value of any field is logged. The old value is not logged.
both	Both the prior field value and the field value after the change are logged.

Field descriptions for page 2

Figure 480: Logging Levels screen - page 2

```
change logging-levels                                       Page   2 of   x

                               LOGGING LEVELS

    Log All Submission Failures? y
        Log PMS/AD Transactions? n
Log IP Registrations and events? n
   Log CTA/PSA/TTI Transactions? y
```

Log All Submission Failures

When set to **y**, an event is logged when Communication Manager rejects a form submission for any reason, such as an invalid entry in a field or a missing value. When the field is set to **n**, a submission failure is not logged. Form submission failures due to a security violation are always logged and are not affected by this field.

Valid entries	Usage
y/n	Enter **y** to record submission failures on the history log.

Log CTA/PSA/TTI Transactions in History Log

Appears when the **Terminal Translation Initialization (TTI) Enabled** field is **y**. Use this field to record when extensions and physical telephones move between ports without additional administration from the administrator of Avaya Communication Manager.

Valid entries	Usage
y	Enter **y** if you want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.
n	Enter **n** if you do not want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Log IP Registrations and events

Allows the logging of IP registrations in the history log.

Valid entries	Usage
y/n	Enter **y** to record IP registrations on the history log.

Log PMS/AD Transactions

Valid entries	Usage
y/n	Enter **y** to record Property Management System (PMS) and Abbreviated Dialing (AD) events to the log.

Loudspeaker Paging

The **Loudspeaker Paging** screen administers voice paging, deluxe voice paging, and chime paging.

Note:

To set up paging on a H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the **Station** screen. No entries on the **Loudspeaker Paging** screen are required.

Field descriptions for page 1

Figure 481: Loudspeaker Paging screen

```
change paging loudspeaker                                        Page   1 of   x
                             LOUDSPEAKER PAGING

                                   CDR? _
            Voice Paging Timeout (sec): ___
           Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                 Voice Paging      Code Calling
 Zone  Port      TAC  COR  TN      TAC  COR  TN      Location:
   1: _______    ____ __   _       ____ __   _       ___________________________
   2: _______    ____ __   _       ____ __   _       ___________________________
   3: _______    ____ __   _       ____ __   _       ___________________________
   4: _______    ____ __   _       ____ __   _       ___________________________
   5: _______    ____ __   _       ____ __   _       ___________________________
   6: _______    ____ __   _       ____ __   _       ___________________________
   7: _______    ____ __   _       ____ __   _       ___________________________
   8: _______    ____ __   _       ____ __   _       ___________________________
   9: _______    ____ __   _       ____ __   _       ___________________________
 ALL: _______    ____ __   _       ____ __   _
```

CDR

This field determines whether CDR data is collected for the paging ports.

Valid entries	Usage
y/n	Enter **y** if you want the server running Communication Manager to collect CDR data on the paging ports.

Code Calling — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **995**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Code Calling Playing Cycles

This field sets the number of times a chime code will play when a user makes a chime page. To determine the best setting, consider who your code calling users are and whether they are likely to hear the code chime the first time.

Valid entries	Usage
1 to **3**	Enter the number of times you want the chime code to play when a user makes a page.
blank	The field cannot be blank when you administer chime paging (code calling).

Code Calling — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users can activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*	Can be used as first digit.
#	Can be used as first digit.
blank	Leave this field blank for unused paging zones.

Code Calling — TN

Valid entries	Usage
1 to **20** (DEFINITY CSI)	If your system uses Tenant Partitioning, you can use this field to assign a paging zone to a specific tenant partition.
1 to **100** (S87XX/S8300 Servers)	

Location

Valid entries	Usage
1 to 27 characters	Assign a descriptive name for the physical location corresponding to each zone. Typical entries might be "conference room A," "warehouse," or "storeroom."

Port

This field assigns a port on an auxiliary trunk circuit pack to a paging zone. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit
blank	Leave this field blank for unused paging zones.

Voice Paging — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 to **995**	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Voice Paging — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the **ALL** field, users can activate speakers in all the zones by dialing that code.

Valid entries	Usage
1 to 4 digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*	Can be used as first digit.
#	Can be used as first digit.
blank	Leave this field blank for unused paging zones.

Voice Paging Timeout (sec)

This field limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages you expect to broadcast and then add another 4 to 5 seconds.

Valid entries	Usage
10 to **600** seconds	Enter the maximum number of seconds you want any page to last.
blank	The field cannot be blank when you administer voice paging.

Note:

To use a port that has no hardware associated with it, place an **x** in this field.

Voice Paging — TN

Valid entries	Usage
1 to **20** (DEFINITY CSI)	If your system uses Tenant Partitioning, you can use this field to assign a paging zone to a specific tenant partition.
1 to **100** (S8300/S87XX Servers)	

Related topics

See Setting up Voice Paging Over Loudspeakers on page 515 for instructions.

See "Loudspeaker Paging" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Maintenance-Related System Parameters

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

MCT Group Extensions

See **Extensions Administered to have an MCT-Control Button**.

Media-Gateway

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

Mode Code Related System Parameters

This screen establishes parameters associated with the Mode Code Voice Mail System Interface.

Note:

You can only administer this screen if the **Mode Code Interface** field on the **Feature-Related System Parameters** screen is set to **y**.

Field descriptions for page 1

Figure 482: Mode Code Related System Parameters screen

```
change system-parameters mode-code                              Page   1 of x

                   MODE CODE RELATED SYSTEM PARAMETERS

MODE CODES  (FROM SWITCH TO VMS)
      Direct Inside Access: __
Direct Dial Access - Trunk: __
         Internal Coverage: __
         External Coverage: __

           Refresh MW Lamp: __

     System In Day Service: __
   System In Night Service: __

                     OTHER RELATED PARAMETERS
DTMF Duration On (msec): __ Off (msec): __ Sending Delay (msec):__
 VMS Hunt Group Extension : ____
Remote VMS Extensions - First:     Second:
```

MODE CODES (FROM SWITCH TO VMS)

Direct Dial Access - Trunk

This value defines a mode code that the Avaya S8XXX Server sends when an external caller dials the VMS access number.

Valid entries	Usage
0 to **9**, **#**, *, **#00**	Up to six digits that can include these characters.

Direct Inside Access

This value defines a mode code that the Avaya S8XXX Server sends when a caller at an internal extension dials the Voice Mail System (VMS) access number.

Valid entries	Usage
0 to **9**, **#**, *, **#00**	Up to six digits that can include these characters.

External Coverage

This value defines a mode code that the Avaya S8XXX Server sends when an external caller tries to reach a user at another extension and the call goes to the user's voice-mail coverage.

Valid entries	Usage
0 to **9**, **#**, *, **#00**	Up to six digits that can include these characters.

Internal Coverage

This value defines a mode code that Communication Manager sends when an internal caller tries to reach a user at another extension and the call goes to the user's voice mail coverage.

Valid entries	Usage
0 to **9**, **#**, *, **#00**	Up to six digits that can include these characters.

Refresh MW Lamp

This value defines a mode code that Communication Manager sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that can include these characters.

System In Day Service

This value indicates to the VMS that the Avaya Communication Manager has changed from Night to Day Service.

Valid entries	Usage
0 to **9**, **#**, *****, **#11**	Up to six digits that can include these characters.

System In Night Service

This value indicates to the VMS that the Avaya Communication Manager has changed from Day to Night Service.

Valid entries	Usage
0 to **9**, **#**, *****, **#12**	Up to six digits that can include these characters.

OTHER RELATED PARAMETERS

DTMF Duration On

Valid entries	Usage
Between **75** and **500** in multiples of 25	Define in milliseconds the length of mode code digits sent to the VMS. This field cannot be blank.

Off

Valid entries	Usage
Between **75** and **200** in multiples of 25	Define in milliseconds the pause between mode code digits as they are sent to the VMS. This field cannot be blank.

Remote VMS Extensions - First

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. Specifies the first remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the first UDP VMS hunt group extension.

Remote VMS Extensions - Second

You can administer this field if the **Mode Code for Centralized Voice Mail** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. Specifies the second remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the second UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension.

Sending Delay

Valid entries	Usage
75 to **1000** in multiples of 25	Define in milliseconds the delay between the time answer supervision is received from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Valid entries	Usage
Valid extension.	Enter the extension of a hunt group containing VMI extensions.

Modem Pool Group

There are two types of conversion resources for Modem Pooling. The first type, an *integrated conversion resource*, is a circuit pack that emulates a Trunk Data Module connected to a 212A-type modem. Two conversion resources are on each circuit pack.

The second type, a *combined conversion resource*, is a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource can be either a Modular Trunk Data Module (MTDM) or 7400A Data Module and connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port.

Field descriptions for page 1

Figure 483: Modem Pool Group screen — if Group Type is integrated

```
change modem-pool num                                          Page 1 of x
                                   MODEM POOL GROUP
                    Group Number: 1                   Group Type: integrated
Receiver Responds to Remote Loop? n              Hold Time (min): 5
           Send Space Disconnect? y     Receive Space Disconnect? y
                    CF-CB Common? y   Loss of Carrier Disconnect? y

Speed: LOW/300/1200      Duplex: full     Synchronization: a/sync

CIRCUIT PACK ASSIGNMENTS
               Circuit Pack   Circuit Pack
               Location       Location
               1: ___          9: ___
               2: ___         10: ___
               3: ___         11: ___
               4: ___         12: ___
               5: ___         13: ___
               6: ___         14: ___
               7: ___         15: ___
               8: ___         16: ___
```

Figure 484: Modem Pool Group screen — if Group Type is combined

```
change modem-pool num                                           Page 1 of x
                                 MODEM POOL GROUP
                   Group Number: _            Group Type: combined
                     Modem Name: ______ Hold Time (min): 5_
                     Time Delay: 0_            Direction: two-way
Answer Supervision Timeout(sec): _

     Speed: LOW/300/1200___     Duplex: full     Synchronization: async

PORT PAIR ASSIGNMENTS
       Analog Digital    Analog Digital     Analog Digital      Analog Digital
     1: _____ _____   9: _____ _____    17: _____ ______    25: _____ _____
     2: _____ _____  10: _____ _____    18: _____ ______    26: _____ _____
     3: _____ _____  11: _____ _____    19: _____ ______    27: _____ _____
     4: _____ _____  12: _____ _____    20: _____ ______    28: _____ _____
     5: _____ _____  13: _____ _____    21: _____ ______    29: _____ _____
     6: _____ _____  14: _____ _____    22: _____ ______    30: _____ _____
     7: _____ _____  15: _____ _____    23: _____ ______    31: _____ _____
     8: _____ _____  16: _____ _____    24: _____ ______    32: _____ _____
```

Note:

The **Speed**, **Duplex**, and **Synchronization** fields cannot be filled out for the "integrated" pooled modem screens but can be assigned on the "combined" pooled modem screen. The integrated conversion resource automatically will adjust its speed and synchronization to the endpoint it is connected to. In synchronous mode, the integrated modem pool can operate at 1200 baud. In asynchronous mode, it can operate at 300 or 1200 baud. Full-duplex operation is always used.

Answer Supervision Timeout (sec)

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait before the far-end answers.
0	No answer supervision

CF-CB Common

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to indicate that the CF and CB leads on the conversion resource are logically connected.

Direction

Enter the direction of the call for which modem pool will operate. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
incoming	Converts an analog signal to digital for the data endpoint.
outgoing	Converts analog to digital (or digital to analog) for data calls.
two-way	Allows incoming and outgoing data communication.

Duplex

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the duplex mode of the conversion resources in the group.

Valid entries	Usage
full	Can talk and listen at the same time.
half	Cannot talk and listen at the same time.

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

This field designates what physical model pool you are going to.

Valid entries	Usage
integrated	Maps to the Pooled Modem circuit pack.
combined	Maps to an external modem pool (when you have a data module and a modem).

Hold Time (min)

Valid entries	Usage
1 to **99**	Enter the maximum number of minutes that a conversion resource in the group can be held while a call waits in a queue or reserved after Data Call Preindication.

Loss of Carrier Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to permit conversion resource to disconnect if it detects a dropped carrier.

Modem Name

Indicates the name of the modem pool. This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
1 to 6 alphanumeric character string	Enter the name of the modem pool.

Receive Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to disconnect after receiving 1.6 seconds of space.

Receiver Responds to Remote Loop

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the far-end modem to put conversion resource into loop back mode.

Send Space Disconnect

This field appears only when the **Group Type** field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to send 4 seconds of space before disconnecting.

Speed

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the communication speed in bits per second of the conversion resources in the group. Enter one to three speeds separated by slashes (for example, 300/1200/2400) to indicate a maximum of three running speeds.

Valid entries	Usage
LOW	0 to 300 blind sampled
300	
1200	
2400	
	1 of 2

Valid entries	Usage
4800	
9600	
19200	
	2 of 2

Synchronization

Display-only when the **Group Type** field is **integrated**. When the **Group Type** field is **combined**, enter the synchronization mode of the conversion resources in the group.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

CIRCUIT PACK ASSIGNMENTS are optional on "integrated" conversion resource screens only.

Time Delay

This field appears only when the **Group Type** field is **combined**.

Valid entries	Usage
0 to **255**	Enter the time delay in seconds to insert between sending the ringing to the modem and the off-hook alert to the data module.

CIRCUIT PACK ASSIGNMENTS

Circuit Pack Location

Displays when the **Group Type** field is **integrated**. Enter the port associated with the conversion resource on the integrated modem pool circuit pack. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

PORT PAIR ASSIGNMENTS are optional on "combined" pooled modem screens only.

PORT PAIR ASSIGNMENTS

Analog Digital

Displays when the **Group Type** field is **combined**. Enter the port numbers of the modem/TDM pair in a conversion resource.

Two port entries are required. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITY CSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Note:

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

MOH Group

Use the **MOH Group** screen to define a collection of analog station and/or aux trunk port circuit pack ports that are connected to external audio sources for use with the Music on Hold feature.

Field descriptions for page 1

Figure 485: MOH Group screen

```
change moh-analog-group n                                          Page 1 of x
                              MOH Group 2

                            Group Name:

MOH SOURCE LOCATION
     1:         16:         31:         46:         61:
     2:         17:         32:         47:         62:
     3:         18:         33:         48:         63:
     4:         19:         34:         49:         64:
     5:         29:         35:         50:         65:
     6:         21:         36:         51:         66:
     7:         22:         37:         52:         67:
     8:         23:         38:         53:         68:
     9:         24:         39:         54:         69:
    10:         25:         40:         55:         70:
    11:         26:         41:         56:         71:
    12:         27:         42:         57:         72:
    13:         28:         43:         58:         73:
    14:         29:         44:         59:         74:
    15:         30:         45:         60:         75:
```

MOH Source Location

Type in the Music-on-hold analog or aux-trunk port location: enter the port; cabinet(1-64):carrier(A-E):slot(1-20):circuit(1-31) OR gateway(1-250):module(V1-V9):circuit(1-31).

Group Name

Enter an alpha-numeric name of the MOH group for identification.

Multifrequency-Signaling-Related Parameters

This screen sets the system or location parameters associated with multifrequency signaling. With the Multinational Locations feature enabled, multifrequency signaling can be administered per location, rather than system-wide. This screen appears when **Incoming Call Type** is **group-ii-mfc** and **Outgoing Call Type** is **none**. Page 2 of this screen appears when both **Incoming Call Type** and **Outgoing Call Type** are **group-ii-mfc**.

If the field **Use COR for All Group II Responses** is set to **y**, the **Group II Called Party Category** and **Use COR for Calling Party Category** fields do not appear.

Note:

With the Multinational Locations feature enabled, you can assign MFC signal sets per trunk group, rather than system-wide.

Field descriptions for page 1

Figure 486: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling                                 Page  1 of  X

             MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

          Incoming Call Type:                         ANI Prefix:
          Outgoing Call Type:                        Default ANI:
       Maintenance Call Type:             NEXT ANI DIGIT
                                                        Incoming:
     Maximum Resend Requests:                           Outgoing:
   Received Signal Gain (dB):
Transmitted Signal Gain (dB):

                           Request Incoming ANI (non-AAR/ARS)?
                    Outgoing Forward Signal Present Timer (sec):
                     Outgoing Forward Signal Absent Timer (sec):
MF Signaling Intercept Treatment - Incoming? _  Outgoing: ____________ _____
                              Collect All Digits Before Seizure?
                   Overlap Sending on Link-to-Link Tandem Calls?
          Private Group II Permissions and Public Interworking?
                              Convert First Digit End-of-ANI To: _
                             Use COR for All Group II Responses? _
                                  Group II Called Party Category:
                             Use COR for Calling Party Category?
```

The **ANI Prefix**, **Default ANI**, and **Collect All Digits Before Seizure** fields appear only when the value of the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

If **Collect All Digits Before Seizure** is **y**, **Overlap Sending on Link-to-Link Tandem Calls and Convert First Digit End-of-ANI** are not displayed.

Default ANI

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
2 to **15**	Enter the PBX identification number that is sent to the CO when ANI is requested (by the CO) on a particular call but is not available, such as on tandem tie trunk calls.
blank	Use for tandeming. If this field is blank, you must enter a value in the **ANI-Not-Available** field.

ANI Prefix

This field appears only when **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
1 to 6 digits or blank	Enter the prefix to apply to an extension when ANI is sent to the CO.

Backward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to send the check frequency after receiving an MFE signal.

Collect All Digits Before Seizure

Appears when the **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Valid entries	Usage
y	The system collects all the digits before seizing the trunk and the **ANI Req** field on the **AAR and ARS Digit Conversion Table** does not apply.
n	Enter **n** to control ANI collection via the **ARS** screens.

Convert First Digit End-of-Dial To

Appears when the **Private Group II Permissions and Public Interworking** field is **y**.

Valid entries	Usage
0 to **9**, **#**, or blank	Enter the digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Forward Cycle Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to receive the check frequency after sending an MFE signal. Communication Manager drops the call if the time runs out before it receives check frequency.

Group II Called Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Enter the type of group II signals that should be used on the outgoing R2-MFC call.

Valid entries	Usage
user-type	The type of telephone making the call determines the type of group II signal that the server sends (normal = ordinary telephone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of group II signal that the server sends.

Incoming Call Type

This field defines the signal type that a CO uses to place an incoming call to the server.

Valid entries	Usage
group-ii-mfc	If the value of this field is **group-ii-mfc**, the second page of the screen will display entries for all group-I, group-II, group-A, and group-B signal types with a set of default values (see page 2 of screen).
non-group-ii-mfc	If the value is **non-group-ii-mfc**, the second page displays only group-I and group-A signal types.
mfe	Use only in Spain (multi-frequency Espanol)

Incomplete Dial Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
45 to **255**	Enter the number of seconds to wait from the start of a call until the end of the check frequency of the last signal. Communication Manager drops the call if the time runs out before it receives the check frequency.

Maintenance Call Type

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

Valid entries	Usage
1	The Belgium maintenance sequence is indicated when the CO sends an MFC maintenance tone.
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.
none	

Maximum Resend Requests

Valid entries	Usage
1 to **99**	Enter the threshold number of resend type MFC signals your server running Communication Manager accepts during an outgoing call.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

MF Signaling Intercept Treatment - Incoming

Valid entries	Usage
y	Send the group B signal for the intercept to the CO and play intercept tone on the trunk.
n	Use normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Displays when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

MFE Type

This field only appears when **Incoming Call Type** is **mfe**.

Valid entries	Usage
2/5	Determines which public signaling Communication Manager will use.
2/6	

Outgoing Call Type

This field defines the signal type that the PBX uses to place an outgoing call into a CO.

Valid entries	Usage
group-ii-mfc	If the content of this field is **group-ii-mfc**, the system displays the third page of the screen. The third page displays entries for all group-I, group-II group-A, and group-B signal types with a set of default values.
mfe	Use only in Spain (multi-frequency Espanol)
none	If the content of this field is **none**, the system does not display the third page. In addition, **Outgoing Forward Signal Present Timer**, **Outgoing Forward Signal Absent Timer**, **ANI Prefix**, **ANI for PBX**, **Next ANI Digits**, and **Collect All Digits Before Seizure** will not display on field descriptions for page 1.

Outgoing Forward Signal Absent Timer (sec)

This field appears only when the content of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
11 to **255**	Enter the maximum number of seconds to elapse between forward signals on outgoing calls. The timer starts (and restarts) when a forward tone is taken off the link and it stops when the next forward tone is applied to the link.

Outgoing Forward Signal Present Timer (sec)

This field appears only when the value of **Outgoing Call Type** is **group-ii-mfc**.

Valid entries	Usage
1 to **255**	Enter the maximum number of seconds to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when Communication Manager begins sending a forward signal and stops when Communication Manager receives the backward signal.

Outgoing Start Timer (sec)

Appears when the **Incoming Call Type** field is **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to time from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Overlap Sending on Link-to-Link Tandem Calls

Does not appear if the **Collect All Digits Before Seizure** field is **y**. When Avaya Communication Manager has this field set to **y**, and calls are tandeming between servers, then ANI for PBX will be sent to the terminating switch if that switch requests ANI before Avaya Communication Manager receives it from the originating server/switch. The terminating server/switch can request ANI before the receipt of the last address digit if it is not running Avaya Communication Manager, or if it is Avaya Communication Manager with the **Request Call Category at Start of Call** field set to **y**.

Valid entries	Usage
y/n	If **y**, Avaya Communication Manager sends and receives digits one digit at a time instead of enbloc. (With enbloc, digits are not sent until the entire group of digits is received).

Private Group II Permissions and Public Interworking

Displays when the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **none**.

Valid entries	Usage
y/n	If **y**, then Avaya Communication Manager: ● Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls. ● Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls. ● Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls. ● Checks station permissions if you call forward off-net.

Received Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the MFC port listens to the trunk port. Communication Manager listens with a range of -5 to -35 and this value moves the range (for example, a value of -5 provides a range of -10 to -40). This value also applies to Russian MF Shuttle trunks.

Request Incoming ANI (non-AAR/ARS)

Appears when the **Incoming Call Type** field is **group-ii-mfc** or **mfe** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**. This field only applies if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Valid entries	Usage
y/n	If **y**, ANI should be requested on incoming R2-MFC calls.

Transmitted Signal Gain (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE, and this field adds gain or loss to the starting value of -5. This value also applies to Russian Shuttle trunks and Russian multi-frequency ANI.

Use COR for All Group II Responses

Appears if the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
y/n	Enter **y** to allow the COR administered category to be used for both the calling party and called party categories.

Use COR for Calling Party Category

Appears when the **Outgoing Call Type** field is **group-ii-mfc** and the **Use COR for All Group II Responses** field is **n**. Indicates the category to send with ANI if requested on an outgoing R2-MFC call.

Valid entries	Usage
y	Use the calling facility's COR to determine category.
n	Use the calling party's user-type COR to determine category.

NEXT ANI DIGIT

Incoming

Appears when the **Incoming Call Type** field is **group-ii-mfc** and the **Outgoing Call Type** field is **group-ii-mfc** or **mfe**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the Next ANI Digit signal will be the same as the "send-ani" signal or the "next-digit" signal or another signal defined as "next_ani_digit."

Outgoing

Appears when the **Outgoing Call Type** field is **group-ii-mfc**.

Valid entries	Usage
next-digit **next_ani_digit** **send-ani**	Enter a value to determine whether the Next ANI Digit signal will be the same as the "send-ani" signal or the "next-digit" signal or another signal defined as "next_ani_digit."

Field descriptions for page 2

The fields on Page 2 define call category and ANI information. For India, the ANI can be requested without the call category information.

Figure 487: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling n                          Page   2 of   x
             MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS  1

     Request Call Category at Start of Call? n            Outgoing II by COR
           Restart ANI from Caller Category? y             1:  1
              Number of Incoming ANI Digits: 0             2:  2
              Number of Outgoing ANI Digits: 0             3:  3
             Truncate Station Number in ANI: no            4:  4
Address Digits Include End-of-digits Signal? n             5:  5
         Call Category for Vector ii-digits? n             6:  6
               Request CPN at Start of Call? n             7:  7
          Do Not Send Group B Signals to CO? n             8:  8
   ANI Source for Forwarded & Covered Calls: caller        9:  9
                                                          10: 10
                        INCOMING    OUTGOING
          ANI Available:
      ANI Not Available:
```

Address Digits Include End-of-Digits Signal

Indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the Central Office for outgoing calls.

Valid entries	Usage
y/n	Enter **y** to send an outgoing forward Group I end-of-digit signal after completion of address digits upon request from the Central Office for outgoing calls.

ANI Source for Forwarded & Covered Calls

Valid entries	Usage
caller	Send the calling party's ANI when calls are redirected.
forwarder	Send the forwarding party's ANI when calls are redirected.

Call Category for Vector ii-digits

Allows you to use the call category digit as the ii-digits on call vector steps.

Valid entries	Usage
y/n	If **y**, the call category digit, which is a part of ANI, is used as the ii-digits on call vector steps.

Do Not Send Group B Signals to CO

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. This field allows completion of a call without Group-B signals.

Valid entries	Usage
y	If **y**, does not send Group-B signals to complete an incoming call.
n	If **n**, sends Group-B signals to complete an incoming call.

Number of Incoming ANI Digits

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls - Group I, Avaya Communication Manager appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected.

Outgoing II by COR

Appears only if either **Use COR for Calling Party Category** or **Use COR for All Group II Responses** on page 1 are set to **y**. The Group II signal sent to the CO on outgoing calls can be administered per COR (Class of Restriction) and per trunk group. The Group II signal is administered per COR. That per-COR value in turn can be mapped into a possibly different outgoing signaling parameter set. The values in the **Outgoing II by COR** fields administer that outgoing mapping.

Valid entries	Usage
1 to **10**	Enter a number between 1 and 10 that maps to the Group II signal Communication Manager sends to the CO on outgoing calls

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information will be sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

Valid entries	Usage
y/n	If **y**, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Avaya Communication Manager sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Communication Manager sends the Send-ANI backward signal upon receipt of the first address signal.

Request CPN at Start of Call

This field appears only if the **Incoming Call Type** field is **group-ii-mfc**. Provides for Communication Manager to collect ANI and call category immediately after receipt of the first address digit.

Valid entries	Usage
y/n	If **y**, provides ANI (Calling Party Number (CPN) and call category) immediately after receiving the first address digit.

Restart ANI from Caller Category

Display-only field.

Valid entries	Usage
y/n	If **y**, Avaya Communication Manager sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Truncate Station Number in ANI

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
beginning **ending** **no**	This field defines the side of the extension number from which to truncate when station ANI is sent to the CO and the combined length of the ANI prefix and extension number is greater than Number of Outgoing ANI Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if ANI for PBX is sent.

INCOMING / OUTGOING

ANI Available

Valid entries	Usage
1 to **15** or blank	Enter the number of incoming and outgoing ANI-Available signals.

ANI Not Available

You must enter a value if the **ANI** for **PBX** field is blank.

Valid entries	Usage
1 to **15** or blank	Enter the number of the incoming and outgoing ANI-Not-Available signals. Communication Manager outpulses the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Communication Manager outpulses the ANI-Not-Available forward signal to the CO on outgoing calls where ANI is not possible.

Field descriptions for page 3

The fields shown on Page 3 of the **Multifrequency-Signaling-Related System Parameters** screen define the meaning of MFC tones for calls originated at the CO. See Definitions of Group I, II, A, and B signals on page 1376 for more information. This screen appears only if the **Incoming Call Type** field is **group-ii-mfc** or **non-group-ii-mfc**.

When the screen initially appears, either of two sets of default values is possible. One set is for the group II call type; the other set is for non-group II call type. In each set, the default value for each field is set to the most common value.

The **Multifrequency-Signaling-Related Parameters** screen shows the defaults when the **Incoming Call Type** field is **group-ii-mfc**. A variation appears if **Incoming Call Type** is **non-group-ii-mfc**. When **Incoming Call Type** is **non-group-ii-mfc**, group II and group B columns do not appear.

Figure 488: Multifrequency-Signaling-Related Parameters screen

```
change multifrequency-signaling                                 Page 3 of X
               MULTIFREQUENCY-SIGNALING-RELATED PARAMETERS

  INCOMING FORWARD SIGNAL TYPES           INCOMING BACKWARD SIGNAL TYPES
  (Tones from CO)                         (Tones to CO)

        Group-I             Group-II            Group-A             Group-B
   11: ignored          1: normal           1: next-digit      1: free
   12: ignored          2: normal           3: end-of-dial     2: busy
   13: ignored          3. normal          __:                 4: congestion
   14: ignored          4: normal          __: ________________7: intercept
   15: ignored          5: normal          __: ________________ __: __________
                        6: normal          __: ________________ __: __________
                        7: normal          __: ________________ __: __________
                        8: normal          __: ________________ __: __________
                        9: normal          __: ________________ __: __________
                       10: normal          __: ________________ __: __________
                       11: normal          __: ________________ __: __________
                       12: normal          __: ________________ __: __________
                       13: normal          __: ________________ __: __________
                       14: normal          __: ________________ __: __________
                       15: normal          __: ________________ __: __________
```

INCOMING FORWARD SIGNAL TYPES (Tones from CO)

Group I

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1376 for signal type.

Valid entries	Usage
drop	If **Incoming Call Type** is **group-ii-mfc**
ani-avail	
end-of-ani	
end-of-dial	
ignored	
maint-call	
ani-not-avail	
send-congest	
drop	If the **Incoming Call Type** is **non-group-ii-mfc**
ignored	

Group II

Message codes 1 to 15 display. Assign a meaning to each code.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
busy-rt-attd	
data-call	
data-verify	
drop	
	1 of 2

Valid entries	Usage
maint-call	
send-intercept	
toll-auto	
toll-operator	
normal	
	2 of 2

INCOMING BACKWARD SIGNAL TYPES (Tones to CO)

Group A

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code.

Valid entries	Usage
congestion	See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
end-of-dial	
intercept	
next-ani-digit	
next-digit	
send-ani	
setup-sppath	

Group B

This field does not appear if the **Do Not Send Group B Signals to CO** field is **y**. Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
busy	See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
congestion	
free	
mct	
tariff-free	
tie-free	
toll-busy	
intercept	

Field descriptions for page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the PBX. See Definitions of Group I, II, A, and B signals on page 1376 for more information.

Page 4 of the **Multifrequency-Signaling-Related System Parameters** screen only appears if **Outgoing Call Type** is **group-ii-mfc** or **mfe**.

Figure 489: Multifrequency-Signaling-Related System Parameters screen

```
change system-parameters multifrequency-signaling                  Page 4 of x
                MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

  OUTGOING FORWARD SIGNAL TYPES           OUTGOING BACKWARD SIGNAL TYPES
  (Tones to CO)                           (Tones from CO)

        Group-I             Group-II            Group-A             Group-B
   11: ______________   2: normal           1: next-digit       1: free
   12: ______________   5: attendant        2: congestion       2: busy
   13: ______________   6: data-call        3: end-of-dial      3: congestion
   14: ______________  __: ______________   4: congestion       4: congestion
   15: ______________  __: ______________   5: call-info-ani    5: congestion
                       __: ______________   6: congestion       6: free
                       __: ______________   7: last-2-digits    7: intercept
                       __: ______________   8: last-3-digits    8: congestion
                       __: ______________   9: congestion       9: congestion
                       __: ______________  10: congestion      10: congestion
                       __: ______________  11: congestion      11: congestion
                       __: ______________  12: congestion      12: congestion
                       __: ______________  13: congestion      13: congestion
                       __: ______________  14: congestion      14: congestion
                       __: ______________  15: congestion      15: congestion
```

OUTGOING FORWARD SIGNAL TYPES (Tones to CO)

Group I

Message codes 11 to 15 appear. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.)

Valid entries	Usage
end-of-digits	Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
ani-avail	
end-of-ani	
ani-not-avail	

Group II

Message codes between 1 and 15 display. Assign a meaning to each code. Each entry can only appear once in the group II column.

Valid entries	Usage
attendant	See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
data-call	
toll-auto	
normal	

OUTGOING BACKWARD SIGNAL TYPES (Tones from CO)

Group A

Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
send-ani	See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
congestion	
drop	
end-of-dial	
last-2-digits	
last-3-digits	
last-digit	
next-ani-digit	
next-digit	
restart	
intercept	
	1 of 2

Valid entries	Usage
resend-digit	
setup-sppath	
	2 of 2

Group B

Valid entries	Usage
busy	Message codes between 1 and 15 display. Assign a meaning to each code. See Definitions of Group I, II, A, and B signals on page 1376 for signal type.
congestion	
free	
mct	
tariff-free	
toll-busy	
intercept	

Definitions of Group I, II, A, and B signals

Group I signals

Group I signals are a set of forward signals generated by the originating Avaya S8XXX Server.

ani-avail

Used in Hungary. If this signal is defined and ANI is requested on outgoing R2-MFC calls, ANI is sent to the CO before ANI caller digits are sent. This signal is sent after the ANI caller category signal.

ani-not-avail

Used on DOD calls in Brazil and Columbia. Communication Manager sends this signal to the CO when it receives an ANI request and the caller's number is not available.

digits 1 to 10

The signals from group I.1 to I.10 are reserved for address digits 0 to 9.

drop

When this signal is received from the CO, Avaya Communication Manager starts the disconnect sequence and drops the call.

end-of-ani

This signal is used on DOD and DID calls. Communication Manager sends this signal to indicate the end-of-ANI digits when ANI digits are sent to the CO.

end-of-dial

This signal is used when open numbering is used on DID calls. The CO sends this signal to indicate the end-of-dial digits and Communication Manager responds with a request for a group II signal.

end-of-digits

This signal is sent by the originating Avaya S8XXX Server that makes outgoing calls, sends digits, and receives a next-digit group A signal from the destination server or switch when there are no more digits to be sent.

This signal is also sent when Communication Manager does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani group A signal from the destination end when there are no more ANI digits to be sent.

If both end-of-digits and end-of-ani are assigned, Communication Manager uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

ignored

If this signal is received from the CO, Communication Manager sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO. This signal can be used on DID calls in Saudi Arabia.

send-congestion

When Communication Manager receives this signal from the CO on a DID call, it returns a congestion signal (group A), in compel (not pulse) mode, to the CO.

Group II signals

Group II signals are a more elaborate set of forward signals generated by the originating server.

attendant

If Communication Manager receives this signal on DID calls, the call terminates at an attendant regardless of the extension dialed. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.

busy-rt-attd

If Communication Manager receives this signal on DID calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.

data-call

This signal is treated the same as the data-verify signal except that it does not require a terminating extension to be a data extension.

data-verify

If Communication Manager receives this signal on DID calls and the terminating extension is not a data extension, it sends intercept treatment. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.

drop

When this signal is received from the CO, Communication Manager starts the disconnect sequence and drops the call.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and Avaya Communication Manager prepares the special maintenance call sequences for the CO.

normal

This signal indicates that the caller is a normal subscriber. If it is received on a DID call, the call is terminated at the called extension. For an outgoing MF signaling call that uses group II signaling, this signal is sent to the CO when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and DOD calls.

send-intercept

If Communication Manager receives this signal from the CO on a DID call, it returns group B intercept signal to the CO.

toll-auto

This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the CO. You can define the special busy signal by choosing the option toll-busy on the incoming group B signals.

toll-operator

This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

Group A signals

Group A signals are backward signals generated by the destination server/switch.

send-ani

The CO sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on DID calls. This signal is used on both DOD and DID calls.

congestion

The CO sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.

drop

When this signal is sent, the receiving end starts the disconnect sequence.

end-of-dial

This signal is sent to indicate the end of the address digit string. For MF group II calls, this signal requests a group II signal and switches the sender over to the group B signaling mode. This signal is used on both DID and DOD calls.

intercept

The CO sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on DOD calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.

resend-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on DOD calls.

last-digit

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 2 digits can be resent. This signal is used on DOD calls.

last-2-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 3 digits can be resent. This signal is used on DOD calls.

last-3-digits

Communication Manager sends this signal to adjust the outpulsing pointer so that the last 4 digits can be resent. This signal is used on DOD calls.

next-digit

Communication Manager sends this signal to request the next digit. This signal is used on both DID and DOD calls.

next-ani-digit

Communication Manager sends this signal to request the next ANI digit. This signal is used on DID and DOD calls.

restart

Communication Manager sends this signal to request the whole digit string again. This signal is used on DOD calls.

setup-sppath

The CO sends this signal to Communication Manager to set up a speech path. This signal is used on DOD calls and on DID calls in Belgium.

Group B signals

Group B signals enhance group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses group II signals, the destination end answers with group B signals. Group B signals are as follows:

busy

This signal is sent to indicate that the called party is busy. On DID calls, the signal is sent to the CO if there is no coverage point to terminate the call. If Communication Manager receives this signal on DOD calls, it plays busy tone to the calling party and drops the trunk.

congestion

This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On DID calls, the signal is sent to the CO to indicate that a resource is not available. On DOD calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.

free

This signal indicates that the called party is idle. On DID calls, the signal is sent to the CO to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on DOD calls, it connects the trunk to the calling party.

intercept

This signal indicates that the called party number is not in service or is not correct. On DID calls, if intercept treatment is set to provide a tone, tone is sent to the CO to indicate that the called number is not valid. If Communication Manager receives the signal on DOD calls, it plays intercept tone to the calling party and drops the trunk.

mct

This signal identifies the call as one that needs to be traced by the CO. Avaya Communication Manager then generates an MFC Call Trace Backward Signal (administered on the **Multifrequency-Signaling-Related System-Parameters** screen) during call setup instead of the "free" signal. If the terminating station's COR has this feature set to **y**, the CO collects trace information before releasing the calling party.

Note:

If the station's COR has **MF Incoming Call Trace** set to **y** and the "mct" signal is not defined, then the "free" signal is sent.

tariff-free

This signal is sent when the trunk group provides an "800" service. Avaya Communication Manager generates an MFC tariff-free backward signal (administered on the **System-Parameters Multifrequency-Signaling** screen) during call setup instead of the "free" signal, facilitating CO billing.

Note:

If the trunk is administered as a tariff-free trunk and the "tariff-free" signal is not defined, then the "free" signal is sent.

tie-free

This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.

toll-busy

This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Multiple Level Precedence & Preemption (MLPP) Parameters

Use this screen to set up system parameters for the Multiple Level Precedence & Preemption feature.

Field descriptions for page 1

Figure 490: Multiple Level Precedence and Preemption Parameters screen

```
change system-parameters mlpp                                  Page  1 of  x

                 MULTIPLE LEVEL PRECEDENCE & PREEMPTION PARAMETERS
  ANNOUNCEMENTS
          Blocked Precedence Level: 6801          Service Interruption: 6803
     Unauthorized Precedence Level: 6802            Busy, Not Equipped: 6804
                       Vacant Code: 6805

  PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT
     Flash Override: 0  Flash: 1  Immediate: 2  Priority: 3  Routine: 4

         Attendant Diversion Timing (sec): 60
            Remote Attendant Route String:
     Worldwide Numbering Dial Plan Active? y      Default Route Digit: 0
            Precedence Call Timeout (sec): 30
      Line Load Control Restriction Level: 0
          WNDP Emergency 911 Route String:
                   Preempt Emergency Call?
                   Default Service Domain: 1
       ISDN Precedence Call Timeout (sec): 30
```

ANNOUNCEMENTS

Blocked Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Blocked Precedence Level announcement you want to use.

Busy, Not Equipped

Valid entries	Usage
Valid extension or blank	Enter the extension of the Busy, Not Equipped for Preemption announcement you want to use.

Service Interruption

Valid entries	Usage
Valid extension or blank	Enter the extension of the Service Interruption announcement you want to use.

Unauthorized Precedence Level

Valid entries	Usage
Valid extension or blank	Enter the extension of the Unauthorized Precedence Level announcement you want to use.

Vacant Code

Valid entries	Usage
Valid extension or blank	Enter the extension of the Vacant Code announcement you want to use.

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

CAUTION:
Avaya recommends that you do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits do not match across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

Attendant Diversion Timing (sec)

Valid entries	Usage
10-99 or blank	

Default Route Digit

Appears only when Worldwide Numbering Dial Plan Active is **y**. You must enter a valid digit in this field.

Valid entries	Usage
0	Voice call (the default value)
1	Circuit switched data call
2	Satellite avoidance call
3	(reserved)
4	(reserved)
5	Hotline voice grade call
6	Hotline data grade call
7	(reserved)
8	(reserved)
9	(reserved)

Default Service Domain

Valid entries	Usage
0 to **16777215**	This number defines the system service domain, and must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

Flash

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash precedence level calls.Default is **1**.

Flash Override

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Flash Override precedence level calls. Default is **0**.

Immediate

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Immediate precedence level calls.Default is **2**.

ISDN Precedence Call Timeout (sec)

Valid entries	Usage
4 to **30**	This timeout is used instead of the Precedence Call Timeout when the call is from an MLPP ISDN-PRI trunk. Default is **30** seconds.

Line Load Control Restriction Level

These system levels determine what stations, based on their COR, will be restricted from originating calls.

Valid entries	Usage
0 **2** **3** **4**	Feature not active (no restrictions) (default). Restrict stations with a COR assigned to LLC levels 2, 3, and 4. Restrict stations with a COR assigned to LLC levels 3 and 4. Restrict stations with a COR assigned to LLC level 4.

Precedence Call Timeout (sec)

A busy user receives a precedence call waiting tone only if the incoming call cannot be connected and cannot preempt the user. The called party hears the tone every 10 seconds until answered or the administered time-out occurs. If ignored, the caller is diverted to an attendant or a call-forwarded station.

Valid entries	Usage
4 to **30**	Default is **30**. Enter the number of seconds before a precedence call remains in call waiting status before it is diverted.

Preempt Emergency Call

When this field is set to **y**, an Emergency 911 call made from a preemptable station can be preempted by a higher precedence call

Valid entries	Usage
y/n	Enter **y** to allow preemption of an Emergency 911 call by a higher precedence call.

Priority

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Priority precedence level calls.Default is **3**.

Remote Attendant Route String

Valid entries	Usage
1 to 24 digits or blank	Enter a user-defined telephone to which a precedence call can be routed when no console or night telephone is administered.

Routine

Valid entries	Usage
0 to **9** or blank	Enter the digit assignment for Routine precedence level calls. Default is **4**.

WNDP Emergency 911 Route String

Valid entries	Usage
1 to 24 digits or blank	Valid entries for this field can be a trunk access code (TAC), the AAR or the ARS access code, a WNDP access code, or an extension. If you use a WNDP access code, use the access code for the lowest precedence calling level in the system. **Note:** An Emergency/911 call is a call that routes using the ARS table with the call type defined as either "alrt" or "emer."

Worldwide Numbering Dial Plan Active

Valid entries	Usage
y/n	Enter **y** to enable the Worldwide Numbering Dial Plan. Default is **n**.

Music Sources

This screen appears only when, on the System Parameters Customer-Options (Optional Features) screen, **Tenant Partition?** is **y**. Use this screen to define music sources for Tenant Partitions. Each music source defined on this screen can be used by one or more Tenant Partitions. However, a partition can have only one music source.

Note:

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic Hold system, which does not require such a license, from Avaya Inc. or Avaya's business partners.

Field descriptions for page 1

Figure 491: Music Sources screen

```
change music-source            Music Sources                    Page  1 of  X

         Source No     Type       Source              Description

                 1     music      Type: ext 30002   music-on-extension
                 2     music      Type: group 10    music-on-group
                 3     music      Type: a0904       music-on-part
                 4     tone                         tone-on-hold
                 5     none
                 6     none
                 7     none
                 8     none
                 9     none
                10     none
                11     none
                12     none
                13     none
                14     none
                15     none
```

Description

This field appears only if you entered **music** or **tone** in **Type**.

Note:
When Tenant Partitioning is enabled, **Music/Tone on Hold** on the **Feature-Related System Parameters** screen disappears. However, the value in that field (**tone**, **music**, or **none**) will appear as the first entry on the **Music Sources** screen. If the value was **music**, the port number also appears on the **Music Sources** screen. When Tenant partitioning is disabled, **Music/Tone on Hold** reappears on the **Feature-Related System Parameters** screen, along with the values from the **Music Sources** screen.

Valid entries	Usage
20 alpha-numeric character (max)	Enter a description of the administered music source.

Source

This field appears only if you entered **music** in **Type**. Enter the necessary characters.

Valid entries	Usage
ext	audio source extension for a single or group audio source
group	a Music-on-Hold analog group number
port	an analog or auxiliary trunk source location

Source No

Display only field - the number assigned to this source. The maximum number of music sources is 20 for DEFINITY CSI. This screen appears with the appropriate pages to accommodate the number of music sources your system can support.

Type (column)

If you entered a value in **Music/Tone on Hold** on the **Feature-Related System Parameters** screen, that value will appear in this field.

Valid entries	Usage
music	Enter the type of treatment to be provided by the music source.
tone	Only one music source can use this value.
none	

Type (field)

This field appears only when the entry in the **Type** column is **music**.

Valid entries	Usage
ext **group** **port**	Indicate whether the source is an announcement extension, an audio group, or a port on a VAL board. **Note:** After a valid value is entered, a blank field appears for entry of the appropriate source identifier (extension number, audio group number, or port number).

Network Facilities

The ISDN **Network-Facilities** screen is used to administer new network-provided service or feature names and corresponding ISDN PRI (network specific facilities information element) encodings, for call-by-call trunk groups. Values for pre-defined facilities are displayed at the top of the screen and are display-only. User-defined facilities and services can be entered in the fields below.

When **Usage Allocation Enhancements** on the **System Parameters Customer Options** screen is set to **y**, page 2 of the **Network Facilities** screen appears, allowing for administration of additional user-defined entries.

For more information on usage allocation, see "Call-by-call Service Selection" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Figure 492: Network Facilities screen - page 1

```
change isdn network-facilities                                 Page 1 of x

                           NETWORK-FACILITIES
              Facility                                   Facility
   Name      Type  Coding                  Name      Type  Coding
sub-operator  0    00110                mega800       1    00010
operator      0    00101                megacom       1    00011
outwats-bnd   1    00001                inwats        1    00100
sdn           1    00001                wats-max-bnd  1    00101
accunet       1    00110                lds           1    00111
i800          1    01000                multiquest    1    10000
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
____________ ___   ______              ____________ ___   ______
```

Figure 493: Network Facilities screen - page 2

```
change isdn network-facilities                                Page 2 of x

                              NETWORK-FACILITIES
             Facility                                    Facility
   Name        Type  Coding                      Name        Type  Coding
___________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
____________ ___    ______                ____________ ___    ______
```

Field descriptions for page 1

Name

Valid entries	Usage
printable alphanumeric characters	Enter the name for the feature or service.

Facility Type

Enter the facility type. For types **2** and **3**, **Usage Allocation Enhancements** on the **System Parameters Customer Options** screen must be **y**.

Valid entries	Usage
0 - feature	Enter **0** for predefined features.
1 - service	Enter **1** for predefined services.
2 - incoming	Enter **2** for an incoming-type user-defined entry.
3 - outgoing	Enter **3** for an outgoing-type user-defined entry.

Facility Coding

Valid entries	Usage
characters	Enter the ISDN-specified value for this service or feature.

Node Names

See **IP Node Names**.

Node Number Routing

This screen specifies the routing pattern associated with each node in a public or private network. Node Number Routing is a required capability for Extension Number Portability (ENP) and is associated with the Uniform Dial Plan (UDP).

Field descriptions for page 1

Figure 494: Node Number Routing screen

```
change node-routing n                                            Page 1 of x
                                NODE NUMBER ROUTING
                             Partitioned Group Number: 1

    Route      Route      Route      Route      Route      Route      Route
     Pat        Pat        Pat        Pat        Pat        Pat        Pat

              15 ___     30 ___     45 ___     60 ___     75 ___     90 ___
   1 ___      16 ___     31 ___     46 ___     61 ___     76 ___     91 ___
   2 ___      17 ___     32 ___     47 ___     62 ___     77 ___     92 ___
   3 ___      18 ___     33 ___     48 ___     63 ___     78 ___     93 ___
   4 ___      19 ___     34 ___     49 ___     64 ___     79 ___     94 ___
   5 ___      20 ___     35 ___     50 ___     65 ___     80 ___     95 ___
   6 ___      21 ___     36 ___     51 ___     66 ___     81 ___     96 ___
   7 ___      22 ___     37 ___     52 ___     67 ___     82 ___     97 ___
   8 ___      23 ___     38 ___     53 ___     68 ___     83 ___     98 ___
   9 ___      24 ___     39 ___     54 ___     69 ___     84 ___     99 ___
  10 ___      25 ___     40 ___     55 ___     70 ___     85 ___
  11 ___      26 ___     41 ___     56 ___     71 ___     86 ___
  12 ___      27 ___     42 ___     57 ___     72 ___     87 ___
  13 ___      28 ___     43 ___     58 ___     73 ___     88 ___
  14 ___      29 ___     44 ___     69 ___     74 ___     89 ___
```

Partitioned Group Number

This display-only field displays the partitioned group number associated with the node numbers being administered.

Valid entries	Usage
Display only	The partitioned group number is either specified on the command line or defaults to partitioned group number 1.

Node Number

This display-only field lists the node number to be changed.

Valid entries	Usage
Display only	Two pages display simultaneously for a total of 200 nodes (100 per page). For example, entering change node-routing 87 displays nodes 1 through 199, and entering change node-routing 151 displays nodes 100 through 299. However, entering change node-routing 999 displays nodes 900 through 999 on one page.

Route Pat

Enter the routing pattern associated with the corresponding node number. This field repeats the same number of times as there are node numbers on the page.

Valid entries	Usage
1 to **254**	Enter a number between 1 and 254, or blank.

Numbering — Private Format

This screen supports Private Numbering Plans (PNP). It allows you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Avaya Communication Manager supports private-network numbers up to 15 digits long. If the total number — including the level 1 and 2 prefixes, the PBX identifier, and the extension — is more than 15 digits long, neither QSIG Party Numbers nor the information elements are created or sent.

Field descriptions for page 1

Figure 495: Numbering — Private Format screen

```
change private-numbering 0                                Page 1 of 2

                          NUMBERING - PRIVATE FORMAT

Ext Ext        Trk       Private Total
Len Code       Grp(s)    Prefix  Len                   Total Administered: 8
5   attd                 3920    4                        Maximum Entries: 540
5   70                   30353   10
4   200                  303538  10
5   510                  30353   10
4   2100                 303538  10
5   5000                 30353   10
5   5200                 30353   10
6   6000                 3035    10
```

Ext Code

Allows for groups of extensions to be administered.

Note:

When **0** alone is entered, the **Ext Len** field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
0 to **13** or blank	The **Ext Code** can be up to 13-digits long depending on the **Ext Len** field entry. The entry cannot be greater than the **Ext Len** field entry. For example, in the case of a 4-digit **Ext Len** field entry, an Ext Code of **12** is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	To generate a private calling number for a call from the attendant group.

Ext Len

Specifies the number of digits the extension can have. On page 1, this field displays the extension length entered as a qualifier on the command line (change private-numbering n).

Valid entries	Usage
0 to **13** or blank	Corresponds to the extension lengths allowed by the dial plan.

Maximum Entries

Valid entries	Usage
System maximum	Display only. Indicates the maximum number of private numbering entries that can be administered on the system.

Private Prefix

Valid entries	Usage
0 to **9**, or blank	The number that is added to the beginning of the extension to form a Private Identification Number. The length of the prefix and the extension must at least equal the total length.

Total Administered

Valid entries	Usage
0 to system maximum	Display only. Indicates the number of private numbering entries that are currently administered on the system.

Total Len

Valid entries	Usage
0 to **13**	The total number of digits to send.

Trk Grp(s)

Communication Manager generates the station's identification number if there is an entry in the **Ext Code** field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10** to **24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

Field descriptions for page 2

Page 2 of the **Numbering — Private** screen provides blank fields for new entries. See page 1 for field descriptions.

Numbering — Public/Unknown Format

The screen allows you to specify the desired digits for the Calling Number IE and the Connected Number IE (in addition to the QSIG Party Number) for any extension in the Public and/or Unknown Number Plans.

This screen is used for ARS public trunks as well as SIP Enablement Services (SES) trunks. It supports the ISDN Call Identification Display feature. The feature provides a name/number display for display-equipped stations within an ISDN network. The system uses the caller's name and number and displays it on the called party's display. Likewise, the called party's name and number can be displayed on the caller's display.

In Communication Manager 3.1 and later, the **Public-Unknown Numbering** screens support 9,999 entries. The ANI table, which this screen uses, is increased from 240 to 9,999 entries. This increase is for S8500 and S87XX Servers only. The other servers keep the maximum of 240 entries.

Access the **Numbering — Public/Unknown** screen with the command **`change public-unknown-numbering`** ***`n`***, where ***`n`*** is the length of a value between **0** and **7** appearing in the **Ext Code** column.

Note:

Use the command **`change public-unknown-numbering`** ***`n [ext-digits x][trunk-group trunk-group-number]`*** to administer the desired digits for name and number display on display-equipped stations in an ISDN network. This trunk-group command option displays valid results only when used in conjunction with the ***`ext-digits`*** option. Otherwise, an error message is returned.

The screen consists of two pages: page 1 displays up to 30 **Ext Code** entries matching the requested **Ext Code** length entered on the command line, and page 2 provides 30 blank entries for new user input. If there is sufficient room on the screen, **Ext Code** entries that are longer than the specified length are also displayed. Enter a length of **0** to designate the attendant. If there are more entries of length ***`n`*** than can be displayed, modify your command to use the **`ext-digits`** ***`x`*** command line modifier.

Administer these screens if either the **Send Calling Number**, **Send Connected Number** field is specified, or the **Supplementary Service Protocol** field is **b** on the **Trunk Group** screen.

Note:

If the table is not properly administered and the **Send Calling Number** or **Send Connected Number** field is **y** or **r** and the **Numbering Format** field on the **ISDN Trunk Group** screen is **public** or **unknown**, the Calling Number and Connected Number IE are not sent. If the table is not administered, but the **Send Calling Number** or **Send Connected Number** field is **public** or **unknown**, the Identification Number (PartyNumber data type) is not sent for QSIG PartyNumbers. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) will be sent marked as **PresentationRestricted** with **NULL** for the associated digits.

Following are examples and explanations of the output of common public-unknown-numbering commands.

The command **`list public-unknown-numbering`** operates as follows:

- **`list public-unknown-numbering start 4`**—displays the first entry starting with Ext Len of 4 followed by subsequent entries.
- **`list public-unknown-numbering start 4 count 50`**—displays the first 50 entries starting with Ext Len 4.
- **`list public-unknown-numbering`**—displays all entries.

The command **`change/display public-unknown-numbering`** operates as follows:

- **`change/display public-unknown-numbering 0`**—the screen displays the attendant entry first, followed by the subsequent entries.
- **`change/display public-unknown-numbering 4`**—the screen displays the first Ext Code of length 4 followed by the subsequent entries.
- **`change/display public-unknown-numbering 5 ext-digits 10010`**—the screen displays the first entry of Ext Code 10010 followed by the subsequent entries
- **`change/display public-unknown-numbering 5 ext-digits 10020`**—If 10020 has not been assigned, the screen displays the next entry following 10020 and subsequent entries.
- When used with the `Ext-Len` argument, for example, **`change public 5`**, the display starts with the first record found that matches the entered Extension Length, that is, 5. Then the system displays subsequent records.

Field descriptions for page 1

Figure 496: Numbering Public/Unknown screen

```
change public-unknown-numbering 5                                     Page 1 of X
                       NUMBERING - PUBLIC/UNKNOWN FORMAT

                                                 Total
Ext Extension       Trk       CPN                  CPN
Len Code            Grp(s)    Prefix               Len
 12 1234567890123   123456789 123456789012345 12

 5 4                777777                     10
 5 4                250       30379            10
 5 4                253       30379            10
 5 41               40        303222           11
 5 41               45                         5
 5 41               87        30323            10
 5 43               538                        7
 5 45               222                        7
 5 47               2222                       9
 5 61               45                         5
 5 406              250       30379            10
 5 406              253       30379            10
 5 418                        303538           11
 5 419                        222222222222222 15
 5 770                        970              8
```

CPN Prefix

Use this field to specify the number that is added to the beginning of the extension to form a Calling or Connected Number.

Valid entries	Usage
1 to 15 digits	Only digits are allowed in the CPN Prefix column. Leading spaces, or spaces in between the digits, are not allowed. ● If the length of the CPN Prefix matches the Total CPN Length, the extension number is not used to formulate the CPN number. ● If the number of digits in the CPN Prefix plus the extension length exceeds the administered Total CPN Length, excess leading digits of the extension are deleted when formulating the CPN number. ● If the number of CPN Prefix digits plus the extension length is less than the Total CPN Length, the entry is not allowed. ● If the Total CPN Length is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.
blank	If this field is blank, the extension is sent unchanged. This is useful in countries where the public network is able to insert the appropriate CPN Prefix to form an external DID number.

Ext Code

Allows for groups of extensions to be administered.

Note:

When **0** alone is entered, the **Ext Len** field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
leading extension digits (**0** to **9**)	The **Ext Code** can be up to 13 digits long depending on the **Ext Len** field entry. The entry cannot be greater than the **Ext Len** field entry. For example, in the case of a 4-digit **Ext Len** field entry, an Ext Code of **12** is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	For attendant
blank	No extension code is entered.

Ext Len

Specifies the number of digits the extension can have. On page 1, this field displays the extension length entered as a qualifier on the command line (**`change public-unknown-numbering`** ***`n`***).

Valid entries	Usage
0 to **13** or blank	Corresponds to the extension lengths allowed by the dial plan.

Total CPN Len

Valid entries	Usage
0 to **15**	Enter the total number of digits to send.
Blank	This is the default. Leave blank when deleting an entry.

Trk Grp(s)

Communication Manager generates the station's identification number if there is an entry in the **Ext Code** field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10** to **24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

Field descriptions for page 2

Page 2 of the **Numbering — Public/Unknown** screen provides blank fields for new entries. See page 1 for field descriptions.

Off-PBX Telephone Configuration Set

See **Configuration Set**.

Off-PBX Telephone Feature-Name-Extensions

See **Extensions to Call Which Activate Features by Name**.

Off-PBX Telephone Mobile Feature Extension

Field descriptions for page 1

Figure 497: Off-PBX Telephone Mobile Feature Extension screen

```
change off-pbx-telephone mobile-feature-ext            Page 1 of x
                 OFF-PBX TELEPHONE MOBILE FEATURE EXTENSIONS

             Mobile Call (CTI) Extension:
```

Mobile Call (CTI) Extension

Valid entries	Usage
numeric value of an unassigned extension	A CTI call to this Mobile Feature Extension (MCE) creates an OPTIM call under CTI influence. A call to the MCE triggers an Off-PBX extend-call from a desk phone to its mapped cell phone number and to the destination. All calls made using the MCE appear to the destination as if they were dialed from the desk phone.

Off-PBX Telephone Station-Mapping

See **Stations With Off-PBX Telephone Integration**.

Optional Features

See System Parameters Customer-Options (Optional Features).

Partition Routing Table

Use this table to identify routing for partition groups associated with an ARS analysis entry.

Field descriptions for page 1

Figure 498: Partition Routing Table screen

```
change partition route-table                                     Page 1 of X
                              Partition Routing Table

                                 Routing Patterns
 Route
 Index        PGN 1    PGN 2    PGN 3    PGN 4    PGN 5    PGN 6    PGN 7    PGN 8
 -----        -----    -----    -----    -----    -----    -----    -----    -----
 196          ____     ____     ____     ____     ____     ____     ____     ____
 197          ____     ____     ____     ____     ____     ____     ____     ____
 198          ____     ____     ____     ____     ____     ____     ____     ____
 199          ____     ____     ____     ____     ____     ____     ____     ____
 200          ____     ____     ____     ____     ____     ____     ____     ____
 201          ____     ____     ____     ____     ____     ____     ____     ____
 202          ____     ____     ____     ____     ____     ____     ____     ____
 203          ____     ____     ____     ____     ____     ____     ____     ____
 204          ____     ____     ____     ____     ____     ____     ____     ____
 205          ____     ____     ____     ____     ____     ____     ____     ____
 206          ____     ____     ____     ____     ____     ____     ____     ____
 207          ____     ____     ____     ____     ____     ____     ____     ____
 208          ____     ____     ____     ____     ____     ____     ____     ____
 209          ____     ____     ____     ____     ____     ____     ____     ____
 210          ____     ____     ____     ____     ____     ____     ____     ____
```

PGN 1 (through PGN 8)

Enter the routing for each partition group associated with each route index number.

Valid entries	Usage
1 to **640**	Specifies the route pattern used to route the call
r1 to **r32**	Specifies the remote home numbering plan area table used to route the call
node	Designates node number routing
deny	Blocks the call

Personal CO Line Group

Use this screen to set up a personal central office line trunk group.

Field descriptions for page 1

Figure 499: Personal CO Line Group screen

```
add personal-CO-line                                             Page 1 of x
                            PERSONAL CO LINE GROUP

 Group Number: __                   Group Type:_____________  CDR Reports: _
   Group Name: _________________                                      TAC: ____
Security Code: ____              Coverage Path: ____     Data Restriction? _
                              Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________           Trunk Direction: ________
                 Trunk Port: _______         Disconnect Timing(msec): ____
                 Trunk Name: __________             Trunk Termination: ______
          Outgoing Dial Type: _________             Analog Loss Group: ___
                   Prefix-1? _                     Digital Loss Group: ___
Disconnect Supervision - In? _                        Call Still Held? _
 Answer Supervision Timeout: ___         Receive Answer Supervision? _
                 Trunk Gain: ____                             Country: __
          Charge Conversion: _____             DS1 Echo Cancellation: _
              Decimal Point: ______
            Currency Symbol: ___
                Charge Type: _______
```

The **Coverage Path** and **Security Code** fields are unique to this screen and are described below. For descriptions of other fields on this screen, see Trunk Group on page 1669.

Coverage Path

Valid entries	Usage
1 to **9999**	Enter the number of the call coverage path you want to use for incoming calls.
t1 to **t999**	Enter the number of a time-of-day table.
blank	Assigning a coverage path is optional: leave this field blank if you do not want to assign one.

Security Code

Valid entries	Usage
3 to 8 digits	Enter a code that users must dial to retrieve voice messages and to use the Demand Print Message feature.
blank	Leave this field blank if you do not want to use a security code to control access.

Field descriptions for page 2

Figure 500: Personal CO Line Group screen

```
change personal-CO-line 1                                       Page 2 of 3
                             PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

    Ext             Name
    1234567890123   123456789012345678901234567

 1: 1010            tst 4bri 1b0701
 2:
 3:
 4:
 5:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
16:
```

Ext

This display-only field shows the extension of telephones that have a **CO Line** button.

Name

This display-only field shows the name assigned to telephones that have a **CO Line** button.

Field descriptions for page 3

Administrable timers for Personal CO Line groups appear on field descriptions for page 3. See Administrable Timers for Trunk Group screen for standard field definitions of the available timers.

Related topics

See Adding a PCOL trunk group on page 485 for instructions.

See Trunk Group on page 1669 for definitions of all trunk group fields that are *not* unique to the **PCOL** screen.

Pickup Group

This screen implements call pickup groups with up to 50 extensions per group. A pickup group is a group of users authorized to answer calls to a telephone extension within that group of users. A telephone extension can belong to only one pickup group.

Field descriptions for pages 1 and 2

Figure 501: Pickup Group screen

```
add pickup-group 1                                                 Page 1 of x
                                   PICKUP GROUP

           Group Number: 1
             Group Name: Test Group
GROUP MEMBER ASSIGNMENTS

    Ext              Name
 1:1234567890123   Mohandas Karamchand Gandhi
 2:
 3:
 4:
 5:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
```

Ext

Enter the extension assigned to a station.

Valid entries	Usage
Valid extension number.	A VDN cannot be assigned to a Call Pickup group.

Extended Group Number

This field appears only when the **Group Call Pickup** field is set to **flexible** on the **Feature-Related System Parameters** screen. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Valid entries	Usage
1 to **100** (DEFINITY CSI)	Enter the extended group number or blank.

Group Number

Valid entries	Usage
Pickup Group number	This display-only field appears when the screen is accessed using an administration command such as **add** or **change**.

Name

This display-only field shows the name assigned to the above extension number when the users and their associated extensions were administered.

Precedence Routing Digit Analysis Table

Avaya Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern of an outgoing Multiple Level Precedence and Preemption (MLPP) call.

Field descriptions for page 1

Figure 502: Precedence Routing Digit Analysis Table screen

```
change precedence-routing analysis nn                        Page 1 of x

                      PRECEDENCE ROUTING DIGIT ANALYSIS TABLE
                                                                Percent Full: 22

          Dialed           Total     Route    Preempt
          String          Min  Max  Pattern   Method
    002383                9    9     36       group
    002385                9    9     35       group
    002388                9    9     86       group
    003032383             12   12    36       group
    003032388             12   12    86       group
    003033383             12   12    34       group
    003033388             12   12    84       group
    003034383             12   12    32       group
    003034388             12   12    82       group
    003035383             12   12    30       group
    003035388             12   12    80       group
    003383                9    9     34       group
    003385                9    9     33       group
    003388                9    9     84       group
    004383                9    9     32       group
```

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the call-processing server analyzes.
***, x, X**	wildcard characters

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Percent Full

Valid entries	Usage
0 to **100**	Display only. Shows the percent of the Precedence Routing Digit Analysis Table that is currently in use.

Preempt Method

Enter the preemption method you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
group	The system checks the first trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the system connects the call. This is the default.
route	The system checks each trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the call is connected.

Route Pattern

Enter the route number you want the server running Avaya Communication Manager to use for this dialed string.

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call

Precedence Routing Digit Conversion Table

Use the **Precedence Routing Digit Conversion** screen to assign the Precedence Routing digit conversion. Digit conversion takes digits dialed on incoming calls and converts the digits to local telephone numbers, usually extension numbers.

Field descriptions for page 1

Figure 503: Precedence Routing Digit Conversion Table screen

```
change precedence-routing digit-conversion n            Page   1 of   x

                   PRECEDENCE ROUTING DIGIT CONVERSION TABLE
                                                               Percent Full: 11

  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv

  x2386                8     8     4                          ext   n
  x3032386             11    11    7                          ext   n
  x3033386             11    11    7                          ext   n
  x3034386             11    11    7                          ext   n
  x3035386             11    11    7                          ext   n
  x3386                8     8     4                          ext   n
  x4386                8     8     4                          ext   n
  x5386                8     8     4                          ext   n
  x6                   5     5     1                          ext   n
  xx2386               9     9     5                          ext   n
  xx3032386            12    12    8                          ext   n
  xx3033386            12    12    8                          ext   n
  xx3034386            12    12    8                          ext   n
  xx3035386            12    12    8                          ext   n
```

Conv

Valid entries	Usage
y	Enter **y** to allow more conversions.
n	Enter **n** to prevent further conversions.

Del

Valid entries	Usage
0 to **Min**	Enter the number of leading digits to delete.

Matching Pattern

Valid entries	Usage
0 to **9**, ***, **x**, or **X**	Enter the precedence digit and the address digits. For WNDP dialing, you must also enter the route code.

CAUTION:

The **Matching Pattern** field requires the following format for routing DSN numbers: For precedence dialing (non-WNDP dialing), enter the precedence digit (typically 0-4) and the address digits. For WNDP dialing, enter the precedence digit (typically 0-4), the route code, and the address digits. An **x** in the digit string is a wildcard that matches on any single digit.

Max

Valid entries	Usage
Between **Min** and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Min

Valid entries	Usage
Between **1** and **Max**	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Net

Valid entries	Usage
ext	Extension. Uses ARS tables or AAR tables to route the call.
pre	Precedence routing. Uses the Precedence Analysis Tables to route the call.

Replacement String

Valid entries	Usage
0 to **9**, *****, **#,** or blank, up to 18 characters	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits. The **#** sign, if present in the string, should be the last character in the string. This signifies the end of the modified digit string.

Route Pattern

Valid entries	Usage
1 to **999**	Specifies the route pattern used to route the call.
deny	Blocks the call.

PRI Endpoint

This screen administers PRI Endpoints for the Wideband Switching feature.

Note:

A PRI Endpoint with a width greater than 1 can be administered only if the **Wideband Switching** feature has been enabled on the **System Parameters Customer-Options (Optional Features)** screen.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system. For information on endpoint applications connected to line-side non-ISDN T1 or E1 facilities, see Access Endpoint on page 745 in this module.

A PRI Endpoint is defined as 1 to 31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

Field descriptions for page 1

Figure 504: PRI Endpoint screen

```
add pri-endpoint next                                           Page   1 of  x
                                  PRI ENDPOINT

                     Extension: 300
                          Name: 27 character PRI Endpoint 1
               (Starting) Port:                          Width: 1
  Originating Auto Restoration? n              Signaling Group:
                           COR: 1                          COS: 1
                            TN: 1           Simultaneous Calls? n
             Maintenance Tests? y

                             WIDEBAND SUPPORT OPTIONS

                                H0? n
                               H11? n
                               H12? n
                             NXDS0? y   Contiguous? n
```

COR

Valid entries	Usage
0 to **995**	Enter class of restriction (COR) to determine calling and called party privileges

COS

Valid entries	Usage
0 to **15**	Enter the Class of Service (COS) to determine the features that can be activated by, or on behalf of, the endpoint.

Extension

A display-only field when the screen is accessed using an administration command such as **change** or **display**.

Valid entries	Usage
Extension	This is the extension number used to access the PRI endpoint. Enter a valid unassigned extension number when completing a paper screen.

Maintenance Tests

Valid entries	Usage
y/n	Enter **y** to run hourly maintenance tests on this PRI Endpoint.

Name

Identifies the endpoint.

Valid entries	Usage
Up to 27 alphanumeric characters.	Enter a name for the endpoint.

Originating Auto Restoration

Valid entries	Usage
y/n	Enter **y** to automatically restore calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid entries	Usage
1 to **416**, blank (S8300/S87XX Servers) **1** to **110** or blank (DEFINITY CSI)	Enter the D-channel or D-channel pair that will provide the signaling information for the set of B-channels that make up the PRI Endpoint.

Simultaneous Calls

Valid entries	Usage
y/n	Enter **y** to specify that multiple simultaneous calls can be placed to/from the PRI Endpoint.

(Starting) Port

Enter the seven-character starting port of the PRI Endpoint. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number

TN

Valid entries	Usage
1 to **20** (DEFINITY CSI) **1** to **100** (S8300/S87XX Servers)	Enter the Tenant Partition number.

Width

Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the PRI Endpoint.This field cannot be blank.

Valid entries	Usage
1 to **31**	A width of **6** defines a PRI Endpoint that can support data rates up to 384 Kbps.

WIDEBAND SUPPORT OPTIONS

Contiguous

Specifies whether to hunt contiguous NXDS0 channels. This field only appears if **y** is entered in **NXDS0**. The hunt algorithm to satisfy an NXDS0 call is as follows:

Valid entries	Usage
y/n	Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).
n	Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points.

H0

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

Valid entries	Usage
y/n	Enter **y** to specify the NXDS0 multi-rate service.

Processor Channel Assignment

Use this screen to assign each local processor channel to an interface link channel, and to define the information associated with each processor channel on an Ethernet link.

Note:

You cannot remove a service from this screen if that service has overrides defined on the **Survivable Processor** screen. For more information, see Setting up Processor Ethernet on page 593.

Field descriptions for page 1

Figure 505: Processor Channel Assignment screen

```
change communication-interface processor-channel                  page 1 of x

                                                                       page 2 of x

                      SURVIVABLE PROCESSOR - PROCESSOR CHANNELS

Proc          Gtwy                Interface           Destination       Session     Mach ID
Chan   Enable  To    Appl.    Mode  Link/Chan          Node        Port    Local/Remote
 1      y            mis       s     9    5001        CMS_hogan      0        1       1      1
 2      y            ccr       s     10   5002        ccrhost1       0        2       2      1
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
 16
```

Appl

Use this field to specify the server application type or adjunct connection used on this channel.

Valid entries	Usage
audix	Voice Messaging
ccr	Contact Center Reporting
dcs	Distributed Communication System
echo	
fp-mwi	ISDN Feature Plus Message Waiting Indication. This channel passes message waiting light information for subscribers on the messaging system, from a messaging adjunct on a main switch for a phone on a satellite switch. The terminating location (far end) of this channel must be an Avaya Communication Manager system compatible with ISDN Feature Plus proprietary protocol.
gateway	Supports an X.25 connected AUDIX connected to an ISDN DCS network.
gteway-tcp	Supports a TCP-connected AUDIX connected to an ISDN DCS network.
mis	Management Information System, otherwise known as CMS (Communication Management System)
msaaawl **msaclk** **msahlwc** **msallwc** **msamcs**	All **msa** entries refer to an obsolete product. The system does not accept these entries.
qsig-mwi	QSIG Message Waiting Indication. Used with a QSIG-based interface to a messaging system, this channel passes message waiting light information for subscribers on the messaging system.

Destination Node

Use this field to identify the server or adjunct at the far end of this link.

Valid entries	Usage
valid administered node name	Enter an adjunct name, server name, far end IP address, node name, or leave blank for services local to this Avaya S8XXX Server. For ppp connections, match the **Destination Node Name** on the ppp **Data Module** screen.

Destination Port

Use this field to identify the port number of the destination.

Valid entries	Usage
0, **5000** to **64500**	Enter the number of the destination port. An entry of **0** means any port can be used.

Enable

Use this field to enable or disable this processor channel.

Valid entries	Usage
y/n	Enter **y** to enable this processor channel on the main server. Enter **n** to disable this processor channel.

Gtwy to

This field identifies which processor channel the specified processor channel is serving as a gateway to.

Valid entries	Usage
1 to system max or blank	Enter the number of the processor channel.

Interface Channel

This field identifies the channel number or the TCP/IP listen port channel to carry this processor (virtual) channel. For TCP/IP, interface channel numbers are in the range **5000** to **64500**. The value **5001** is recommended for CMS, and **5003** is recommended for DCS.

Valid entries	Usage
0, **5000** to **64500**	For **ethernet** or **ppp**. The channel number **0** means any port can be used.

Interface Link

This field identifies the physical link carrying this processor (virtual) channel.

Valid entries	Usage
1 to **254**	Enter the physical link carrying this processor (virtual) channel.
p (processor)	Enter **p** to use the Communication Manager's Processor Ethernet interface for adjunct connectivity.
blank	

Mach ID

Valid entries	Usage
1 to **63** for MWI, **1** to **63** for DCS, **1** to **99** for AUDIX, or blank	Enter the destination server ID defined on the dial plan of the destination server.

Mode

Valid entries	Usage
c(lient) **s(erver)** blank	Indicate whether the IP session is passive (client) or active (server). This field must be blank if the interface link is **procr-intf**. This field cannot be blank if the type of interface link is **ethernet** or **ppp**.

Proc Chan

This display-only field shows the number assigned to each processor channel you administer. Range is from **1** to **384**.

Session - Local/Remote

Local and Remote Session numbers must be consistent between endpoints.

Valid entries	Usage
1 to **256** (si) **1** to **384** (r) or blank	For each connection, the Local Session number on this Avaya S8XXX Server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections.

QSIG to DCS TSC Gateway

The **QSIG to DCS TSC Gateway** screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

This screen only appears if the **Interworking with DCS** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Field descriptions for page 1

Figure 506: QSIG to DCS TSC Gateway screen

```
change isdn qsig-dcs-tsc-gateway                              Page 1 of x

                               QSIG TO DCS TSC GATEWAY

    Subscriber         Sig   TSC          Subscriber          Sig       TSC
     Number            GRP  Index          Number             GRP      Index
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
 __________________   ___   ___       __________________      ___        ___
```

Sig Grp

Valid entries	Usage
1 to **110**	Enter the assigned signaling group number for DEFINITY CSI.
1 to **650**	Enter the assigned signaling group number for S8300/S87XX Servers.

Subscriber Number

You can enter up to 28 subscriber numbers.

Valid entries	Usage
0 to **9**, *, '**x**', '**X**'	Enter a subscriber number up to 20 characters in length. You can use wildcards '**x**' and '**X**' to enter subscriber numbers.

TSC Index

You must complete the **TSC Index** field for each machine ID.

Valid entries	Usage
1 to **64**	Enter the assigned signaling group number for **qsig-mwi** application type on the **Signaling Group** screen.

Valid entries	Usage
0 to **9**, *, #	Enter up to 4-digit access code.

Reason Code Names

Use the **Reason Code Names** screen to assign names to reason codes. You can assign a different name to each reason code for Aux Work and for Logout. Pages 2 and 3 appear when, on the **Feature-Related System Parameters** screen, the **Two-Digit Aux Work Reason Codes** field is **y**. These additional pages accommodate names for Aux Work Reason Codes 10 to 99. Note that **Logout** reason codes can only be in the range of 0 to 9, even if the Two-Digit Aux Work Reason Codes option is active.

Figure 507: Reason Code Names screen - page 1

```
change reason-code-names                                         Page   1 of   x

                                  REASON CODE NAMES

                            Aux Work            Logout

      Reason Code 1:
      Reason Code 2:
      Reason Code 3:
      Reason Code 4:
      Reason Code 5:
      Reason Code 6:
      Reason Code 7:
      Reason Code 8:
      Reason Code 9:

Default Reason Code:
```

Figure 508: Reason Code Names screen - page 2

```
change reason-code-names                                    Page   2 of   x

                        REASON CODE NAMES - AUX WORK

   10:                       28:                       46:
   11:                       29:                       47:
   12:                       30:                       48:
   13:                       31:                       49:
   14:                       32:                       50:
   15:                       33:                       51:
   16:                       34:                       52:
   17:                       35:                       53:
   18:                       36:                       54:
   19:                       37:                       55:
   20:                       38:                       56:
   21:                       39:                       57:
   22:                       40:                       58:
   23:                       41:                       59:
   24:                       42:                       60:
   25:                       43:                       61:
   26:                       44:                       62:
   27:                       45:                       63:
```

Figure 509: Reason Code Names screen - page 3

```
change reason-code-names                                    Page   3 of   x

                        REASON CODE NAMES - AUX WORK

   64:                       82:
   65:                       83:
   66:                       84:
   67:                       85:
   68:                       86:
   69:                       87:
   70:                       88:
   71:                       89:
   72:                       90:
   73:                       91:
   74:                       92:
   75:                       93:
   76:                       94:
   77:                       95:
   78:                       96:
   79:                       97:
   80:                       98:
   81:                       99:
```

Field descriptions for page 1

Aux Work

Valid entries	Usage
up to 16 alphanumeric characters	Enter the name to be associated with a reason code when the agent uses the reason code to enter Aux Work mode. Default is blank.

Default Reason Code

Use this field to enter a name for the default reason codes. You can enter a separate name for the Aux Work Reason Code of 0 and for the Logout Reason Code of 0. If an agent changes to Aux Work mode and the Aux Work Reason Code Type is set to none, the agent is put into Aux Work mode with the default Aux Work reason code, even if you have administered a different reason code for the Aux button. If an agent logs out when the Logout Reason Code Type is set to none, the agent is logged out with the default Logout reason code.

Valid entries	Usage
up to 16 alphanumeric characters	Enter a name for the default reason code. Default is blank.

Logout

Valid entries	Usage
up to 16 alphanumeric characters	Enter the name to be associated with a reason code when the agent uses the reason code to log out. Default is blank. Note that Logout reason codes can only be in the range of 0 to 9, even if the Two-Digit Aux Work Reason Codes option is active.

Remote Access

The **Remote Access** screen is used to implement the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

To ensure the security of your system, consider the following:

- Make all remote access facilities unlisted directory telephone numbers.
- Require users to enter a Barrier Code of at least seven random digits AND an Authorization Code of at least 13 random digits to make network calls.
- Make Authorization Codes nonconsecutive (random) and change them, at least, quarterly.
- Deactivate Authorization Codes immediately if the user leaves the company or changes assignments.
- Assign the minimum level of calling permissions required to each Authorization Code.
- Block off-hours and weekend remote access calling, when possible. Use Alternative Facility Restriction Levels, if available.
- Use a voice recording, warble tone, or no tone and avoid use of a dial tone as a prompt when the remote access unit answers.
- Assign the lowest possible FRL to only allow calls internal to Avaya Communication Manager.

As an additional step to ensure system security, you can permanently disable the Remote Access feature if you do not intend to use it now or in the future. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

CAUTION:

Your attempt to disable the Remote Access feature will be lost if the server running Communication Manager is rebooted without saving translations. Therefore, execute a **`save translation`** command after permanently disabling the Remote Access feature.

Field descriptions for page 1

Figure 510: Remote Access screen

```
change remote-access

                                 REMOTE ACCESS
   Remote Access Extension: ________              Barrier Code Length:____
Authorization Code Required? y           Remote Access Dial Tone: n

     Barrier      COR  TN  COS   Expiration    No. of     Calls
      Code                          Date       Calls      Used
   1:________     1__  1_  1__   __/__/__      ______     ______
   2:________     1__  1_  1__   __/__/__      ______     ______
   3:________     1__  1_  1__   __/__/__      ______     ______
   4:________     1__  1_  1__   __/__/__      ______     ______
   5:________     1__  1_  1__   __/__/__      ______     ______
   6:________     1__  1_  1__   __/__/__      ______     ______
   7:________     1__  1_  1__   __/__/__      ______     ______
   8:________     1__  1_  1__   __/__/__      ______     ______
   9:________     1__  1_  1__   __/__/__      ______     ______
  10:_______      1__  1_  1__   __/__/__      ______     ______

Permanently Disable? __  Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

Authorization Code Required

When you use an authorization code in conjunction with a barrier codes it increases the security of the Remote Access feature.

Valid entries	Usage
y/n	Enter **y** to require an authorization code be dialed by Remote Access users to access the system's Remote Access facilities.

Barrier Code

You must assign a barrier code that conforms to the number entered in the **Barrier Code Length** field. You can enter up to 10 barrier codes per system. Duplicate entries are not allowed. You must keep your own records regarding the distribution of these barrier codes to your personnel.

Note:

After you make an entry in the **Barrier Code** field, additional fields in the same row (**COR**, **TN**, **COS**, **Expiration Date**, and **No. of Calls**) become editable.

Valid entries	Usage
0 to **9** or blank	Enter a 4- to 7-digit number in any combination of digits.
none	Must be specified in the first **Barrier Code** field, if the **Barrier Code Length** field is blank.

Barrier Code Length

Assign a barrier code length of **7** to provide maximum security.

Valid entries	Usage
4 to **7** or blank	Enter a number to indicate the length of the barrier code.

Calls Used

This display-only field shows the number of calls placed using the corresponding barrier code. This field is incremented each time a barrier code is successfully used to access the Remote Access feature. A usage that exceeds the expected rate indicates improper use.

COR

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign the most restrictive class of restriction (COR), that provides only the level of service required, to provide the maximum security.

Valid entries	Usage
0 to **995**	Enter the COR number associated with the barrier code that defines the call restriction features.

COS

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign the most restrictive class of service (COS), that provides only the level of service required to provide the maximum security.

Valid entries	Usage
0 to **15**	Enter the COS number, associated with the barrier code, that defines access permissions for Call Processing features.

Disable Following a Security Violation

This field appears on the screen when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to disable the remote access feature following detection of a remote access security violation. The system administrator can re-enable Remote Access using the `enable remote-access` command.

Expiration Date

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. Assign an expiration date based on the expected length of time the barrier code will be needed. If it is expected the barrier code is to be used for a 2-week period, assign a date two weeks from the current date. If the Expiration Date is assigned, a warning message is displayed on the **System Copyright** screen seven days prior to the expiration date. The system administer can modify the expiration date to extend the time interval if needed.

Valid entries	Usage
A date greater than the current date or blank	Enter the date you want the barrier code to expire.

No. of Calls

This field changes from display-only to editable after you make an entry in the **Barrier Code** field. The **Expiration Date** and **No. of Calls** fields can be used independently or in conjunction to provide the maximum security. If both the **Expiration Date** and **No. of Calls** fields are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Valid entries	Usage
1 to **9999** or blank	Enter the number of Remote Access calls that can be placed using the associated barrier code.

Permanently Disable

Reactivation of remote access to the interface requires Avaya Services intervention.

Valid entries	Usage
y/n	Enter **y** to permanently block remote access to the administration interface.

Remote Access Dial Tone

Set this field to **n** for maximum security. This field appears when the **Authorization Code Required** field is set to **y**.

Valid entries	Usage
y/n	Enter **n** so that there is no Remote Access Dial Tone prompt.

Remote Access Extension

The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered on the trunk group's incoming destination field.

Valid entries	Usage
Extension number	Enter the extension number for Remote Access associated with each trunk that supports the Remote Access feature. You cannot assign a Vector Directory Number (VDN) extension as the remote access extension. Can be blank if no barrier codes.

TN

This field changes from display-only to editable after you make an entry in the **Barrier Code** field.

Valid entries	Usage
1 to **20** (DEFINITY CSI) **1** to **100** (S8300/S87XX Servers))	Enter the Tenant Partition number.

Related Topics

See Setting up Remote Access on page 442 for step-by-step instructions for configuring remote access.

See "Remote Access" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Remote Call Coverage Table

The Remote Call Coverage Table allows you to provide automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers can be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits or blank, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code. Up to 999 remote call coverage points can be entered on the multiple pages of this screen.

Field descriptions for page 1

Figure 511: Remote Call Coverage Table screen

```
change coverage remote

                       REMOTE CALL COVERAGE TABLE
                       ENTRIES FROM 1  TO 1000

01: ________________     16: ________________     31: ________________
02: ________________     17: ________________     32: ________________
03: ________________     18: ________________     33: ________________
04: ________________     19: ________________     34: ________________
05: ________________     20: ________________     35: ________________
06: ________________     21: ________________     36: ________________
07: ________________     22: ________________     37: ________________
08: ________________     23: ________________     38: ________________
09: ________________     24: ________________     39: ________________
10: ________________     25: ________________     40: ________________
11: ________________     26: ________________     41: ________________
12: ________________     27: ________________     42: ________________
13: ________________     28: ________________     43: ________________
14: ________________     29: ________________     44: ________________
15: ________________     30: ________________     45: ________________
```

01 to 1000

Valid entries	Usage
0 to **9** * (DTMF digit asterisk) **#** (DTMF digit pound) **L** (use coverage point only when in LSP or ESS mode) **D** (represents the called extension digits) **’** (pause for 1.5 seconds) **’** (pause for 1.5 seconds) **%** (rest of digits are for end-to end signaling) blank (**L**, **D**, ’,’, and **%** use 2 places)	Enter the destination coverage point up to 16 digits.

Remote Office

This screen supports an arrangement whereby a user can set up a remote office without having an on-premises physical desk-set. An R300 is issued to connect remote DCP and analog telephones, IP telephones, and H.323 trunks to the Communication Manager server via IP.

Field descriptions for page 1

Figure 512: Remote Office screen

```
change remote office x                                          Page 1 of x

                                    REMOTE OFFICE

     Node Name: _______________
Network Region: ___
      Location: __
     Site Data: ______________________________
                ______________________________
                ______________________________
```

Location

Valid entries	Usage
1 to **64**	Specify the location (comprised of the associated time zone and the appropriate numbering plan).

Network Region

Valid entries	Usage
1 to **250** or blank	Specify the network region to be assigned to all stations supported on this remote office.

Node Name

Valid entries	Usage
character string	Specify the node name of the remote office.

Site Data

Valid entries	Usage
30 characters or blank	Any desired information.

RHNPA Table

The **RHNPA Table** defines route patterns for specific 3-digit codes, usually direct distance dialing (DDD) prefix numbers. The appearance of the screen is different slightly depending on the type of Avaya S8XXX Server.

Figure 513: RHNPA Table screen

```
change rhnpa                                                   Page 1 of X
                                RHNPA TABLE: __
                                CODES: 000-999

                             Pattern Choices
          1: ___    3: ___    5: ___    7: ___    9: ___    11: ___
          2: ___    4: ___    6: ___    8: ___   10: ___    12: ___

       Code - Pattern Choice Assignments (from 1-12 above)
00: 1__ 10: 1__ 20: 1__ 30: 1__ 40: 1__ 50: 1__ 60: 1__ 70: 1__ 80: 1__ 90: 1__
01: 1__ 11: 1__ 21: 1__ 31: 1__ 41: 1__ 51: 1__ 61: 1__ 71: 1__ 81: 1__ 91: 1__
02: 1__ 12: 1__ 22: 1__ 32: 1__ 42: 1__ 52: 1__ 62: 1__ 72: 1__ 82: 1__ 92: 1__
03: 1__ 13: 1__ 23: 1__ 33: 1__ 43: 1__ 53: 1__ 63: 1__ 73: 1__ 83: 1__ 93: 1__
04: 1__ 14: 1__ 24: 1__ 34: 1__ 44: 1__ 54: 1__ 64: 1__ 74: 1__ 84: 1__ 94: 1__
05: 1__ 15: 1__ 25: 1__ 35: 1__ 45: 1__ 55: 1__ 65: 1__ 75: 1__ 85: 1__ 95: 1__
06: 1__ 16: 1__ 26: 1__ 36: 1__ 46: 1__ 56: 1__ 66: 1__ 76: 1__ 86: 1__ 96: 1__
07: 1__ 17: 1__ 27: 1__ 37: 1__ 47: 1__ 57: 1__ 67: 1__ 77: 1__ 87: 1__ 97: 1__
08: 1__ 18: 1__ 28: 1__ 38: 1__ 48: 1__ 58: 1__ 68: 1__ 78: 1__ 88: 1__ 98: 1__
09: 1__ 19: 1__ 29: 1__ 39: 1__ 49: 1__ 59: 1__ 69: 1__ 79: 1__ 89: 1__ 99: 1__
```

Field descriptions for page 1

CODES

Display-only field showing the desired 100-block, for example 000 through 099 or 900 through 999 based upon the **`change rhnpa`** command. A separate screen displays for each 100-block.

Code-Pattern Choice Assignments

Valid entries	Usage
1 to **24**	For S87XX Series IP-PNC.

Pattern Choices

There are 12 pattern choices for DEFINITY CSI; there are 24 pattern choices for the S8300/S87XX Servers. Enter the route pattern number you want associated with each code. The pattern choice you list on one screen automatically defaults to the other screens of the same table. If you use one pattern for most of the codes, assign that pattern to choice 1.

Valid entries	Usage
1 to **999** or blank	For S8300/S87XX Servers
1 to **254**	For DEFINITY CSI.

RHNPA TABLE

Display-only field indicating the table number.

Route Pattern

The **Route Pattern** screen defines the route patterns used by your server running Communication Manager. Each route pattern contains a list of trunk groups that can be used to route the call. The maximum number of route patterns and trunk groups allowed depends on the configuration and memory available in your system.

Use this screen to insert or delete digits so AAR or ARS calls route over different trunk groups. You can convert an AAR number into an international number, and insert an area code in an AAR number to convert an on-network number to a public network number. Also, when a call directly accesses a local central office (CO), if the long-distance carrier provided by your CO is not available, Avaya Communication Manager can insert the dial access code for an alternative carrier into the digit string.

Field descriptions for page 1

Figure 514: Route Pattern screen

```
change route-pattern 1                                            Page 1 of X
                       Pattern Number: 1_    Pattern Name:
                                SCCAN? n      Secure SIP? n

                            No.                                     DCS/
  Grp.  FRL NPA Pfx Hop Toll Del   Inserted                         QSIG   IXC
   No.          Mrk Lmt List Dgts  Digits                           Intw
1: ___   _  ___  _  __   __   __   ________________________________   n   user
2: ___   _  ___  _  __   __   __   ________________________________   n   user
3: ___   _  ___  _  __   __   __   ________________________________   n   user
4: ___   _  ___  _  __   __   __   ________________________________   n   user
5: ___   _  ___  _  __   __   __   ________________________________   n   user
6: ___   _  ___  _  __   __   __   ________________________________   n   user

     BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature BAND  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts   Format
                                                      Subaddress
 1: y y y y y n  y  none   ___ both ept outwats-bnd____    _    ________  none
 2: y y y y y n  y             rest     _______________    _    ________  next
 3: y y y y y n  y             rest     _______________    _    ________  rehu
 4: y y y y y n  y             rest     _______________    _    ________  none
 5: y y y y y n  y             rest     _______________    _    ________  none
 6: y y y y y n  y             rest     _______________    _    ________  none
```

Band

Enter a number that represents the OUTWATS band number (US only).

WATS is a voice-grade service that provides both voice and low-speed data transmission calls to defined areas (bands) for a flat rate charge.

This field appears when the **Services/Features** field is **outwats-bnd** and when **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. **Band** is required by Call-by-Call Service Selection.

BCC Value

Bearer Capability Class (BCC) identifies the type of call appropriate for this trunk group, such as voice calls and different types of data calls. This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** in appropriate BCC column (0, 1, 2, 3, 4, or W) if the BCC is valid for the associated route pattern. A trunk group preference can have more than one BCC.

The following table explains BCC values

:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
M	Multimedia call
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

BCIE (Bearer Capability Information Element)

This field applies to ISDN trunks and appears if ITC is **both**.

Valid entries	Usage
ept (endpoint) **unr** (unrestricted)	Use BCIE to determine how to create the ITC codepoint in the setup message.

CA-TSC Request

Use CA-TSC on ISDN B-channel connections.

Valid entries	Usage
as-needed	The CA-TSC is set up only when needed. This causes a slight delay. Avaya recommends this entry for most situations.
at-setup	The CA-TSC is automatically set up for every B-channel call whether or not it is needed.
none	No CA-TSC is set up. Permits tandeming of NCA-TSC setup requests0.

DCS/QSIG Intw

This field only appears if the **Interworking with DCS** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable DCS/QSIG Voice Mail Interworking.

FRL

Valid entries	Usage
0 to **7**	Enter the Facility Restriction Level (FRL) associated with the entries on this row (preference). **0** is the least restrictive and **7** is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.

SECURITY ALERT:
For system security reasons, Avaya recommends using the most restrictive FRL possible.

Grp No

Valid entries	Usage
1 to **666**	Enter the trunk group number associated with this row (preference). For DEFINITY CSI.
1 to **2000**	For S8300/S87XX Servers.

Hop Lmt

Enter the number of hops for each preference. A hop is when a call tandems through a server to another destination. Limiting the number of hops prevents circular hunting, which ties up trunk facilities without ever completing the call. Avaya Communication Manager blocks a hop equal to or greater than the number you enter

.

Valid entries	Usage
blank	Indicates that there is no limit to the number of hops for this preference.
1 to **9**	To limit the number of hops if using the tandem hop feature.
1 to **32**	If using the transit feature.

Inserted Digits

Enter the digits you want inserted for routing. Communication Manager can send up to 52 digits. This includes up to 36 digits you can enter here plus up to 18-digits originally dialed. Special symbols count as two digits each.

Valid entries	Usage
0 to 36 digits (**0** to **9**)	Enter the digits you want inserted for routing.
*	When * is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the *.
#	When # is in the route pattern and the outgoing trunk is signaling type "mf", the MFC tone for the "end-of-digits" is sent out to the CO in place of the #.
	1 of 2

Valid entries	Usage
‘,’	Use 2 places. Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone based on **Out Pulse Without Tone** y/n on the **Feature-Related System Parameters** screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI (used for Russian pulse trunks)
p	The associated trunk group must be of type **sip**. Enter the single digit **p** for fully qualified E.164 numbers. The p is translated to a + and is prepended to the digit string.
	2 of 2

ITC (Information Transfer Capability)

Use Information Transfer Capability (**ITC**) to identify the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be **unre** or **both** if the **BCC** is **y** and the **BCC** value is **W**

Valid entries	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ricted)	Calls from restricted endpoints can access the route pattern.
unre(stricted)	Calls from unrestricted endpoints can access the route pattern.

IXC

Inter-Exchange Carrier (**IXC**) identifies the carrier, such as AT&T, used for calls that route via an IXC, and for Call Detail Recording (CDR).

This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
Valid carrier code	Identifies the carrier for IXC calls
user	For presubscribed carrier. Used when an IXC is not specified.
none	This field must be **none** for non-ISDN trunk groups and for Bellcore NI-2 Operator Service Access. If you need to send an IXC code for a non-ISDN trunk group, enter the IXC code in the Inserted Digits field.

LAR

Enter the routing-preference for Look Ahead Routing. Following are the causes that trigger LAR:

* #3 - No Route to Destination..CV_NRTD
* #6 - channel unacceptable...CV_CU
* #34 - No Circuit or Channel Available.............................CV_NCOCA
* #38 - Network Failure..CV_NETFAIL
* #41 - Temporary Failure...CV_TFAIL
* #42 - Switching Equipment Congestion...........................CV_SEC
* #43 - User Information Discarded...................................CV_UID
* #44 - Requested Circuit/Channel Not Available...............CV_RCCNA
* #47 - Resources Unavailable, Unspecified......................CV_RUU
* #58 - bearer capability not presently available................CV_BCNPA
* #65 - bearer capability not implemented..........................CV_BCNI
* #79 - service/option not implemented, inspect.................CV_SOONIU
* #82 - identified channel does not exist.......................... CV_ICDNE

* #102 - recover on timer expiry......................................CV_ROTE

Valid entries	Usage
next	Go to the next routing preference and attempt the call again.
rehu	Rehunt within the current routing-preference for another trunk to attempt the call again.
none	Look Ahead Routing is not enabled for the preference.

No. Del. Digits

Use this field to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in servers/switches with different dial plans.

Valid entries	Usage
0 to **28** or blank	Enter the total number of digits you want the system to delete before it sends the number out on the trunk. Use for calls that route: ● to or through a remote server ● over tie trunks to a private network server ● over Central Office (CO) trunks to the serving CO

No. Dgts Subaddress

Allows a caller to reach a number where the Avaya S8XXX Server's digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant. Administrable when, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN Feature Plus** field is **y**.

Valid entries	Usage
1 to **5** or blank	Enter the number of dialed digits to send in the calling party subaddress IE.

NPA

This entry is not required for AAR.

Valid entries	Usage
3-digit number	Enter the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group. Call your local telephone company to verify this number if you need help. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.
blank	For AAR calls and for tie trunks

Numbering Format

This field applies only to ISDN trunk groups. Enter a value from table below. This field specifies the format of the routing number used for the trunk group for this preference.

This field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Numbering Plan Identifier	Type of Numbering
blank	E.164(1)	1-MAX
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)
levl0-pvt (enter to allow Network Call Redirection/ Transfer		
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

Note:
To access Bellcore NI-2 Operator Service Access, the **Inserted Digits** field must be **unk-unk**.

Pattern Name

Enter an alphanumeric name for identification purposes.

Pattern Number

This display-only field shows the route pattern number (**1** to **640**).

Prefix Mark

This entry is not required for AAR. For ARS, enter a number from **0** to **4** or blank.

Prefix Marks set the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

For a WATS trunk, the Prefix Mark is the same as the local CO trunk.

Valid entries	Usage
0	• Suppress a user-dialed prefix digit 1 for 10-digit FNPA calls. • Leave a user-dialed prefix digit 1 for 7-digit HNPA calls. • Leave a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Do not use **Prefix Mark 0** in those areas where all long-distance calls must be dialed as 1+10 digits. Check with your local network provider.
1	• Send a 1 on 10-digit calls, but not on 7-digit calls. Use Prefix Mark 1 for HNPA calls that require a 1 to indicate long-distance calls.
2	• Send a 1 on all 10-digit and 7-digit long-distance calls. Prefix Mark 2 refers to a Toll Table to define long distance codes.
3	• Send a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long distance calls are 10-digit calls. The NPA is inserted when a user dials a Prefix digit 1 plus 7-digits. Prefix Mark 3 refers to a Toll Table to define long distance codes.

Valid entries	Usage
4	• Always suppress a user-dialed Prefix digit 1. Use Prefix Mark 4, for example, when ISDN calls route to a server that rejects calls with a prefix digit 1.
blank	For tie trunks, leave this field blank.

SCCAN

This field appears when **Enhanced EC500** on the **System Parameters - Customer Options** screen is set to **y**.

Note:
When the **SCCAN** field is set to **y**, non-SCCAN-associated fields are hidden. Only the **Pattern Number**, **Pattern Name**, **SCCAN**, **Secure SIP**, and **Grp No** fields appear.

Valid entries	Usage
y/n	Enter **y** to indicate that this route pattern supports incoming SCCAN calls.

Secure SIP

Valid entries	Usage
y/n	Specify whether the SIP or SIPS prefix will be used, if the call is routed to a SIP Enablement Services (SES) trunk preference. If SES trunks are not specified on the **Route Pattern** screen, the call will be routed over whatever trunk is specified. Therefore, to ensure a SES TLS connection when such a route pattern is invoked, only SES trunks should be specified. The only instance for entering **y** in this field is when the source provider *requires* a secure SIP protocol. Default is **n**.

Service/Feature

This field appears when **ISDN-PRI** or **ISDN-BRI Trunks** is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Enter up to 15 characters to identify the Service/Feature carried by the information element (IE) in a call in this route pattern. This field is required by Call-by-Call Service Selection, and Network Call Redirection and Transfer.

Note:

User-defined service types, defined on the **Network Facilities** screen, can also be used. In addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature), **1** (service), or **3** (outgoing) on the **Network Facilities** screen is allowed. See the description of the **Network Facilities** screen for more information on usage allocation.

Valid entries		
accunet	**multiquest**	**sdn** (Enter to allow Network Call Redirection/Transfer)
i800	**operator**	**sub-operator**
inwats	**oper-lds** (operator and lds)	**sub-op-lds** (sub-operator and lds)
lds	**oper-meg** (operator and megacon)	**sub-op-meg** (sub-operator and megacom)
mega800	**oper-sdn** (operator and sdn)	**sub-op-sdn** (sub-operator and sdn)
megacom	**outwats-bnd**	**wats-max-bnd**

Toll List

This entry is not required for AAR.

Valid entries	Usage
1 to **32** or blank	For ARS, enter the number of the ARS Toll Table associated with the terminating NPA of the trunk group. You must complete this field if **Prefix Mark** is **2** or **3**.

TSC

Set **TSC** to **y** for feature transparency on DCS+ calls and to use QSIG Call Completion.

Valid entries	Usage
y/n	Enter **y** to allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference.

Security-Related System Parameters

Use this screen to determine when Avaya Communication Manager reports a security violation. Many of the fields on this screen repeat for each type of security violation. We have explained them once here, but the usage is the same for all. See "Security Violations Notifications" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Field descriptions for page 1

Figure 515: Security-Related System Parameters screen

```
change system-parameters security                            Page 1 of x
                     SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Barrier Code Threshold: 10         Time Interval: 0:03
       Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: _____
         Referral Destination: _____
 Authorization Code Threshold: __        Time Interval: _____
       Announcement Extension: _____
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Login (Violation Notification, Remote Access, Authorization Code) Enabled

Valid entries	Usage
y/n	Set to **y** if you want Communication Manager to notify you when a login violation occurs. If this field is **y**, the next 5 fields appear so you can establish the parameters for what is considered a security violation.

Announcement Extension

If you enter a value in this field, the server running Communication Manager *calls the referral destination, then plays this announcement upon answer.*

Valid entries	Usage
Valid extension	The announcement extension where SVN violation announcement resides.

Originating Extension

The originating extension initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination.

Valid entries	Usage
An unassigned extension	If you establish notification for more than one type of security violations, you must assign a different extension to each one. When Communication Manager generates a referral call, *this extension and the type of violation appear on the display at the referral destination.*

Referral Destination

The referral destination receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless the you assign an Announcement Extension.

Valid entries	Usage
An extension	Enter the extension of the telephone, attendant console, or vector directory number (VDN) that you want to receive the referral call for each type of violation. This can be the same extension for all type of violations. If you use a VDN, you must complete the Announcement Extension field. You can also use Call Vectoring Time-of-Day routing to route the referral call to different destinations based on the time of day or the day of the week.

Time Interval

Use this field to enter a time interval for the violation notification.

Valid entries	Usage
0:01 to **7:59**	The range for the time interval is one minute to eight hours. Entered in the screen **x:xx**. For example, if you want the time interval to be one minute, you enter 0:01. If you want the time interval to be seven and one-half hours, you enter 7:30.

SVN Remote Access Violation Notification Enabled

Use the **SVN Remote Access Violation Notification Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on.

SVN Authorization Code Violation Notification Enabled

Use the **SVN Remote Access Violation Notification Enabled** and the **SVN Authorization Code Violation Notification Enabled** fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned these fields on.

Field descriptions for page 2

Figure 516: Security-Related System Parameters screen (for DEFINITY CSI)

```
change system-parameters security                            Page 2 of x
                     SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
            Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10                Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                       Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y
                      Receive Unencrypted from IP Endpoints? n

  REMOTE MANAGED SERVICES
                                           RMS Feature Enabled? y
                            Port Board Security Notification? y
                   Port Board Security Notification Interval? 60

  ACCESS SECURITY GATEWAY PARAMETERS

        MGR1? n    INADS? n
         EPN? n      NET? n
```

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Station Security Code Violation Notification Enabled

Station Security codes are used to validate logins to a particular extension (for example, a home agent using an extender, or two part-time workers using the same telephone, but different extensions, through personal station access.) Enter **y** *here to establish parameters for this.*

STATION SECURITY CODE VERIFICATION PARAMETERS

Minimum Station Security Code Length

This determines the minimum required length of the Station Security Codes that you enter on the **Station** screen.

Valid entries	Usage
3 to **8**	Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

Receive Unencrypted from IP Endpoints

Valid entries	Usage
y/n	Enter **y** to allow unencrypted data from IP endpoints. Default is **n**.

Security Code for Terminal Self Administration Required

Specifies if a Personal Station Access code is required to enter the Self-Administration mode.

Valid entries	Usage
y/n	Enter **y** to indicate that a security code is required.

REMOTE MANAGED SERVICES

RMS Feature Enabled

Use this field to enable Remote Managed Services. When you set this field to **y**, the **Port Board Security Notification** and **Port Board Security Notification Interval** fields appear.

Valid entries	Usage
y/n	Enter **y** to enable the Remote Managed Services feature. Default is **n**.

Port Board Security Notification

This field appears when **RMS Feature Enabled** is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable port board denial of service notification. Default is **n**. When you enter **y** in this field, the **Port Board Security Notification Interval** field appears.

Port Board Security Notification Interval

This field appears when the **RMS Feature Enabled** and **Port Board Security Notification** fields are set to **y**.

Valid entries	Usage
60 to **3600** in increments of 10	Enter the desired interval, in seconds, between port board Denial of Service notifications (traps). Default is **60**. **NOTE:** There is no delay before the first trap is sent. The interval administered in this field applies only to the period *between* the sending of the traps.

ACCESS SECURITY GATEWAY PARAMETERS

These fields appear only if the **Access Security Gateway (ASG)** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

EPN

A direct connection to the Expansion Port Network.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

INADS

A direct cable connection to the Initialization and Administration System used to remotely initialize and administer Avaya Communication Manager.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

MGR1

The direct connect system administration and maintenance access interface located on the processor circuit pack. For more information on the circuit pack, see the *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

NET

A dialed-in (or out) connection to the Network Controller circuit pack. For more information on the circuit pack, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

Translation-ID Number Mismatch Interval (days)

A display-only field for all logins except *init*; only *init* logins can change this field. This field specifies the interval (in days) that the system allows a mismatch between the translation ID stored in the Processor circuit pack(s) and on the flash card. Following expiration of this interval, the ability to execute system administration commands that modify translation data is denied for all logins, except for *init*.

Valid entries	Usage
1 to **90**	Enter a number to indicate the number of days the system allows access to system administration commands.

Service Hours Table

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, ATM, and SIP Enablement Services (SES) trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the **Group Type** field. Field descriptions are alphabetized for easier reference.

Field descriptions for page 1

Figure 517: Service Hours Table screen

```
change service-hours-table 4                                    Page   1 of   1
                               SERVICE HOURS TABLE:

                                    Number: 4
                               Description:
        Use time adjustments from location:

         MON           TUE           WED           THU           FRI
      Start  End    Start  End    Start  End    Start  End    Start  End
        :     :       :     :       :     :       :     :       :     :
        :     :       :     :       :     :       :     :       :     :
        :     :       :     :       :     :       :     :       :     :
        :     :       :     :       :     :       :     :       :     :
        :     :       :     :       :     :       :     :       :     :

                              SAT           SUN
                           Start  End    Start  End
                             :     :       :     :
                             :     :       :     :
                             :     :       :     :
                             :     :       :     :
                             :     :       :     :


ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh
```

Description

Provides a description for the table. You can enter a 1 to 27-character alphanumeric table name.The default is blank. Example: Call-ahead Reservations

Number

Displays the table number that you entered on the command line.

Start/End

Defines the range of office hours for each day of the week. Always make sure that the start time is earlier than the end time.

- hour - 0-23
- minute - 0-59

The hour range must be within the specified day, from 00:00 (midnight) until 23:59. If a time range goes past midnight (for example, Friday 19:00 to Saturday 02:00), enter the time in two ranges. Set up the first range as Friday from 19:00 to 23:59 and the second range as Saturday from 00:00 to 01:59.

A time is considered to be in the table from the first second of the start time (for example, 08:00:00). Also, it is still considered to be in the table until the last second of the end time (for example, 17:00:59).

Use time adjustments from location

Points to a field on the **Locations** screen for time zone offset and daylight savings time rule time adjustments.

- The Multiple Locations option must be enabled in order to administer more than one location (locations 2-250).
- You can assign a location to a gateway or to a network region.
- Administer the location where the incoming trunk terminates.

Signaling Group

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, ATM, and SIP Enablement Services (SES) trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the **Group Type** field. Field descriptions are alphabetized for easier reference.

Field descriptions for page 1

Figure 518: Signaling Group screen when the Group Type field is atm

```
add signaling-group nnn                                           Page 1 of x
                               SIGNALING GROUP

Group Number  ___           Group Type:  atm___       Name:
                                               Max Number of NCA TSC: ___
                             D-Channel:         Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
    TSC Supplementary Service Protocol: _       Network Call Transfer? n

CIRCUIT PARAMETERS
     Virtual Path Identifier: 0
  Virtual Channel Identifier: 0

              Signaling Mode: isdn-pri            Circuit Type: T1
                   Idle Code: 11111111                 Connect: network
        Interface Companding: mulaw
            Country Protocol: 1
            Protocol Version: d

                                   DCP/Analog Bearer Capability:
                                           Interworking Message:
```

Figure 519: Signaling Group screen when the Group Type field is h.323

```
add signaling-group n                                     Page 1 of x
                               SIGNALING GROUP

Group Number  ___           Group Type:  h.323
                         Remote Office? n      Max Number of NCA TSC: ___
                                   SBS? y       Max number of CA TSC:  ___
                              IP Video?      Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
    TSC Supplementary Service Protocol: _
                       T303 Timer(sec): 10

         Near-end Node Name:             Far-end Node Name:
       Near-end Listen Port:           Far-end Listen Port:
                                    Far-end Network Region:
               LRQ Required? n     Calls Share IP Signaling Connection? n
               RRQ Required? n
           Media Encryption? y
                 Passphrase:          Bypass If IP Threshold Exceeded? y
                                               H.235 Annex H Required? n
                      DTMF over IP:     Direct IP-IP Audio Connections? y
Link Loss Delay Timer (sec):                     IP Audio Hairpinning? y
Enable Layer 3 Test?                      Interworking Message: PROGress
H.323 Outgoing Direct Media? n     DCP/Analog Bearer Capability:
```

Figure 520: Signaling Group screen when the Group Type field is isdn-pri

```
add signaling-group n                                       Page 1 of x
                              SIGNALING GROUP
Group Number  ___           Group Type:  isdn-pri
                  Associated Signaling?        Max Number of NCA TSC: ___
                     Primary D-Channel:         Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___ X-Mobility/Wireless Type: NONE
    TSC Supplementary Service Protocol: _
```

Figure 521: Signaling Group screen when the Group Type field is sip

```
add signaling-group n                                       Page 1 of x
                               SIGNALING GROUP

Group Number:  ___          Group Type: sip
                      Transport Method: tls

   IP Video?             Priority Video?



  Near-end Node Name:                Far-end Node Name:
Near-end Listen Port:              Far-end Listen Port: ____
                                Far-end Network Region: __
      Far-end Domain: __________________________________


                             Bypass If IP Threshold Exceeded? y

       DTMF over IP: rtp-payload         Direct IP-IP Audio Connections? y
                                                   IP Audio Hairpinning? n

Session Establishment Timer (min):3
```

Associated Signaling

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
y	Enter **y** to use associated signaling.
n	Enter **n** to use non-facility associated signaling.

Bypass If IP Threshold Exceeded

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** to automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below limits administered on the **IP-Options System Parameters** screen.

Calls Share IP Signaling Connection

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter **y** for inter-connection between servers running Avaya Communication Manager. If **y**, then **LRQ Required** must be **n**. This field should be set to **n** if either the local and/or remote server is not running Avaya Communication Manager. **Note:** For an H.323 signaling group type, Avaya recommends a value of **y** for **Enable Layer 3 Test** when **Calls Share IP Signaling Connection** is **y** and the far-end is Communication Manager. When both the near and far-end servers are running Avaya Communication Manager, the value in this field must be the same for both. When you change the value in this field, the system displays the following message: "Far end Communication Manager Signaling-Group must have same value."

Circuit Type

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
T1	Results in page 2 displaying 24 channels.
E1	Results in page 2 displaying 31 channels.

Connect

Appears when the **Group Type** field is **atm**. In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this link.

Valid entries	Usage
host	Enter **host** when the link connects Communication Manager to a computer.
network	Enter **network** when the link connects Communication Manager to a central office or any other public network switch.
pbx	Enter **pbx** if this link is connected to another switch in a private network. If **pbx** is entered, the **Interface** field appears.

Country Protocol

Appears when the **Group Type** field is **atm**. The entry in this field must match the country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using. For a list of country codes, see the Country code table on page 1579.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the **Signaling Mode** field is **isdn-pri**.

D Channel

Appears when the **Group Type** field is **atm**. Enter the necessary characters.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **09** to **32**	Six and seventh characters are the circuit number

DCP/Analog Bearer Capability

Appears when the **Group Type** field is **atm** or **h.323**. This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**.

Valid entries	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability. This is the default.
speech	Provides speech encoding in the information transfer capability.

Direct IP-IP Audio Connections

Appears when the **Group Type** field is **h.323** or **sip**. Allows direct audio connections between H.323 endpoints. For SIP Enablement Services (SES) trunk groups, this is the value that allows direct audio connections between SES endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP (VoIP) transmissions.

DTMF Over IP

Appears when the **Group Type** field is **sip**.

Valid entries	Usage
rtp-payload	Support for SIP Enablement Services (SES) trunks requires the default entry of **rtp-payload**.

Enable Layer 3 Test

Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** if you want Communication Manager to run the Layer 3 test that verifies that all connections known at the near-end are recognized at the far-end. The default value is **y** (test enabled) for new Communication Manager installations, **n** for upgrades from previous releases. **Note:** For an H.323 signaling group type, Avaya recommends a value of **y** for **Enable Layer 3 Test** when **Calls Share IP Signaling Connection** is **y** and the far-end is Communication Manager. If this field is administered as **y** (test enabled) and the **Far-end Node Name** field does not have an administered IP address, then you cannot submit the form, and the Layer 3 test aborts.

Far-end Domain

Appears when the **Group Type** field is **sip**. The number of the network region that is assigned to the far-end of the trunk group. For example, to route SES calls within your enterprise, enter the domain assigned to your proxy server. For external SES calling, the domain name could be that of your SES service provider. If blank, the far-end IP address is used.

Valid entries	Usage
Max. 40 character string	Enter the name of the IP domain for which the far-end proxy is responsible (i.e., authoritative), if different than the near-end domain. If the domains are the same, leave this blank.
blank	Far-end domain is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Listen Port

Appears when the **Group Type** field is **h.323** or **sip**.

Valid entries	Usage
1 to **65535**	Enter the same number as entered in the **Near-end Listen Port** field. Typically, this is the default of **5061** for SIP over TLS.
blank	Far-end listen port is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Network Region

Appears when the **Group Type** field is **h.323** or **sip**. The number of the network region that is assigned to the far-end of the trunk group.

Valid entries	Usage
1 to **250**	Enter the network region number that is assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. Leave blank to select the region of the near-end node.
blank	Far-end network region is unspecified. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Far-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the far-end Control Lan (C-LAN) IP interface used for trunks assigned to this signaling group. The node name must be administered on the **IP Node Names** screen.

Valid entries	Usage
Name of an administered IP node.	Describe the far-end node. Note that If you leave this field blank, the system might display the following message: "Warning: unspecified far-end IP address is vulnerable to denial of service attacks."

Tip:

For SIP Enablement Services (SES) signaling groups, if either the node name or port differs for each group, you have different SES signaling connections, and you should administer a maximum of 10 using TLS. If you administer more than 10 TLS signaling connections, and they are all in use at the same time, the results can be unpredictable. Note that if the node names and ports match, you can administer as many identical SES signaling groups using TLS as desired.

Group Number

A display-only field identifying the signaling group.

Group Type

This field describes the type of protocol to be used with the signaling group.

Valid entries	Usage
atm	Use for Asynchronous Transfer Mode signaling trunks
h.323	Use for h.323 protocol or when using SBS signaling trunks.
isdn-pri	Integrated Service Digital Network Primary Rate Interface
sip	For SIP Enablement Services (SES) on the Avaya S8300, S8400, S8500, and S87XX.

H.235 Annex H Required

Appears only for signaling group type **h.323**.

Valid entries	Usage
y/n	Enter **y** to indicate that the Communication Manager server requires the use of H.235 Annex H (now called H.235.5) protocol for authentication during registration. **NOTE:** If this field is set to **y**, then **LRQ Required** on the **Station** screen must also be set to **y**, or the signaling group will not work. This is because the LRQ is required for the exchanges of authentication data.

H.323 Outgoing Direct Media

Appears only when **Group Type** is **h.323** and **Direct IP-IP Audio Conections** is **y**.

Valid entries	Usage
y	When set to **y**, a call from an H.323 station over a trunk that uses this signaling group starts as a direct media call. The IP address and port of the H.323 station are sent as the media and medic control channel addresses in the SETUP message. **Note**: If, on an outgoing Direct Media call from an IP (H.323) telephone over an H.323 trunk, you attempt to transfer the call, conference another party, or put the call on hold while the call is still in ringing state, the operation will fail.
n	The IP address of the MEDPRO board is sent in the SETUP Message. This is the default.

Idle Code

Appears when the **Group Type** field is **atm**. This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch/server.

Valid entries	Usage
0, **1**	Enter an 8-digit string.

Interface

This field only appears when the **Connect** field is **pbx**. The **Interface** field controls how your server negotiates glare with the far-end switch.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your server overrides the other end when glare occurs. If you are connecting your server to a host computer, set this field to **network**.
user	Enter **user** if your server releases the contested circuit and looks for another when glare occurs. If you are connecting your server to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the **Peer Protocol and Side** fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

Appears when the **Group Type** field is **atm**. Indicates the companding algorithm expected by the system.The entry in this field must match the companding method used by the far-end switch.

Valid entries	Usage
alaw **mulaw**	Enter **alaw** for E-1 service. Enter **mulaw** for T-1 service.

Interworking Message

Appears when the **Group Type** field is **atm**, **h.323**, or **sip**. This field determines what message Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

IP Audio Hairpinning

Appears when the **Group Type** field is **h.323** or **sip**. The **IP Audio Hairpinning** field entry allows the option for H.323 and SIP Enablement Services (SES)-enabled endpoints to be connected through the IP circuit pack in the Avaya S8XXX Server, without going through the time division multiplexing (TDM) bus.

Valid entries	Usage
y/n	Type **y** to enable hairpinning for H.323 or SES trunk groups. Default is **n**.

IP Video

Appears for signaling group type **h.323** and **sip**.

Valid entries	Usage
y/n	Enter **y** to enable IP video capability for this signaling group. Default is **n**.

Link Loss Delay Timer (sec)

Use this field to specify how long to hold the call state information in the event of an IP network failure or disruption. Communication Manager preserves calls and starts this timer at the onset of network disruption (signaling socket failure). If the signaling channel recovers before the timer expires, all call state information is preserved and the signaling channel is recovered. If the signaling channel does not recover before the timer expires, the system:

- raises an alarm against the signaling channel
- maintains all connections with the signaling channel
- discards all call state information about the signaling channel

.

Valid entries	Usage
1 to **180**	Enter the number of seconds to delay the reaction of the call controller to a link bounce. Default is **90**.

LRQ Required

Appears when the **Group Type** field is **h.323**. Allows IP trunk availability to be determined on a per call basis. When this option is enabled, a RAS-Location Request (LRQ) message is sent to the far-end gatekeeper prior to each call over the IP trunk. The far-end gatekeeper responds with a RAS-Location Confirm (LCF) message, and the call proceeds.

Note:

If the **H.235 Annex H Required** field on the **Signaling Group** screen is set to **y**, then **LRQ Required** must also be set to **y**, or the signaling group will not work. This is because the LRQ is required for the exchanges of authentication data.

Valid entries	Usage
y	Enter **y** if **H.235 Annex H Required** is **y**, or if the far-end server is not an Avaya DEFINITY or Avaya S8XXX Server, and requires a location request to obtain a signaling address in its signaling protocol. If this field is set to **y**, **Calls Share IP Signaling Connection** must be **n**.
n	Enter **n** if the far-end server is running Avaya Communication Manager, unless **H.235 Annex H Required** is set to **y**.

Max number of CA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **619**	Maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.

Max number of NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
0 to **256**	Maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Media Encryption

Appears when the Media Encryption feature is enabled in Communication Manager and the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** to enable encryption for trunk calls assigned to this signaling group. If encryption for the signaling group is not enabled, then trunk calls using this signaling group will not be encrypted regardless of IP Codec Set administration.

Name

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
up to 15 alphanumeric characters	Enter 15 alphanumeric characters for identification. **NOTE:** Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Near-end Listen Port

Appears when **Group Type** is **h.323** or **sip**. Defaults to **5061** for SIP Enablement Services (SES) over TLS.

Valid entries	Usage
1719, 1720, or 5000 to **9999**	Enter an unused port number. The default for SIP is **5061**. Avaya recommends **1720** for h.323 and **1719** if LRQ is **y**.

Near-end Node Name

Appears when the value of the entry in the **Group Type** field is **atm** or **sip**. Enter the node name for the Control Lan (C-LAN) IP interface in this Avaya S8XXX Server. The node name must be administered on the **IP Node Names** screen and the **IP Interfaces** screen.

Valid entries	Usage
Name of an administered IP node	Describe the near-end node.

Network Call Transfer

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
y/n	Enter **y** to indicate D-channels are supporting ENCT.

Passphrase

Appears when Media Encryption is enabled or the **H.235 Annex H Required** field is **y**. The passphrase is used for both Media Encryption and authentication. This field cannot be left blank.

Valid entries	Usage
8 to 30 alphanumeric characters	Enter a value for the passphrase used to generate a shared "secret" for symmetric encryption of the media session key. The same passphrase must be assigned to the corresponding signaling groups at both ends of an IP trunk. The passphrase: ● Is case sensitive ● Must contain at least 1 alphabetic and at least 1 numeric ● Valid characters also include letters, numbers, and these symbols: **!&*?;'^(),.:-**

Primary D Channel

Appears when the **Group Type** field is **isdn-pri**.

Valid entries	Usage
Cabinet number (**1** to **44**), Carrier (**A** to **E**), Slot (**00** to **20**), Circuit (**09** to **32**)	Enter the letter or number of the Primary D Channel.

Priority Video

Appears when the **Group Type** field is **h.323** or **sip**. For **sip** signaling groups, this field appears when, on the **System-Parameters Customer Options** screen, **Multimedia SIP Trunking** is **y**.

Valid entries	Usage
y/n	Enter **y** to specify that incoming video calls have an increased likelihood of receiving bandwidth and are also allocated a larger maximum bandwidth per call. Default is **n**.

Protocol Version

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
a **b** **c** **d**	In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, use this field to select the protocol that matches your network service provider's protocol.

RRQ Required

Appears when the **Group Type** field is **h.323**. This field specifies the signaling group that serves as a gateway rather than gatekeeper.

Valid entries	Usage
y/n	Displays **y** if the signaling group serves a remote office (gateway). Displays **n** if the signaling group serves a gatekeeper.

Remote Office

Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
y/n	Enter **y** if the signaling group serves a remote office.

SBS

Appears when the **Group Type** field is set to **h.323**. If you set this to **y**, you must set both the **Trunk Group for NCA TSC** field and the **Trunk Group for Channel Selection** field equal to the signaling group number administered for the SBS trunk group. The **TSC Supplementary Service Protocol** field should always be set to **b** to obtain full QSI.

Valid entries	Usage
y/n	Enter **y** to use SBS signaling trunk groups. The default is **n**.

Session Establishment Timer (min)

Appears when the **Group Type** field is **sip**. This field determines how long the system waits before tearing down a ringing call. The default is **3** minutes.

Valid entries	Usage
3 to **120**	The time in minutes Communication Manager waits before tearing down a ring no answer call.

Signaling Mode

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **isdn-pri**.

T303 Timer (sec)

Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the **Group Type** field is **h.323**.

Valid entries	Usage
2 to **10**	Enter a number of seconds between 2 and 10. Default is 10.

Transport Method

Appears when the **Group Type** field is **sip**.

Valid entries	Usage
tls	The default is **tls**. No other value is supported.

Trunk Group for Channel Selection

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **2000**	For S8300/S87XX Servers.

Trunk Group for NCA TSC

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**.

Valid entries	Usage
1 to **2000** or blank	For Avaya S8300/S87XX Servers.

TSC Supplementary Service Protocol

Appears when the **Group Type** field is **atm**, **h.323**, or **isdn-pri**. Indicate the supplementary service protocol to use for temporary signaling connections.

Valid entries	Usage
a	AT&T, Bellcore, Nortel. When the **Country Code** field on the **DS1 Circuit Pack** screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the **DS1 Circuit Pack** screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the **DS1 Circuit Pack** screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	ISO QSIG. Also, use this entry for SBS signaling groups.
c	ETSI (appears only for **Group Type isdn-pri**)
d	ECMA QSIG

Valid entries	Usage
e	Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the **Trunk Group** screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**.

Virtual Channel Identifier

Appears when the **Group Type** field is **atm**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

Virtual Path Identifier

A display-only field that appears when the **Group Type** field is **atm**. This field always sets to **0**.

Valid entries	Usage
32 to **1023** or blank	Enter a number between 32 and 1023 or blank.

X-Mobility/Wireless Type

Appears when the **Group Type** field is **isdn-pri**. This field indicates the type of X-Mobile endpoints allowed.

Valid entries	Usage
DECT	Indicates to Communication Manager that the remote end of the trunk group controlled by the signaling group is a DECT mobility controller. This allows X-Mobility to work over ISDN-PRI trunks between the server/switch and adjunct.
none	

Field descriptions for page 2

This screen appears only when the **Group Type** is **atm**.

Figure 522: Signaling Group screen (when the Group Type field is atm)

```
add signaling-group next                                         Page 2 of x

SIGNALING GROUP

Chan Port Chan Port
  1:       17:
  2:       18:
  3:       19:
  4:       20:
  5:       21:
  6:       22:
  7:       23:
  8:       24:
  9:
 10:
 11:
 12:
 13:
 14:
 15:
 16:
```

Chan Port

Displays when the **Group Type** field is **atm**. If the **Circuit Type** field on page 1 is **T1**, this field displays 24 channels; if you specified **E1**, it displays 31 channels.

You must fill this screen in for ATM signaling groups. This provides two things:

- It allows you to define fractional T1 and fractional E1 facilities, specifying how many and which channels to use.
- It allows you to choose the port numbers to use (port numbers must be unique for all signaling groups on the same ATM board).

The signaling channel (port 16 for an E1 and port 24 for a T1) must be a port between **9** and **32**. A port number used on this screen cannot be used on any other ATM signaling group on the same board.

The channels used must match exactly the channels used on the other end of the signaling group. For example, if your T1 is set up to use channels 1 through 5, 7, and 24 (the signaling channel), the far end must use channels 1 through 5, 7, and 24.

Valid entries	Usage
009 to **256** or blank	Enter the port number for non-signaling channels.

Signaling Group Administered NCA TSC Assignments page

Figure 523: Signaling Group screen (Administered NCA-TSC Assignment Page)

```
                                                              Page 2 of x
                              ADMINISTERED NCA TSC ASSIGNMENT

Service/Feature: _________________    As-needed Inactivity Time-out (min):
  TSC    Local                                                  Adj.     Mach.
 Index    Ext.    Enabled  Established   Dest. Digits    Appl.   Name      ID
   1:    _____       __     _________   _____________  _______ _______    __
   2:    _____       __     _________   _____________  _______ _______    __
   3:    _____       __     _________   _____________  _______ _______    __
   4:    _____       __     _________   _____________  _______ _______    __
   5:    _____       __     _________   _____________  _______ _______    __
   6:    _____       __     _________   _____________  _______ _______    __
   7:    _____       __     _________   _____________  _______ _______    __
   8:    _____       __     _________   _____________  _______ _______    __
   9:    _____       __     _________   _____________  _______ _______    __
  10:    _____       __     _________   _____________  _______ _______    __
  11:    _____       __     _________   _____________  _______ _______    __
  12:    _____       __     _________   _____________  _______ _______    __
  13:    _____       __     _________   _____________  _______ _______    __
  14:    _____       __     _________   _____________  _______ _______    __
  15:    _____       __     _________   _____________  _______ _______    __
```

Appl.

Specifies the application for this administered NCA-TSC.

Valid entries	Usage
audix	Use this for ISDN-PRI D-channel DCS Audix feature.
dcs	Use this for the DCS Over ISDN-PRI D-channel feature.
gateway	Use this when the administered NCA-TSC is used as one end in the gateway channel. If gateway is entered, then the **ISDN TSC Gateway Channel Assignments** screen must be completed.

Valid entries	Usage
masi	Use this when the NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Use this to convert messages from an administered AUDIX NCA-TSC to a QSIG CISC. If you use this application type, then you must enter a Machine ID between 1 and 20.

As-needed Inactivity Time-out (min)

Valid entries	Usage
10 to **90**, or blank	This field only applies to as-needed NCA-TSCs.

Dest. Digits

Valid entries	Usage
Up to 15 characters **0** to **9**, *****, **#**	Enter the extension of the ISDN interface.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the administered NCA-TSC.

Established

Used to indicate the strategy for establishing this administered NCA-TSC.

Valid entries	Usage
permanent	Use permanent so that the administered NCA-TSC can be established by either the near end or the far end.
as-needed	Use as-needed so that the administered NCA-TSC will be established the first time the administered NCA-TSC is needed; it can be set up either by the near end or far end switch.

Local Ext

Valid entries	Usage
Extension	Enter the extension of the ISDN interface.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 to **20**	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the **Processor Channel** screen.
1 to **63** for DCS **1** to **99** for Audix **1** to **15** for MASI **1** to **20** for QSIG-MWI blank	For S87XX Series IP-PNC.

Service/Feature

Valid entries	Usage
accunet **i800** **inwats** **lds** **mega800** **megacom** **multiquest** **operator** **sdn** **sub-operator** **wats-max-band**	Enter the service or feature being assigned. In addition to pre-defined Services/Features, any user-defined **Facility Type** of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

SIT Treatment for Call Classification

This screen is available when, on the System Parameters Customer-Options (Optional Features) screen, **ASAI Link Core Capabilities** and **ASAI Link Plus Capabilities** are **y**. Use this screen to specify the treatment of Special Information Tones (SITs) used for Outbound Call Management type calls with USA tone characteristics. The port network TN744 Call Classifier circuit pack ports or H.248 Media Gateway internal tone detector resources in classified mode are used to detect SITs. The classifiers are capable of detecting the following SITs:

- SIT Ineffective Other
- SIT Intercept
- SIT No Circuit
- SIT Reorder
- SIT Vacant Code
- SIT Unknown
- AMD (Answering Machine Detected) Treatment

Field descriptions for page 1

Figure 524: Sit Treatment for Call Classification screen

```
change sit-treatment
                 SIT TREATMENT FOR CALL CLASSIFICATION

               SIT Ineffective Other: dropped
                       SIT Intercept: answered
                      SIT No Circuit: dropped
                         SIT Reorder: dropped
                     SIT Vacant Code: dropped
                         SIT Unknown: dropped

                       AMD Treatment: dropped
                Pause Duration (sec): 0.5
                 Talk Duration (sec): 2.0
```

AMD Treatment

Answering Machine Detected. An ASAI adjunct can request AMD for a call. If Answering Machine is detected, one of two treatments is specified. Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

AMD Treatment has two separately administrable subfields: **Talk Duration** is for full seconds and **Pause Duration** is for fractions of a second, separated by a display-only decimal point.

Talk Duration - Defaults to 2.0 seconds and allows a range from 0.1 seconds to 5.0 seconds in increments of 0.1 seconds.

Pause Duration - Defaults to 0.5 seconds and allows a range from 0.1 seconds to 2.0 seconds in increments of 0.1 seconds.

Communication Manager looks for voice energy of at least **Talk Duration** seconds. If it finds that much continuous speech, Communication Manager classifies the call as an answering machine. If it finds a pause of duration as long or longer than **Pause Duration** seconds before then, Communication Manager classifies the call as a live person. So the **Talk Duration** timer should be set to a time longer than it takes to say a typical live greeting, e.g. "XYZ Corporation," but shorter than it takes to say a typical answering machine greeting, e.g. "We can't answer the phone so please leave a message." The **Pause Duration** should be set longer than the typical silence between words in an answering machine greeting, but shorter than the typical space between words in a live greeting, e.g. "Hello ... Hello?."

SIT Ineffective Other

Sample announcement following this SIT - *You are not required to dial a 1 when calling this number.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Intercept

Sample announcement following this SIT - *XXX-XXXX has been changed to YYY-YYYY, please make a note of it.* Default is **answered**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT No Circuit

Sample announcement following this SIT - *All circuits are busy, please try to call again later.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Reorder

Sample announcement following this SIT - *Your call did not go through, please hang up and dial again.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Unknown

A situation or condition that is unknown to the network is encountered. Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

SIT Vacant Code

Sample announcement following this SIT - *Your call cannot be completed as dialed, please check the number and dial again.* Default is **dropped**.

Valid entries	Usage
answered	Enter **answered** to specify that these call are classified as answered, and are therefore sent to an agent.
dropped	Enter **dropped** to specify that these calls are classified as not answered, and are therefore not sent to an agent.

Site Data

Use this screen to enter information about buildings, floors and telephone set colors. You must supply values on this screen before you can enter information in the **Site Data** section of the **Station** screen.

Field descriptions for page 1

Figure 525: Site Data screen

```
change site-data
SITE DATA USER DEFINITION
VALID BUILDING FIELDS



   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
   _________   _________   _________   _________   _________
```

These pages are available for you to enter descriptive information about the buildings, floors and telephone set colors. You can enter any valid keyboard character. If you want to indicate that a particular floor is in a particular building, you must include this in the floor entry, for example, B301-Fl4.

Station

Use the **Station** screen to administer individual telephone sets or virtual telephones. This section provides descriptions of all of the fields that can appear on the **Station** screens. Some of the fields are used for specific telephone types; others are used for all telephone types. The screen examples shown might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. To make it easier to find a specific field description, they are listed in alphabetical order by field name.

Field descriptions for Station screens

Figure 526: Station screen

```
add station next                                              Page 1 of X
                              STATION

Extension:                    Lock Messages? n                        BCC: 0
     Type:                    Security Code:                           TN: 1
     Port:                  Coverage Path 1:                          COR: 1
     Name:                  Coverage Path 2:                          COS: 1
                            Hunt-to Station:

STATION OPTIONS
      XOIP Endpoint type: auto             Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 3
             Data Module? n                      Message Lamp Ext: 1014
            Speakerphone: 2-way               Mute button enabled? y
        Display Language? English
                   Model:                        Expansion Module?

 Survivable GK Node Name:                       Media Complex Ext:
          Survivable COR:                            IP Softphone? y
   Survivable Trunk Dest?                       Remote Office Phone? y
                                                 IP Video Softphone?
                                                           IP Video?

                                                Customizable Labels?
```

Figure 527: Station screen (page 2)

```
change station nnnn                                                Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                        Call Waiting Indication:
 Per Button Ring Control? n                  Attd. Call Waiting Indication:
   Bridged Call Alerting? n                     Idle Appearance Preference? n
        Switchhook Flash? n                   Bridged Idle Line Preference? y
    Ignore Rotary Digits? n                       Restrict Last Appearance? y
  Active Station Ringing: single         Conf/Trans On Primary Appearance? n
                                                        EMU Login Allowed?
        H.320 Conversion? n         Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed      Busy Auto Callback without Flash? y
         Multimedia Mode: basic
    MWI Served User Type: ______                Display Client Redirection? n
        Automatic Moves:
              AUDIX Name:                      Select Last Used Appearance? n
                               _                 Coverage After Forwarding? _
     Recall Rotary Digit? n                        Multimedia Early Answer? n

Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? n
Emergency Location Ext: 75001      Always use? n        IP Audio Hairpinning? n
Precedence Call Waiting? y
```

Figure 528: Station screen (page 3)

```
add station next                                                   Page 3 of x
                                    STATION

          Conf/Trans on Primary Appearance? y
Bridged Appearance Origination Restriction? y
            Call Appearance Display Format: loc-param-default
     IP Phone Group ID:
                             ENHANCED CALL FORWARDING
                                      Forwarded Destination        Active
Unconditional For Internal Calls To:                                  n
                  External Calls To:                                  n
         Busy For Internal Calls To:                                  n
                  External Calls To:                                  n
     No Reply For Internal Calls To:                                  n
                  External Calls To:                                  n

           SAC/CF Override? n
```

Figure 529: Station screen (page 4)

```
add station nnnn                                               Page 4 of X
                                    STATION

     SITE DATA
          Room: _______                                 Headset? n
          Jack: ___                                     Speaker? n
         Cable: ___                                    Mounting: d
         Floor: _______                            Cord Length: 0_
      Building: _______                              Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                 6:limit-call
 2: call-appr                 7:team     Ext: 5381231          Rg:
 3: call-appr                 8:cfwd-enh Ext:
 4: audix-rec  Ext: 4000      9:cfwd-enh Ext: 5502
 5: release                  10:aux-work RC: 1 Group:

voice-mail Number:
```

Figure 530: Station screen (page 5)

```
change station nnnn                                              Page 5 of x
                                      STATION

FEATURE BUTTON ASSIGNMENTS

 9:
10:
11:
12:
13:
14:
15:
16:
17:
18:
19:
20:
21:
22:
23:
24:
```

If the **Expansion Module** field is **y**, an additional page appears.

Figure 531: Station screen (page 6)

```
change station nnnn                                             Page 6 of x
                                          STATION
EXPANSION MODULE BUTTON ASSIGNMENTS

1:                                        13:
2:                                        14:
3:                                        15:
4:                                        16:
5:                                        17:
6:                                        18:
7:                                        19:
8:                                        20:
9:                                        21:
10:                                       22:
11:                                       23:
12:                                       24:
```

Figure 532: SIP Feature Options page

```
change station nnnn                                              Page 5 of x
                                      STATION

SIP Feature Options
         Type of 3PCC Enabled: none
```

The field descriptions for the **Station** screen are listed alphabetically for easy reference.

1-Step Clearing

Valid entries	Usage
y/n	If set to **y**, the call terminates again at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each telephone.

Valid entries	Usage
enhanced	Allows the telephone user to access the enhanced system abbreviated dialing list.
group	Allows the telephone user to access the specified group abbreviated dialing list. If you enter **group**, you also must enter a group number.
personal	Allows the telephone user to access and program their personal abbreviated dialing list. If you enter **personal**, you also must enter a personal list number.
system	Allows the telephone user to access the system abbreviated dialing list.

Access Code

This field appears when a wireless terminal model number is selected in the **Type** field. The Access Code is a temporary, shorter version of the complete User Authentication Key (UAK) required by the system when the terminal is first put into service. It is then used to automatically generate a unique UAK for that wireless terminal over-the-air.

Valid entries	Usage
5-digit decimal number	Enter the 5-digit access code to place the wireless terminal into service. Default is blank.

Active Station Ringing

Defines how call rings to the telephone when it is off-hook. This field does not affect how calls ring at this telephone when the telephone is on-hook.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this telephone to ring continuously.
single	Enter **single** to cause calls to this telephone to receive one ring cycle and then ring silently.
if-busy-single	Enter **if-busy-single** to cause calls to this telephone to ring continuously when the telephone is off-hook and idle and calls to this telephone to receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent	Enter **silent** to cause all calls to this station to just ring silently.

Adjunct Supervision

Adjunct Supervision appears when the **Type** field is **500**, **2500**, **k2500**, **8110**, **ops**, **ds1fd**, **ds1sa**, **VRU**, **VRUFD**, or **VRUSA**.

Valid entries	Usage
y	Enter **y** if an analog disconnect signal is sent automatically to the port after a call terminates. Analog devices (such as answering machines and speakerphones) use this signal to turn the devices off after a call terminates.
n	Set this field to **n** so hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Always Use

This field does not apply to SCCAN wireless telephones, or to extensions administered as type **h.323**.

Valid entries	Usage
y	When this field is **y**: ● The **Remote Softphone Emergency Calls** field is hidden. A softphone can register no matter what emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used. The softphone's user-entered settings are ignored. ● If an IP telephone dials 911, the **Emergency Location Extension** administered on the **Station** screen is used. ● If a call center agent dials 911, the physical station extension is displayed, overriding the **LoginID for ISDN Display** field on the **Agent LoginID** screen.
n	For more information, see the description for the **Emergency Location Extension** field on the **Station** screen. This is the default,

Assigned Member — Ext

The system automatically assigns this extension. This is the extension of the user who has an associated **Data Extension** button and shares the module.

Assigned Member — Name

Display-only field that shows the name associated with the extension shown in the **Assigned Member - Ext** field.

Att. Call Waiting Indication

Attendant call waiting allows attendant-originated or attendant-extended calls to a busy single-line telephone to wait and sends distinctive call-waiting tone to the single-line user. Must be set to **y** when the **Type** field is set to **H.323**. You should not set this field to **y** if the **Data Restriction** field is **y** or the **Switchhook Flash** field is **n**, or if Data Privacy is enabled for the telephone's class of service (COS). If any of these conditions are true, the telephone cannot accept or handle call waiting calls.

Valid entries	Usage
y	Enter **y** to activate Call Waiting (without Caller ID information) for the telephone. Default.
n	Call Waiting is not enabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the **Type** field is **CallrID**.

Audible Message Waiting

The Audible Message Waiting tone indicates that the user has a waiting message. This field appears only if **Audible Message Waiting** is set to **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Note that this field does not control the Message Waiting lamp.

Valid entries	Usage
y/n	Enter **y** if you want the telephone user to receive stutter dial tone when they have a waiting message and they go off-hook.

Audix Name

Specifies which AUDIX is associated with the station.

Valid entries	Usage
Names assigned to an AUDIX adjunct	Must contain a user-defined adjunct name that was previously administered on the **IP Node Names** screen.

Auto-A/D

When **Per Button Ring Control** is **y**, this field appears next to the **call-appr** field in the **BUTTON ASSIGNMENTS** section of the **Station** screen. Use this field to enable automatic abbreviated/delayed ringing for a call appearance.

Valid entries	Usage
y/n	Enter **y** if you want to enable abbreviated/delayed ringing for this call appearance. Default is **n**.

Auto Answer

In EAS environments, the auto answer setting on the **Agent LoginID** screen can override a station's setting when an agent logs in there.

Note:

For analog stations, if Auto Answer is set to **acd** and the station is off-hook and idle, only the ACD split/skill calls and direct agent calls auto answer; non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the Agent receives call-waiting tone.

Valid entries	Usage
all	Enter all to allow all calls (ACD and non-ACD) terminated to an idle station to be cut through immediately. Does not allow automatic hands-free answer for intercom calls. With non-ACD calls, the set is also rung while the call is cut through. The ring can be prevented by activating the **ringer-off** feature button when, on the **Feature-Related System Parameters** screen, the **Allow Ringer-off with Auto-Answer** field is **y**.
acd	Enter **acd** to allow only ACD split /skill calls and direct agent calls to auto answer. If this field is set to **acd**, Non-ACD calls terminated to a station ring audibly.
none	Enter **none** to cause all calls terminated to this station to receive an audible ringing treatment.
icom	Enter **icom** to allow a telephone user to answer an intercom call from the same intercom group without pressing the **intercom** button.

Automatic Moves

Automatic Moves allows a DCP telephone to be unplugged from one location and moved to a new location without additional Communication Manager administration. Communication Manager automatically associates the extension to the new port.

CAUTION:

When a DCP telephone is unplugged and moved to another physical location, the **Emergency Location Extension** field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid entries	Usage
always	Enter **always** and the DCP telephone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	Enter **once** and the DCP telephone can be unplugged and plugged into a new location once. After a move, the field is set to done the next time that routine maintenance runs on the DCP telephone. Use **once** when moving a large number of DCP telephones so each extension is removed from the move list. Use **once** to prevent automatic maintenance replacement.
no	Enter **no** to require administration in order to move the DCP telephone.
done	Done is a display-only value. Communication Manager sets the field to done after the telephone is moved and routine maintenance runs on the DCP telephone.
error	Error is a display-only value. Communication Manager sets the field to error, after routine maintenance runs on the DCP telephone, when a non-serialized telephone is set as a movable telephone.

Auto Select Any Idle Appearance

Valid entries	Usage
y/n	Enter **y** to allow automatic selection of any idle appearance for transferred or conferenced calls. Communication Manager first attempts to find an idle appearance that has the same extension number as the call being transferred or conferenced has. If that attempt fails, Communication Manager selects the first idle appearance.

Automatic Selection of DID Numbers

Communication Manager chooses a 2- to 5-digit extension from a predetermined list of numbers and assigns the extension to a hotel room telephone.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

BCC

Appears when **ISDN-PRI** or **ISDN-BRI Trunks** is enabled on the **System Parameters Customer-Options (Optional Features)** screen. Display-only field set to **0** for stations (that is, indicates voice or voice-grade data).

See "Generalized Route Selection" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking).

Bearer

This field is useful when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. This field appears on the BRI **Station** screen for 8503, 8510, and 8520 stations in Communication Manager 2.1 and 2.2 only. See Secure Terminal Equip on page 1534 for **Bearer** field functionality in Communication Manager 3.0 and later.

Valid entries	Usage
speech	Force the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.
3.1khz	Leave the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Bridged Appearance Origination Restriction

.

Valid entries	Usage
y	Call origination on the bridged appearance is restricted.
n	Call origination ion the bridged appearance is allowed. This is normal behavior, and is the default.

Bridged Call Alerting

Use this field to control how the user is alerted to incoming calls on a bridged appearance.

If **Bridged Call Alerting** is **n** and **Per Button Ring Control** is **n**, audible ringing is suppressed for incoming calls on bridged appearances of another telephone's primary extension.

Valid entries	Usage
y	Enter **y** if you want the bridged appearance to ring when a call arrives at the primary telephone.
n	Enter **n** if you want the bridged appearance to flash but not ring when a call arrives at the primary telephone. This is the default.

Bridged Idle Line Preference

Use this field to specify that the selected line for incoming bridged calls is always an idle line.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the user connects to the ringing call appearance.

Building

Enter a valid building location. See Site Data on page 1490 for valid entries.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Busy Auto Callback without Flash

Appears on the **Station** screen for analog telephones, only if the **Without Flash** field in the **ANALOG BUSY AUTO CALLBACK** section of the **Feature-Related System Parameters** screen is set to **y**. The **Busy Auto Callback without Flash** field then defaults to **y** for all analog telephones that allow Analog Automatic Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

BUTTON ASSIGNMENTS

Enter the abbreviated software name to assign a feature button. For a list of feature buttons, see the table, Telephone Feature Buttons Table on page 134.

Note:

If you want to use Terminal Translation Initialization (TTI), you must assign a call appearance (**call-appr**) to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

You can use this field to identify the cable that connects the telephone jack to the system. You also can enter this information in the Blank column on the Port Assignment Record.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Call Appearance Display Format

This field only appears on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances are not affected by this field. Use this field to specify the display format for the station.

Note:

This screen sets the display value for an individual station. To set display values for an entire location, see the **Display Parameters** screen.

Valid entries	Usage
loc-param-default	The system uses the default value from the **Display Parameters** screen that applies to this station.This is the default.
inter-location	The system displays the complete extension on downloadable call appearance buttons.
intra-location	The system displays a shortened or abbreviated version of the extension on downloadable call appearance buttons.

Caller ID Message Waiting Indication

Appears when the **Type** field is **CallrID**. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow aliasing of various non-Avaya telephones and adjuncts.

Note:

The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON-lamp Avaya Communication Manager Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID telephone with the **Caller ID Message Waiting Indication** field set to **n** and the **Message Waiting Indicator** field set to **neon**.

Calls Allowed

Appears if the **XMOBILE Type** field is **EC500** and the **Mapping Mode** field is **termination** or **both**. Used to identify the Extension to Cellular call filter type for an XMOBILE station. This field allows an XMOBILE station to function as a bridge and still be restricted.

Valid entries	Usage
internal	External calls are blocked. Internal calls terminate to the XMOBILE station. Attendant-originated and attendant-delivered calls are not delivered
external	Internal calls are blocked. External calls terminate to the XMOBILE station.
all	All calls terminate to the XMOBILE station.
none	Prevents calls from terminating to the XMOBILE station. Can be used to prevent business-related calls from accruing telephone charges on cellular telephones that are lost, being transferred to a new user, or being disabled for other business reasons.

Note:

Interswitch calls on DCS trunks are treated as internal calls. When the **Calls Allowed** field is set to **internal** or **all**, DCS calls are delivered to the cell telephone. When the **Calls Allowed** field is set to **external** or **none**, DCS calls are not delivered.

Incoming calls from other Extension to Cellular users are internal if office caller ID is enabled for the XMOBILE station associated with the cell telephone. When the **Calls Allowed** field is set to **internal** or **all**, calls from other Extension to Cellular users are delivered. When the **Calls Allowed** field is set to **external** or **none**, calls from other Extension to Cellular users are not delivered.

The calling party receives busy treatment when call filtering blocks calls to a standalone XMOBILE. Calls delivered to standalone XMOBILE stations that are not answered will follow the call coverage or call forwarding paths administered for the standalone XMOBILE.

Call Waiting Indication

This allows user, attendant-originated, and outside calls to a busy single-line telephone to wait and sends a distinctive call-waiting tone to the single-line user. This feature is denied if **Data Restriction** is **y** or **Switchhook Flash** is **n**, or if Data Privacy is active by way of the telephone COS assignment. Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y	Enter **y** to activate Call Waiting (without Caller ID information) for the telephone. Default.
n	Call Waiting is not enabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the **Type** field is **CallrID**.

CDR Privacy

This option allows digits in the called number field of an outgoing call record to be blanked, on a per-station basis. You administer the number of blocked digits system-wide in the **Privacy - Digits to Hide** field on the **CDR System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to enable Call Privacy for each station.

Cell Phone Number

Contains the unformatted cell telephone's published external number. This field can contain a 3-digit area code plus the 7-digit main number. If the same Cell Phone Number is administered on multiple XMOBILE **Station** screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
0 to **9**	Enter 1 to15 digits. Avaya recommends that you enter a full 10-digit Cell Phone Number regardless of whether the cell telephone is local or not. Note that your **ARS** screen has to be administered to handle this.

Conf/Trans On Primary Appearance

This feature forces the use of a primary appearance when the held call to be conferenced or transferred is a bridge. This is regardless of the **Auto Select Any Idle Appearance** field.

Valid entries	Usage
y/n	Enter **y** to specify that the primary call appearance is always activated for a bridged transfer or conference.

Configuration Set

This field is used to administer the Configuration Set number that contains the call treatment options desired for the XMOBILE station. This field must be administered if:

- The **XMOBILE Type** field is **EC500**.
- The **Mobility Trunk Group** field is a trunk group number and the administered trunk group is non-DECT or non-PHS.
- The **Mobility Trunk Group** field is **aar** or **ars**.

If the **Mobility Trunk Group** field is a trunk group number and the administered trunk group is **DECT** or **PHS**, this field can be left blank.

Valid entries	Usage
1 to **10** or blank	Enter any value corresponding to the appropriate **Configuration Set** screen.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

Cord Length

The length of the cord attached to the receiver. This is a free-form entry, and can be in any measurement units.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

COS

Enter the desired Class of Service (COS) number to select allowed features.

Country Protocol

Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Valid entries	Usage
1	United States (Bellcore National ISDN)
2	Australia
etsi	ETSI (Europe)
3	Japan
6	Singapore

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the **System Parameters - Call Coverage/Call Forwarding** screen.
n	No coverage treatment is provided after forwarding regardless of the value of the **Coverage After Forwarding** field on the **System Parameters - Call Coverage/Call Forwarding** screen.
s(ystem)	Indicates that call processing uses the **Coverage After Forwarding** field on the **System Parameters Call Coverage/Call Forwarding** screen. To override the system-wide parameter for a given station, set this field to **y** or **n**.

Coverage Msg Retrieval

Applies if the telephone is enabled for LWC Reception.

Valid entries	Usage
y/n	Enter **y** to allow users in the telephone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this telephone.

Coverage Module

Valid entries	Usage
y	Enter **y** to indicate that a coverage module is connected to the station. Once you enter **y**, the system displays an additional page that allows you to assign the buttons for the module.

Coverage Path 1 or Coverage Path 2

Enter a coverage-path number or time-of-day table number from a previously-administered **Call Coverage Path** screen or **Time of Day Coverage Table** screen.

Note:

If Modified Misoperation is active (**Misoperation Alerting** is **y** on the **Feature-Related System Parameters** screen), you must assign a Coverage Path to all stations on Communication Manager.

CRV Length

Only for ASAI stations. Enter **1** or **2** to indicate the length of CRV for each interface.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Customizable Labels

Use this field to enable the Increase Text for Feature Buttons feature for this station. This feature expands the text labels associated with Abbreviated Dial buttons from the current five uppercase alphanumeric characters to a maximum of 13 upper and lower case alphanumeric characters.This field allows you to ensure that there will always be sufficient button customization resources to support VIP users. This field appears when **Type** is one of the following:

- 2410 (Release 2 or later)
- 2420 (Release 4 or later)
- 4610 (IP Telephone Release 2.2 or later)
- 4620 (IP Telephone Release 2.2 or later)
- 4621 (IP Telephone Release 2.2 or later)
- 4622 (IP Telephone Release 2.2 or later)
- 4625 (IP Telephone Release 3.1 or later)

Valid entries	Usage
y	Enter **y** to allow the user of this station to program and store feature button labels with up to 13 alphanumeric characters. This is the default.
n	Enter **n** to disable the Increase Text for Feature Buttons feature for this station.

Data Extension

Enter the extension assigned to the data module.

Data Module

Valid entries	Usage
y/n	Enter **y** if this telephone has an associated data module. When set to **y**, a **Data Module** page is added to the **Station** screen for defining the data module parameters.

Data Option

Valid entries	Usage
analog **data module** **none**	If a second line on the telephone is administered on the I-2 channel, enter **analog**. Otherwise, enter **data module** if applicable or **none**.

Data Restriction

Data restriction provides permanent protection and cannot be changed by the telephone user. Do not assign a Data Restriction if **Auto Answer** is **all** or **acd**. If **y**, whisper page to this station is denied.

Valid entries	Usage
y/n	Enter **y** to prevent tones, such as call-waiting tones, from interrupting data calls.

Default Dialing Abbreviated Dialing Dial Code

Appears only when the **Special Dialing Option** is set to **default**. Enter a list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number. The data call originator also can perform data-terminal dialing by specifying a dial string that might or might not contain alphanumeric names.

Dial Prefix

Contains the unformatted sequence of digits or characters that are prepended to the cell telephone's published cell telephone number before dialing. If the same Cell Phone Number is administered on multiple XMOBILE **Station** screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
up to 4 digits: **0** to **9**, *****, **#**	Enter 1 to 4 digits.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Display Caller ID

Appears when the **Type** field is **CallrID**. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow transmission of calling party information to the Caller ID telephone or adjunct.

Display Cartridge

For 7404 D telephones only. Enter **y** to indicate there is a display cartridge associated with the station. This displays an additional page to allow you to assign display buttons for the display cartridge.

Display Client Redirection

Only administrable if **Hospitality** is enabled on the **System Parameters Customer-Options (Optional Features)** screen. This field affects the telephone display on calls that originated from a station with Client Room Class of Service.

Note:

For stations with an audix station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, **Display Client Redirection** must be set to **y**.

Valid entries	Usage
y	When set to **y**, the redirection information for a call originating from a Client Room and terminating to this station displays.
n	When set to **n**, this station's display does not show the redirection information for all calls originating from a Client Room (even redirected calls) that terminate to this station. Only the client name and extension (or room, depending on what is administered on the **Hospitality** screen) display.

Display Language

Use this field to specify the language in which information is displayed on stations. To view the dial pad letter/number/symbol mapping tables used for the integrated directory, see "Telephone Display" in Feature Description and Implementation for Avaya Communication Manager, 555-245-205.

Tip:

Time of day is displayed in 24-hour format (00:00 - 23:59) for all languages except **english**, which is displayed in 12-hour format (12:00 a.m. to 11:59 p.m.).To display time in 24-hour format and display messages in English, set the **Display Language** field to **unicode**. When you enter **unicode**, the station displays time in 24-hour format, and if no Unicode file is installed, displays messages in English by default. For more information on Unicode, see Administering Unicode display on page 203.

Valid entries	Usage
english	Enter the language you want the user to see on their station display.
french	
italian	
spanish	
user-defined	
unicode	**Note:** Unicode display is only available for Unicode-supported telephones. Currently, 4610SW, 4620SW, 4621SW, 4622SW, Sage, Spark, and 9600-series telephones (Avaya one-X Deskphone Edition SIP R2 or later) support Unicode display. Unicode is also an option for DP1020 (aka 2420J) and SP1020 (Toshiba SIP Phone) telephones when **Display Character Set** on the **System Parameters Country-Options** screen is set to **katakana**. To administer Unicode on the SP1020, use the 4624 station type. For more information on Unicode language displays, see Administering Unicode display on page 203.

Distinctive Audible Alert

Valid entries	Usage
y/n	Enter **y** so the telephone can receive the 3 different types of ringing patterns which identify the type of incoming calls. Distinctive ringing might not work properly for off-premises telephones.

Emergency Location Ext

The **Emergency Location Ext** field defaults to the telephone's extension. This extension identifies the street address or nearby location when an emergency call is made. For more information about the use of this field, see the **Usage** description for the **Remote Softphone Emergency Calls** field later in this section.

Valid entries	Usage
1 to 8 digits	Enter the Emergency Location Extension for this station.

Note:

On the **ARS Digit Analysis Table** screen, you must administer 911 to be call type **emer** or **alrt** in order for the E911 Emergency feature to work properly.

EMU Login Allowed

Valid entries	Usage
y/n	Enter **y** to allow the station to be used as a visited station by an Enterprise Mobility User (EMU) visitor user. Default is **n**. For more information about Enterprise Mobility User, see Setting Up Enterprise Mobility User on page 171.

Endpt ID

Appears only if **Endpt Init** is **y**. Enter a unique 2-digit number (**00** to **62**) for this endpoint. Each **Endpt ID** field must have a unique value for each endpoint on the same port.

This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires the last 2 digits of the Service Profile Identifier (SPID) be between 00 and 63 and be binary unique for each endpoint.

For WorldClass BRI (WCBRI) data modules only, this field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires the SPID programmed into the endpoint contain at least 9 digits. For example, if the SPID is **1234**, and **Endpt ID** is **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.

Endpt Init

Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the SPID, administered into the system, and entered into the ISDN-BRI terminal. For an ISDN-BRI terminal to be operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. Therefore, the SPID of new or reused terminals must be programmed to match the administered SPID value. Appears only if **MIM Support** is **y** and indicates the terminal's endpoint initialization capability.

Valid entries	Usage
y	Enter **y** if the terminal supports Bellcore ISDN-1 terminal initialization procedures.
n	Enter **n** for all other country protocols.

Expansion Module

When this field is **y**, an **Expansion Module Buttons Assignment** page is added to the **Station** screen for administering buttons for the expansion module.

Valid entries	Usage
y/n	Enter **y** if this telephone has an expansion module. This will enable you to administer the buttons for the expansion module.

Extension

Displays the extension for this station—this is either the extension you specified in the station command or the next available extension (if you used **`add station next`**).

For a virtual extension, enter a valid physical extension or a blank. Blank allows an incoming call to the virtual extension to be redirected to the virtual extension's "busy" or "all" coverage path.

> Avaya recommends that you consider the display for emergency notification when you complete the **Name** field on the **Station** screen. Put the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the **Name** field appear on the first display line, followed by the extension. The second line contains the last 3 characters of the **Name** field, followed by the word "EMERGENCY." Characters 18 through 24 of the **Name** field do not appear at all.

Feature Module

Enter **y** to indicate the station is connected to a feature module. Entering **y** displays an additional page to allow you to assign feature buttons to the module.

Fixed TEI

This field appears only for ISDN-BRI data modules, NI-BRI telephones, WCBRI data modules, and ASAI links. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the **TEI** field. For **ASAI**, enter **y**.

Floor

The floor location of this station.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Forwarded Destination

For each of the three types of enhanced call forwarding (**Unconditional**, **Busy**, and **No Reply**), enter the destination extension for both internal and external calls. Valid entries can be up to 18 digits, and the first digit can be an asterisk (*). Blank is also a valid entry. In the **Active** field, indicate whether the specific destination is active (**y**) or inactive (**n**).

H.320 Conversion

Allows H.320 compliant calls made to this telephone to be converted to voice-only. Because the system can only handle a limited number of conversion calls, you might need to limit the number of telephones with H.320 conversion.

Headset

Indicates if the telephone has a headset.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Home

This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
y/n	Indicate the roaming status of the wireless user. This field will specify whether the system is the user's home or roaming system. The default is **y**.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Appears only when **Special Dialing Option** is **hot-line**.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

HOT LINE DESTINATION — Abbreviated Dialing List Number

Enter the abbreviated dialing list where you stored the hotline destination number.

HOT LINE DESTINATION — Dial Code

Enter the dial code in the specified abbreviated dialing list where you stored the hotline destination number.

Hunt-to Station

Enter the extension the system should hunt to for this telephone when the telephone is busy. This field allows you to create a station hunting chain (by assigning a hunt-to station to a series of telephones).

Idle/Active Ringing (Callmaster)

Defines how call rings to the telephone when it is on-hook. This field applies to CALLMASTER telephones.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this telephone to ring continuously.
if-busy-single	Enter **if-busy-single** to cause calls to this telephone to ring continuously when the telephone is off-hook and idle and calls to this telephone to receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent-if-busy	Enter **silent-if-busy** to cause calls to ring silently when this station is busy.
single	Enter **single** to cause calls to this telephone to receive one ring cycle and then ring silently.

Idle Appearance Preference

Use this field to specify that the selected line for incoming calls is always an idle line.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

If this field is **y**, the short switch-hook flash (50 to 150) from a 2500-type set is ignored.

Valid entries	Usage
y	Enter **y** to indicate that rotary digits from the set should be ignored.
n	Enter **n** to make sure they are not ignored.

IPEI

International Portable Equipment Identifier. This field appears when a wireless terminal model number is selected in the **Type** field.

Valid entries	Usage
9-character hexadecimal number; **0** to **9**, **a** to **f**, **A** to **F**, or blank	Enter the unique ID number of the wireless terminal.

IP Audio Hairpinning

Appears when Group Type is **h.323** or **sip**. Allows IP endpoints to be connected through the IP circuit pack in the server, without going through the time division multiplexing (TDM) bus.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the server/switch in IP format, without going through the Avaya DEFINITY TDM bus. Default is **n**.

IP Phone Group ID

Appears for H.323 station types.

Valid entries	Usage
0 to **999** or blank	Enter the Group ID number for this station.

IP Softphone

Appears only for DCP station types and IP Telephones.

Valid entries	Usage
y/n	Enter **y** indicate that this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

IP Video

Appears only for station type **h.323**.

Valid entries	Usage
y/n	Enter **y** to indicate that this extension has IP video capability.

IP Video Softphone

Appears only when **IP Softphone?** is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this extension is a video softphone.

ITC (Information Transfer Capability)

Indicates the type of transmission facilities to be used for ISDN calls originated from this endpoint. The field does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	If you set to **restricted**, either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	If you set to **unrestricted**, only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Jack

Alpha-numeric identification of the jack used for this station.

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Lock Messages

Valid entries	Usage
y/n	Enter **y** to restrict other users from reading or canceling the voice messages or retrieving messages via Voice Message Retrieval.

Loss Group

This field determines which administered 2-party row in the loss plan applies to each station. Does not appear for stations that do not use loss (for example, x-mobile stations and MASI terminals).

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plans.

LWC Activation

Valid entries	Usage
y/n	Enter **y** to allow internal telephone users to leave short LWC messages for this extension. If the system has hospitality, enter **y** for guest-room telephones if the extension designated to receive failed wakeup messages should receive LWC messages that indicate the wakeup calls failed. Enter **y** if **LWC Reception** is **audix**.

LWC Log External Calls

Appears only where the **LWC Reception** field is available. When external calls are not answered, Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
y/n	Enter **y** to make unanswered external call logs available to end-users. Each record consists of the latest call attempt date and time.

LWC Reception

Valid entries	Usage
audix	Enter **audix** if the messages are stored on the Audio Information Exchange System.
none	Enter **none** if LWC messages will not be stored.
spe	Enter **spe** if LWC messages are stored in the system or on the switch processor element (spe).

Mapping Mode

Controls the mode of operation in which the cell telephone will operate when mapped to this XMOBILE extension. An XMOBILE station can be bridged to a deskset. These restrictions/ modes exist because the COR of a bridge is ignored; instead the principal's COR is used. This field allows an XMOBILE station to function as a bridge and still be restricted.

When a cell telephone is mapped to more than one XMOBILE station, then only one of the mapped XMOBILE station can have **origination** or **both** as its Mapping Mode. Therefore, only one of the XMOBILE stations mapped to the cell telephone number is permitted to originate calls.

Valid entries	Usage
both	The cell telephone can be used to originate and terminate calls from its associated XMOBILE extension. This is the default when the **XMOBILE Type** field is **PHS** or **DECT**.
none	The XMOBILE station is disabled administratively and cannot originate and terminate calls from its associated internal extension.

Valid entries	Usage
origination	The cell telephone can be used only to originate calls from its associated internal XMOBILE extension by dialing into the office server running Communication Manager.
termination	The cell telephone can be used only to terminate calls from its associated internal XMOBILE extension. This is the default when the **XMOBILE Type** field is **EC500**.

Map-to Station

This is the physical telephone used for calls to a virtual extension. Do not use an xmobile, xdid or another virtual extension.

Valid entries	Usage
Valid extension	Enter the extension of the physical telephone used for calls to a virtual extension.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

For an IP Telephone or an IP Softphone, this is the extension already administered as an H.323 station type. You can administer this field if the **IP Station** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**.

Valid entries	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Lamp Ext

Enter the extension of the station you want to track with the message waiting lamp.

Message Server Name

Specifies which Message Server is associated with the station.

Valid entries	Usage
Names administered in alphabetical order	Must contain a user-defined adjunct name that was previously administered on the **IP Node Names** screen.

Message Waiting Indicator

This field appears only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, H.323 and VRU telephones. (This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID telephones.) Must be set to a value other than **none** when the **Type** field is set to **H.323**.

Valid entries	Usage
led	Enter **led** if the message waiting indicator is a light-emitting diode (LED).
neon	Enter **neon** if the message waiting indicator is a neon indicator. **Note:** The **neon** message waiting indicator is supported only on a small subset of boards, including older US-only boards, such as the TN746 and the TN793. When you select this option, the following warning appears: "WARNING: neon requires specific hardware/admin support." Check the documentation for your board to see if neon is supported. Note that this option is only available if **Analog Ringing Cadence** on the **Location Parameters** screen is set to **1** (US).
none	No message waiting indicator is selected. This is the default.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other then endpoint initialization. Appears only if **MIM Support** is **y**.

MIM Support (Management Information Message Support)

This field appears only for ISDN-BRI data modules and ASAI. This field supports MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Valid entries	Usage
y	Enter **y** to display Endpt Init and MIM Mtce/Mgt.
n	Enter **n** for **ASAI**.

Mobility Trunk Group

This field associates the XMOBILE station to a trunk.

Valid entries	Usage
2000 (S87XX Server) **99** (DEFINITY CSI, S8300 Server) blank	Enter a valid trunk group number for mobility routing. This trunk group is used for routing. The **Configuration Set** field can be blank if the trunk group is **DECT** or **PHS**. If the trunk group is non-DECT or non-PHS, administer the **Configuration Set** field.
aar	The routing capabilities of Communication Manager will be used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the Avaya S8XXX Server. It will provide ringback to the calling company and might eventually go to coverage.
ars	The routing capabilities of Communication Manager will be used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call will not be extended out of the Avaya S8XXX Server. It will provide ringback to the calling company and might eventually go to coverage.

Model

This field appears only for NI-BRI telephones.

Valid entries	Usage
L-3 Communication STE	The NI-BRI telephone is a model L-3 Communication STE.
Tone Commander	The NI-BRI telephone is a model 6210 and 6220 Tone Commander.
Other	The NI-BRI telephone is another model (for example, a Nortel 5317T).

Mounting

Indicates whether the station mounting is **d**(esk) or **w**(all).

Note:
Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Multimedia Early Answer

Allows you to establish multimedia early answer on a station-by-station basis.

Valid entries	Usage
y/n	If this station will receive coverage calls for multimedia complexes, but is not multimedia-capable, enter **y** to ensure that calls are converted and talk path is established before ringing at this station.

Mute Button Enabled

Valid entries	Usage
y/n	Enter **y** to allow the user to use the **Mute** button on this telephone.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid entries	Usage
fp-mwi	Use if the station is a served user of an fp-mwi message center.
qsig-mwi	Use if the station is a served user of a qsig-mwi message center.
blank	Leave blank if you do not want to audit the served user's MWI or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Enter a name for the person associated with this telephone or data module. The system uses the **Name** field to create the integrated directory.

Note:

For 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones, the **Name** field is supported by Unicode language display. For more information on Unicode language display, see Administering Unicode display on page 203.

Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Off Premises Station

Analog telephones only.

Valid entries	Usage
y	Enter **y** if this telephone is not located in the same building with the system. If you enter **y**, you must complete R Balance Network.
n	Enter **n** if the telephone is located in the same building with the system.

PCOL/TEG Call Alerting

Appears only for 510 telephones.

Valid entries	Usage
y/n	Enter **y** to alert the station for Personal CO Line/Terminating Extension Group calls.

Per Button Ring Control

Valid entries	Usage
y	Enter **y** to allow users to select ring behavior individually for each **call-appr**, **brdg-appr**, or **abrdg-appr** on the station and to enable Automatic Abbreviated and Delayed ring transition for each **call-appr** on the station. Also, enter **y** if you do not want the system to automatically move the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Enter **n** if you want calls on **call-appr** buttons to always ring the station and calls on **brdg-appr** or **abrdg-appr** buttons to always ring or not ring based on the **Bridged Call Alerting** field value. Also, enter **n** if you want the system to move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. Users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing telephone from other telephones in the same area. For virtual stations, this field dictates the ringing pattern on its mapped-to physical telephone.

Enter a Personalized Ringing Pattern. (L = 530 Hz, M = 750 Hz, and H = 1060 Hz).

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Valid entries	Usage
y	All outgoing calls from the station will deliver the Calling Party Number (CPN) information as "Presentation Allowed."
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station will deliver the Calling Party Number information as "Presentation Restricted."
blank	The sending of CPN information for calls is controlled by any administration on the outgoing trunk group the calls are carried on.

Port

Enter 7 characters to specify a port, or an x. If this extension is registered as an IP endpoint, this field will display ***sXXXXXX***, where ***XXXXXX*** is the number of previously registered IP stations. For example, if there are 312 IP sets already registered when you register, your extension would get port **s000313**.

Valid entries	Usage
01 to **64**	First and second numbers are the cabinet number
A to **E**	Third character is the carrier
01 to **20**	Fourth and fifth characters are the slot number
01 to **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have a non-IP set. Or, the extension had a non-IP set, and it dissociated. Use **x** for AWOH and CTI stations, as well as for SBS Extensions.
IP	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have an IP set. This is automatically entered for certain IP station set types, but can be entered for a DCP set with softphone permissions. This changes to the s00000 type when the set registers.

For DCP sets, the port can only be assigned once. ISDN-BRI provides a multipoint configuration capability that allows a previously assigned port to be specified more than once as follows: 2 stand-alone voice endpoints, 2 stand-alone data endpoints, or 1 integrated voice and data endpoint.

However, for the following cases, the port is assumed to be fully assigned:

- Maximum number of users (currently 2) are assigned on the port.
- One of the users on the port is a fixed TEI station.
- One of the users on the port has B-channel voice and B-channel data capability.
- One of the users on the port has no SPID assigned, which includes telephones that have no SPID initialization capability.
- This field is display-only for H.323 set types and 4600 set types. The system assigns an "IP" when the station is first administered.

Note:

Port 1720 is turned off by default to minimize denial of service situations. This applies to all IP softphones release 5.2 or later. You can change this setting, if you have root privileges on the system, by typing the command: **`/opt/ecs/sbin ACL 1720 on`** or **`off`**.

Note:

To set up paging on an H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the **Station** screen. No entries on the **Loudspeaker Paging** screen are required.

Precedence Call Waiting

Valid entries	Usage
y/n	Enter **y** to activate Precedence Call Waiting for this station.

R Balance Network

Valid entries	Usage
y	Enter **y** to select the R Balance Capacitor network. In all other cases except for those listed under **n**, enter **y**.
n	Enter **n** to select the standard resistor capacitor network. You must complete this field if **Off-Premise Station** is **y**. Enter **n** when the station port circuit is connected to terminal equipment (such as SLC carriers or impedance compensators) optioned for 600-ohm input impedance and the distance to the equipment from the system is less than 3,000 feet.

Recall Rotary Digit

This field only appears if type is 500 or 2500.

Valid entries	Usage
y/n	Enter **y** to allow the user of a rotary telephone to dial the administered Recall Rotary Digit to receive recall dial tone. This will enable this user to perform conference and transfer operations. You establish the Recall Rotary Digit on the **Feature-Related System Parameters** screen.

Redirect Notification

Valid entries	Usage
y/n	Enter **y** to give a half ring at this telephone when calls to this extension are redirected (via Call Forwarding or Call Coverage). Enter **y** if **LWC Reception** is **audix**.

Remote Office Phone

Valid entries	Usage
y/n	Enter **y** to use this station as an endpoint in a remote office configuration.

Remote Softphone Emergency Calls

Use this field to tell Communication Manager how to handle emergency calls from the IP telephone. This field appears when either the **IP Softphone** field or the **Remote Office Station** field is set to **y** on the **Station** screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. Please contact your Avaya representative if you have questions about emergency calls from IP telephones.

<table>
<tr><th>Valid entries</th><th>Usage</th></tr>
<tr><td>as-on-local</td><td>Type as-on-local to achieve the following results:
● If the administrator chooses to leave the Emergency Location Extension fields (that correspond to this station's IP address) on the IP Address Mapping screen blank, the value as-on-local sends the extension entered in the Emergency Location Extension field in the Station screen to the Public Safety Answering Point (PSAP).
● If the administrator populates the IP Address Mapping screen with emergency numbers, the value as-on-local functions as follows:
- If the Emergency Location Extension field in the Station screen is the same as the Emergency Location Extension field in the IP Address Mapping screen, the value as-on-local sends the extension to the Public Safety Answering Point (PSAP).
- If the Emergency Location Extension field in the Station screen is different from the Emergency Location Extension field in the IP Address Mapping screen, the value as-on-local sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).</td></tr>
<tr><td>block</td><td>Enter block to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the Avaya S8XXX Server than an adjacent area code served by the same 911 Tandem office.
When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.</td></tr>
<tr><td></td><td>1 of 2</td></tr>
</table>

Valid entries	Usage
cesid	Enter **cesid** to allow Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the Avaya S8XXX Server that an emergency call routed over the it's trunk reaches the PSAP that covers the server or switch. If the server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP telephone in the office automatically selects **extension**.
	2 of 2

Restrict Last Appearance

Valid entries	Usage
y/n	Enter **y** to restrict the last idle call appearance for incoming priority calls and outgoing call originations only.

Rg

When **Per Button Ring Control** is **y**, this field appears next to the **call-appr** field in the **BUTTON ASSIGNMENTS** section of the **Station** screen.

Valid entries	Usage
a(bbreviated-ring) **d(elayed-ring)** **n(o-ring)** **r(ing)**	Enter the desired type of automatic abbreviated/delayed ringing for this call appearance. Default is **r**.

Room

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Valid entries	Usage
Up to 10 characters	To identify the telephone location.
Up to 5 characters	To identify the guest room number, if this station is one of several to be assigned a guest room and the **Display Room Information in Call Display** is **y** on the **Hospitality-Related System Parameters** screen.

SAC/CF Override

This feature allows the user of a station with a **Team** button administered, who is monitoring another station, to directly reach the monitored station by pushing the **Team** button. This overrides any currently active rerouting (e.g., Send All Calls, Call Forwarding) on the monitored station.

Valid entries	Usage
Ask	The system asks if the user wants to follow the rerouting or override it. When the user has the option to decide whether rerouting should take place or not, a message is sent to the station which displays the active rerouting and the number of the forwarded to station. The user of the monitoring station can follow the rerouting by dialing “1,” or by letting the timer that supervises the **Team** button push expire. The user can override the active rerouting by dialing “0,” or by pushing the **Team** button once again.
No	Cannot override rerouting. The station does not have the ability to override the rerouting of a monitored station.
Yes	Can override rerouting. The station has the ability to override the rerouting the monitored station has set, as long as one incoming call appearance is free.

Secure Terminal Equip

This field is useful when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. This field appears on the BRI **Station** screen for 8503, 8510, and 8520 stations in Communication Manager 3.0 and later. See Bearer on page 1501 for **Bearer** field functionality in Communication Manager 2.1 and 2.2.

Valid entries	Usage
n	Force the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.
y	Leave the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Security Code

Enter the security code required by users for specific system features and functions, including the following: Personal Station Access, Redirection of Calls Coverage Off-Net, Leave Word Calling, Extended Call Forwarding, Station Lock, Message Retrieval, Terminal Self-Administration, and Demand Printing. The required security code length is determined by **Minimum Security Code Length** on the **Feature-Related System Parameters** screen.

Select Last Used Appearance

Valid entries	Usage
y	Enter **y** to indicate a station’s line selection is not to be moved from the currently selected line button to a different, non-alerting line button. If you enter **y**, the line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	Enter **n** so the line selection on an on-hook station with no alerting calls can be moved to a different line button, which might be serving a different extension.

Service Link Mode

The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and a server running Avaya Communication Manager which terminates the H.320 protocol. When the user receives or makes a call during a multimedia or IP Softphone or IP Telephone session, a "service link" is established.

Valid entries	Usage
as-needed	Use this setting for most multimedia, IP Softphone, or IP Telephone users. Setting the Service Link Mode to as-needed leaves the service link connected for 10 seconds after the user ends a call so that they can immediately place or take another call. After 10 seconds the link is dropped and a new link would have to be established to place or take another call.
permanent	Use **permanent** for busy call center agents and other users who are constantly placing or receiving multimedia, IP Softphone, or IP Telephone calls. In permanent mode, the service link stays up for the duration of the multimedia, IP Softphone, or IP Telephone application session.

Set Color

Indicatates the set color. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Speaker

Indicates whether the station is equipped with a speaker.

Note:

Fields in the SITE DATA area of the **Station** screen document information related to the station set installation.

Speakerphone

This field controls the behavior of speakerphones.

Valid entries	Usage
1-way	Enter **1-way** to indicate that you want the speakerphone to be listen-only.
2-way	Enter **2-way** to indicate that you want the speakerphone to be both talk and listen.
grp-listen	Group Listen works with only 6400-series and 2420/2410 telephones. Group Listen allows a telephone user to talk and listen to another party with the handset or headset while the telephone's 2-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party via the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party.
none	

Special Dialing Option

This field identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	Causes the **HOT LINE DESTINATION — Abbreviated Dialing Dial Code** field to appear, for administering a hot line destination. The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.
default	Causes the **Default Dialing Abbreviated Dialing Dial Code** field to appear, for entering a list number associated with the abbreviated dialing list. When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number.
blank	For regular (normal) keyboard dialing.

SPID — (Service Profile Identifier)

Enter a variable length parameter. This field appears only if the **Endpt Init** field is **y**.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port are blocked.

Note:

If you have set the **Port** field to **X** for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Survivable COR

This field is for all analog and IP station types. Use this field to set a level of restriction for stations to be used in conjunction with the survivable dial plan in order to limit certain users to only to certain types of calls. The restriction levels are listed in order from most restrictive to least restrictive. Each level assumes the calling ability of the ones above it.

Valid entries	Usage
emergency	This station can only be used to place emergency calls which are defined.
internal	This station can only make intra-switch calls. This is the default.
local	This station can only make calls that are defined as **locl**, **op**, **svc**, or **hnpa** in the **Survivable ARS Analysis Table**
toll	This station can place any national toll calls which are defined as **fnpa** or **natl** on the **Survivable ARS Analysis Table**.
unrestricted	This station can place a call to any number defined in the **Survivable ARS Analysis Table**. Those strings marked as **deny** are also denied to these users as well.

Survivable GK Node Name

Appears only when the **Type** field is **46xx**. Use this field to indicate the gatekeeper to register with when the gateway unregisters (loses call control) with the main server. The media gateway delivers the gatekeeper list to IP endpoints, allowing them to register and subsequently originate/receive calls from other endpoints in this survivable calling zone.

Valid entries	Usage
valid IP node name or blank	Enter any valid IP node name as administered on the **IP Node Names** screen to allow the station to be registered as an IP telephone, associated to an H.323 gateway that is capable of supporting a gatekeeper in survivable mode. Default is blank.

Survivable Trunk Dest

This field is for all analog and IP station types. Use this field to designate certain telephones as not being allowed to receive incoming trunk calls when the Media Gateway is in survivable mode.

Valid entries	Usage
y	Enter **y** to allow stations to be incoming trunk destinations while the Media Gateway is running in survivability mode. This is the default.
n	Enter **n** to prevent this station from receiving incoming trunk calls when in survivable mode.

Switchhook Flash

Must be set to **y** when the **Type** field is set to **H.323**.

Valid entries	Usage
y	Enter **y** to allow users to use the switchhook flash function to activate Conference/Transfer/Hold and Call Waiting.
n	Enter **n** to disable the flash function so that when the switchhook is pressed while active on a call, the call drops. If this field is **n**, you must set **Call Waiting Indication** to **n**.

TEI

Appears only when the **Fixed TEI** field is **y**.

Valid entries	Usage
0 to **63**	1 or 2-digit number

Tests

Valid entries	Usage
y	Enter **y** to enable port maintenance tests.
n	If the equipment (dictaphone) connected to the port does not support these tests, you must enter **n**.

Time of Day Lock Table

Use this field to assign the station to a TOD Lock/Unlock table.

Valid entries	Usage
1 to **5**, or blank	The default is blank, indicating no TOD Lock/Unlock feature is active. The assigned table must have administered a valid time interval entry, and the **Table Active** field on the **Time of Day Lock Table** screen must be set to **y**.

TN

Enter the Tenant Partition number. Also, SBS Extensions can be partitioned.

Type

For each station that you want to add to your system, you must specify the type of telephone in the **Type** field. This is how you distinguish between the many different types of telephones.

The following table lists the telephones, virtual telephones, and personal computers that you can administer on Avaya Communication Manager. To administer telephones that are not in the table, use the **Alias Station** screen.

Note:

You cannot change an analog telephone administered with hardware to a virtual extension if **TTI** is **y** on the **Feature-Related System Parameters** screen. Contact your Avaya representative for more information.

To set up paging on an H.248 gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the **Station** screen. No entries on the **Loudspeaker Paging** screen are required.

Table 19: Telephones

Telephone type	Model	Administer as
Single-line analog	500	500
	2500, 2500 w/ Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
		1 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multiappearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
		2 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	7317H	7317H
Multiappearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
Multiappearance digital	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
		3 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+ **Note:** When adding a new 4601 IP telephone, you must use the 4601+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4601, you receive a warning message, stating that you should upgrade to the 4601+ station type in order to access the Automatic Callback feature.	4601+
	4602+ **Note:** When adding a new 4602 IP telephone, you must use the 4602+ station type. This station type enables the Automatic Callback feature. When making a change to an existing 4602, you receive a warning message, stating that you should upgrade to the 4602+ station type in order to access the Automatic Callback feature.	4602+
	4606	4606
	4610	4610

4 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	4612	4612
	4620SW IP (G3.5 hardware) **Note:** Effective December 5, 2005, Avaya will no longer make 4620 IP telephones commercially available. The 4621SW IP telephone is an appropriate replacement. The 4621SW IP telephone, generally available since May 2005, offers the same functionality as the 4620 and adds a backlit display. For more information, see 4621SW IP telephone.	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
IP Telephone	9610	9610 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4606.
	9620	9620 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4610.
	9630	9630 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
		5 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	9640	9640 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
	9650	9650 **Note:** If your version of Communication Manager is earlier than version 4.0, administer as 4620.
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware • 4620SIP with SIP firmware • 4620SIPCC (Call Center) • Avaya one-X (tm) Deskphone 9620, 9630, 9630G 9640, 96 40G with SIP firmware • SIP Softphone/Avaya one-X Desktop • 1616SIP (Call Center) • Toshiba SP-1020A **Note**: Any model telephone that has SIP firmware and is being used for SIP networking must be administered as a 4620SIP telephone.	4620SIP
IP SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	Any NI-BRI (N1 and N2) telephone	NI-BRI
	7505D	7505D
		6 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		XDID (use when Avaya Communication Manager later assigns a DID number to this station) XDIDVIP (use when the administrator later assigns a DID number to this station) virtual (use to map this and other extensions to one physical telephone)
Key telephone system interface	—	K2500
ASAI	asai link computer telephony adjunct link	asai adjlk
AWOH	any digital set CTI station	same as "Multiappearance Digital" see table above CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
		7 of 8

Table 19: Telephones (continued)

Telephone type	Model	Administer as
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs
		8 of 8

Type of 3PCC Enabled

Valid entries	Usage
Avaya or **none**	Enter the phone number of the off-PBX telephone. Default is none.

Voice mail Number

This field supports the voice mail retrieval feature as a fixed feature button on type 24xx, 46xx, and 96xx telephones. When you enter a number in this field, the telephone's fixed voice mail retrieval button acts an autodial button, dialing the number entered in this field to access voice mail.

Note:

If this field is left blank, the telephone's fixed voice mail retrieval button acts as a "transfer to voice mail" button, which only works for INTUITY Audix or QSIG-integrated voice mail systems. Avaya recommends that for INTUITY Audix and QSIG-integrated voice mail systems, this field should be left blank. For DEFINITY AUDIX and non-QSIG integrated voice mail systems, this field should be filled in with the appropriate number.

Valid entries	Usage
digits (1 to 9) ***** **#** **~p (pause)** **~w/~ (wait)** **~m (mark)** **~s (suppress)**	Enter the complete voice mail number up to 15 digits.

XID

Appears only for an ISDN-BRI data module or an ASAI link. Used to identify Layer 2 XID testing capability.

XMOBILE Type

When the **Type** field is **XMOBILE**, the **Mobility Trunk Group** field must be administered.

Valid entries	Usage
DECT	For the DECT Access System or the AGCS (ROAMEO) IS-136 (TDMA cellular).
EC500	For any public cellular networks.
PHS	For the DENSO 300M.

XOIP Endpoint type

This field appears when a valid analog station is entered in the **Type** field. Use this field for older or non-standard external equipment such as modems, fax, and TTY devices, which are not easily recognized by VoIP resources within Communication Manager. By identifying this external equipment through administration, VoIP firmware can determine whether to immediately attempt to put a call in pass-through mode, or allow the system to handle it normally. Supported station types are 500, 2500, K2500 and CallrID.

Valid entries	Usage
auto **modem** **fax** **tty**	Enter the type of analog endpoint for this station. Default is **auto**. **Note**: This field is intended for exception cases only. For the majority of stations, use the default setting of auto.

Stations With Off-PBX Telephone Integration

Use the **Stations with Off-PBX Telephone Integration** screen to map an office phone to a cell phone through the Extension to Cellular feature. The office phone can be a standard office number or an administration without hardware (AWOH) station.

For more information on Extension to Cellular, see *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Field descriptions for page 1

Figure 533: Stations with Off-PBX Telephone Integration screen, page 1

```
add off-pbx-telephone station-mapping                                  Page 1 of 2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station     Application   Dial     CC    Phone Number          Trunk       Config
Extension                 Prefix                               Selection   Set
43001         EC500       ____ -   1     9736831204            ars         1
43001         OPS         ____ -         12345                 ars         5
43009         EMU         ____ -         67890                 aar         2
43011         CSP         ____ -   998   6095343211            ars         3
_______       ______      ____ -         _______________       ____        __
_______       ______      ____ -         _______________       ____        __
_______       ______      ____ -         _______________       ____        __
```

Command parameters

Action	Object	Qualifier
add	**off-pbx-telephone station-mapping**	
change	**off-pbx-telephone station-mapping**	***<station extension>***
display	**off-pbx-telephone station-mapping**	***<station extension>***
list	**off-pbx-telephone station-mapping**	***<variable>***

The **add off-pbx-telephone station-mapping** command displays the blank **Stations with Off-PBX Integration** screens. You can add up to sixteen associations between an office phone and an external phone.

The **`change off-pbx-telephone station-mapping`** ***`<station extension>`*** command displays the **Stations with Off-PBX Integration** screens. You can change the associations between office telephones and external telephones. The first line on the screen contains the information for the station extension that you entered as the command variable. You can also add additional associations in this screen.

The **`display off-pbx-telephone station-mapping`** ***`<station extension>`*** command displays the **Stations with Off-PBX Integration** screens. The ***`<station extension>`*** variable is not mandatory These screens list up to sixteen entries, starting with the station extension you entered as the command variable. If this extension is not administered for an off-PBX, the display starts with the next administered off-PBX extension in numerical order.

The **`list off-pbx-telephone station-mapping`** ***`<variable>`*** command information about the association between an office phone and an off-PBX phone. The command variable specifies the office phone number or numbers of interest. The ***`<variable>`*** can be:

- a complete phone number
- a partial phone number followed by an asterisk, which is a “wildcard” character
- blank

Station Extension

The **Station Extension** field is an administered extension in your dial plan. This number is the extension of the office phone.

Valid entries	Usage
a valid number in your dial plan	Type an extension number of the office phone up to eight digits. Default is blank.

Application

Indicate the type of off-PBX application that is associated with the office phone. You can assign more than one application to an office phone.

Valid entries	Usage
blank	Default is blank.
CSP	cell phone with Extension to Cellular provided by the cellular service provider
EC500	cell phone with Extension to Cellular
HEMU	Home Enterprise Mobility User
OPS	SIP Enablement Services (SES)-enabled phone
PBFMC	Public Fixed Mobile Convergence
PVFMC	Private Fixed Mobile Convergence
SCCAN	wireless SIP Enablement Services (SES) phone and cell phone
VEMU	Visited Enterprise Mobility User
VIEMU	Visited Initial Enterprise Mobility User

CC

Valid entries	Usage
up to three digits using **0–9**, or blank	Enter the Country Code associated with the extension. Multiple entries that use the same phone number must also have the same Country Code, including entries on XMOBILE **Station** screens. Country Code changes made to existing stations or XMOBILE entries are applied to all instances of the phone number. SAFE (Self-Administered Feature Access Code for EC500) is not recommended on an extension that has an administered Country Code. Origination mapping can occur with or without a country code. Default is blank.

Dial Prefix

The system prepends the **Dial Prefix** to the off-PBX phone number before dialing the off-PBX phone. The system deletes the dial prefix when a user enters their cell phone number using the Self Administration Feature (SAFE) access code. You must set the routing tables properly so that the dial prefix "1" is not necessary for correct routing.

Valid entries	Usage
blank **0–9**, *****, **#**	Type up to four digits, including “*” or “**#**”. If included, “*” or “**#**” must be in the first digit position. Enter a "1" if the phone number is long-distance. Enter "011" if the phone number is international. Default is blank.

Phone Number

Enter the phone number of the off-PBX phone.

Valid entries	Usage
0–9	Enter the phone number of the off-PBX phone. May be blank for the first EC500, CSP or PBFMC phones administered. May be blank for EC500, CSP, PBFMC, which support SAFE (Self-Administered Feature Access Code for EC500). Default is blank.

Trunk Selection

Valid entries	Usage
ars **aar** **trunk group number**	Indicate which trunk group to use for outgoing calls.

Configuration Set

Use the **Configuration Set** field to administer the Configuration Set number. This number contains the desired call treatment options for the Extension to Cellular station. Ninety-nine Configuration Sets exist.

The SCCAN application requires two different configuration sets selected for each station. The first set is the value for the WLAN followed by a slash. The second is the value for the cellular network.

Valid entries	Usage
1–99	Type the number of the Configuration Set. Shows blank for Enterprise Mobility User (EMU). Default is blank.

Field descriptions for page 2

Finish the administration steps to map an office phone to an off-PBX phone on the second page of the **Stations with Off-PBX Telephone Integration** screen (Figure 534: Stations with Off-PBX Telephone Integration screen, page 2 on page 1552). The information you entered in the first page appears as display-only information on the second page.

Figure 534: Stations with Off-PBX Telephone Integration screen, page 2

```
add off-pbx-telephone station-mapping                                Page 2 of x
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station          Call        Mapping         Calls        Bridged
Extension        Limit       Mode            Allowed      Calls
43001            2           both            all          both
43001            2           both            all          both
43009            2           both            all          both
43011            2           both            all          both
43013            2/2         both            all          both
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
                 __          ___________     ________     ___________
```

Call Limit

Valid entries	Usage
blank **1–10**	Set the maximum number of Extension to Cellular (EC500) calls that can be active simultaneously at a single station. Default is **2** for EC500, CSP, OPS, PBFMC, PVFMC.

Mapping Mode

Enter the mode of operation for the Extension to Cellular cell phone. Use these modes to control the degree of integration between the cell phone and the office phone. The modes are valid for Extension to Cellular calls only. For each office phone, you can only assign one cell phone as the **origination** mode. You cannot assign a cell phone as either the **origination** or **both** mode more than once.

Valid entries	Usage
both	Default = **both** when the **Phone Number** field was previously administered for another extension with a **Mapping Mode** of **termination** or **none**. Default = **termination** when the **Phone Number** field was previously administered with a **Mapping Mode** of **origination** or **both**. In the both mode, users can originate and receive calls from the office phone with the cell phone.
termination	In termination mode, users can only use their Extension to Cellular cell phone to receive calls from the associated office phone. Users cannot use the cell phone to originate calls from the associated office phone. Calls originating from the cell phone independent of the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular.
origination	In origination mode, users can only originate Extension to Cellular cell phone calls from the associated office phone. Users cannot use the cell phone to receive calls from the associated office phone.
none	In the none mode, users cannot originate or receive calls from the office phone with the cell phone.

Calls Allowed

Identify the call filter type for an Extension to Cellular station. The **Calls Allowed** values filter the type of calls to the office phone that a user can receive on an Extension to Cellular cell phone.

Valid entries	Usage
all	Default is **all**. The cell phone receives both internal and external calls.
internal	The cell phone receives only internal calls.
external	The cell phone receives only external calls.
none	The cell phone does not receive any calls made to the associated office phone.

Bridged Calls

Use the **Bridged Calls** field to determine if bridged call appearances extend to the Extension to Cellular cell phone. The valid entry definitions are the same as the **Mapping Mode** field entries.

Valid entries	Usage
both **termination** **origination** **none**	Default for OPS and PVFMC applications is **none**. Default for other applications is **both**.

Survivable ARS Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Field descriptions for page 1

Figure 535: Survivable ARA Analysis screen

```
change survivable-ars-analysis                                  Page 1 of X

                              SURVIVABLE ARS ANALYSIS TABLE

                               Total        Call                  Trunk
Dialed String                  Length       Type        Deny      Grp No
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
__________________________       __         ____        _____     _____
```

Call Type

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party. This field cannot be left blank if the **Dialed String** field is not blank.

Valid entries	Usage
emer **npa** **hnpa** **intl** **iop** **locl** **natl** **op** **svc**	Enter the call type to be used for this dialed string.

Deny

The system denies a dialed string that does not match an entered pattern. If there are no available trunks when a match is performed for the given route option, then the user receives a trunk busy indication.

Valid entries	Usage
y/n	Indicate whether or not the dialed string should be blocked.

Dialed String

Valid entries	Usage
up to 18 characters, **0-9**, *, **x**, or **X**	Enter the dialed string digits to be analyzed.

Total Length

This field defines the minimum number of digits required to validate the route. The minimum value when the dial string is populated is the length of the dialed string entry with a maximum value up to **28**. This field cannot be left blank if the **Dialed String** field is not blank.

Valid entries	Usage
0 to **28**	Enter the minimum number of digits required to validate the route. Default is blank.

Trunk Grp No

Valid entries	Usage
Valid trunk group number	Enter the trunk group number that specifies the destination route for the dial plan analysis of this dialed string.

Survivable Processor

Use the **Survivable Processor** screen to add information specific to a Local Survivable Processor (LSP) or to connect certain adjuncts to an LSP or an ESS server. Before administering this screen, you must first assign node names for each LSP and ESS server on the **IP Node Names** screen.

While this screen is administered on the active main server, the information entered applies only to LSPs and ESSs and does not apply to main servers. When translations are copied to an LSP or ESS, the LSP/ESS replaces like translations for the main server with the overrides administered on the **Survivable Processor** screens. That is, use the **Survivable Processor** screen to administer overrides against adjunct links that have already been administered for the main server(s). For more information on Processor Ethernet, see Setting up Processor Ethernet on page 593. For more information about ESS, see *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

Note:

The ESS server is first administered on the **IP Node Names** screen and then on the **System Parameters - ESS** screen before it is administered on the **Survivable Processor** screen. The information from the **System Parameters - ESS** screen is automatically copied to the **Survivable Processor** screen. You do not need to use the **Survivable Processor** screen for an ESS server unless one of the supported adjuncts connects to the PE interface of the ESS server.

Field descriptions for page 1

The first page of the **Survivable Processor** screen displays the Processor Ethernet interface information for the LSP or the ESS server. The information includes the node name, the IP address, the server type, the cluster ID, and the network region. The only administrable field on this page is the **Network Region** field.

Figure 536: Survivable Processor - Processor Ethernet screen

```
add survivable-processor node-name                                page 1 of x

                SURVIVABLE PROCESSOR - PROCESSOR ETHERNET

                    Node Name: node-name
                   IP Address: 135. 9. 72.106
                           ID: 30
                         Type: LSP


               Network Region: 1
```

ID

This display-only field displays the server ID number of the ESS or LSP. Valid entries are **1** to **256**.

IP Address

This display-only field shows the IP address of the node name as it appears in the **IP Node Names** screen.

Network Region

Valid entries	Usage
1 to **250**	Enter the network region in which the PE interface of the LSP or ESS server resides.

Node Name

This display-only field shows the node name that was entered on the **IP Node Names** screen.

Type

This display-only field shows the survivable processor type.

Survivable Processor - Processor Channels page

Use this page of the **Survivable Processor** screen to administer processor channels.

Figure 537: Survivable Processor - Processor Channels screen

```
                                                                  page 2 of x

                      SURVIVABLE PROCESSOR - PROCESSOR CHANNELS

Proc                         Interface            Destination       Session
Chan  Enable  Appl.    Mode  Link/Chan          Node         Port   Local/Remote

 1      i     mis       s     9   5001       cmshost          0        1      1
 1      i     ccr       s     10  5002       ccrhost1         0        1      1
```

Appl

This display-only field identifies the server application type or adjunct connection used on this channel.

Valid entries	Usage
mis	CMS channel assignments.
ccr	CCR channel assignments.

Destination Node

This field identifies the server or adjunct at the far end of this link. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
valid administered node name	Enter an adjunct name, server name, far end IP address, node name, or leave blank for services local to this Avaya S8XXX Server.

Destination Port

This field specifies the far-end port number of this link. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
0, **5000** to **64500**	Enter the number of the destination port. An entry of **0** means any port can be used.

Enable

Use this field to specify how data for this processor channel is transferred to the survivable processor.

Valid entries	Usage
i(nherit)	Enter **i(nherit)** to indicate that this link is to be inherited by the LSP or ESS server. When you enter **i(nherit)**, all fields in the row for this processor channel change to display-only. The survivable processor inherits this processor channel just as it is administered on the main server. You generally use the **i(nherit)** option in the following situations: ● The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity ● The main server connects to the adjuncts using the PE interface of the main server, and you want the LSP or ESS server to connect to the adjunct using its PE interface.
n(o)	This processor channel is disabled on the LSP or the ESS server. The survivable processor does not use this channel. This is the default.
o(verwrite)	The survivable processor overwrites the processor channel information sent in the file sync from the main server. The **o(verwrite)** option The **o(verwrite)** option allows the administered adjunct attributes to be modified uniquely for each individual LSP or ESS server. Use the remaining editable fields to enter the processor channel information for the LSP or ESS server.

Interface Channel

This field identifies the channel number or the TCP/IP listen port channel to carry this processor (virtual) channel. For TCP/IP, interface channel numbers are in the range **5000** to **64500**. Avaya recommends the value **5001** for CMS, and **5003** for DCS. This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
0, **5000** to **64500**	For **ethernet** or **ppp**. The channel number **0** means any port can be used.

Interface Link

This field identifies the communication interface link carrying this processor (virtual) channel. A **p** in this field indicates that the link is the Processor Ethernet interface. Otherwise, the CLAN link number is used. When you enter **o(verwrite)** in the **Enable** field, the value of this field changes to **p** (processor).

Mode

Indicate whether the IP session is passive (client) or active (server). This field changes to display-only when you enter **i(nherit)** in the **Enable** field, showing the value administered on the main server.

Valid entries	Usage
c(lient)	Indicate whether the IP session is passive **c(lient)** or active **s(erver)**.
s(erver)	

Proc Chan

This display-only field shows the processor channel number used for this link when it was administered on the **Processor Channel Assignment** screen.

Session - Local/Remote

Local and Remote Session numbers must be consistent between endpoints. These fields change to display-only when you enter **i(nherit)** in the **Enable** field, showing the values administered on the main server.

Valid entries	Usage
1 to **256** (si) **1** to **384** (r) or blank	For each connection, the Local Session number on this Avaya S8XXX Server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections.

Survivable Processor - IP Services page

Use this page of the **Survivable Processor** screen when an AESVCS or a CDR connects to the LSP or ESS server that was identified on the **Survivable Processor - Processor Ethernet** screen. If the AESVCS or the CDR is administered on the **IP Services** screen, it automatically appears on the **Survivable Processor - IP-Services** screen.

Figure 538: Survivable Processor - IP-Services screen

```
                                                                 page 3 of x

                     SURVIVABLE PROCESSOR - IP-SERVICES

Service    Enabled   Store    Local         Local    Remote        Remote
 Type                to dsk    Node          Port     Node          Port
 CDR1        o         y     gert_clan6       0      cdr_1          9003
 CDR2        i         y     gert_clan1       0      cdr_rsp        9000
```

Enabled

Use this field to specify how data for each specified service type is transferred to the survivable processor.

Valid entries	Usage
i(nherit)	Enter **i(nherit)** to indicate that this link is to be inherited by the LSP or ESS server. When you enter **i(nherit)**, all fields in the row for this service type change to display-only. The survivable processor inherits this service type just as it is administered on the main server. You generally use the **i(nherit)** option in the following situations: ● The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity ● The main server connects to the adjuncts using the main server's PE interface and you want the LSP or ESS server to connect to the adjunct using its PE interface
n(o)	This IP services link is disabled on the LSP or the ESS server. This is the default.
o(verwrite)	Enter **o(verwrite)** to overwrite the processor channel information sent in the file sync from the main server. The overwrite option allows the administered CDR or AE Services attributes to be modified uniquely for each individual LSP or ESS server. Entering **o(verwrite)** causes the **Local Node** field to change to **procr**. Use the remaining editable fields to enter the information for the LSP or ESS server.

Local Node

This display-only field contains the node name as defined on the IP Node Names screen.

Local Port

This display-only field contains the originating port number. For client applications such as Call Detail Recording (CDR), this field defaults to **0**.

Remote Node

This field becomes editable when you enter **o(verwrite)** in the **Enable** field. Specify the name at the far end of the link for the CDR. The remote node should not be defined as a link on the IP Interfaces or Data Module screens. This field does not apply for AESVCS.

Remote Port

This field becomes editable when you enter **o(verwrite)** in the **Enable** field. Specify the port number of the destination. Valid entries range from **5000** to **65500** for CDR or AESVCS. The remote port number must match the port administered on the CDR or AESVCS server.

Service Type

This field is display-only and reflects the value administered in the **Service Type** field on the **IP Services** screen. Valid entries include **CDR1 or CDR2**, and **AESVCS**.

Note:

For service type **CDR1** and **CDR2**, if the **Enable** field on the **Survivable Processor - IP Services** screen is **n**, the corresponding CDR1/CDR2 entry (for the Primary or Secondary CDR link) is removed from the **System Parameters CDR** screen on the LSP or the ESS server. The primary must be up and running before adding the secondary. The secondary must be removed first before removing the primary. On duplicated ESS, for CDR1/CDR2/AESVCS, the **Enabled** field defaults to **i**, and the rest of the fields are display-only.

Store to dsk

Use this field to enable or disable the storage of the CDR data on the local hard drive of the LSP or ESS. This column only pertains to rows which have the **Service Type** set to **CDR1** or **CDR2**.

Valid entries	Usage
y/n	Enter **y** to enable storage of CDR data on the local hard drive of an LSP or ESS.

Survivable Processor - IP-Services - Session Layer Timers page

This page appears if **CDR1** or **CDR2** is administered on the SURVIVABLE PROCESSOR - IP-SERVICES page of the **Survivable Processor** screen. These fields are only administrable if the **Enable** field on that page is set to **o(verwrite)**.

Figure 539: Survivable Processor - IP-Services - Session Layer Timers screen

```
                                                              page 4 of x

           SURVIVABLE PROCESSOR - IP-SERVICES - Session Layer Timers

Service    Reliable    Packet Resp    Session Connect   SPDU    Connectivity
 Type      Protocol      Timer         Message Cntr     Cntr       Time

 CDR1         n            30                3            3          60
 CDR2         y            30                3            3          60
```

Connectivity Time

Use this field to set the amount of time that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the **Survivable Processor** screen is set to **o(verwrite)**.

Valid entries	Usage
1 to **255**	Enter the desired interval in seconds. Default is **60**.

Packet Resp Timer

Use this field to specify the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the **Survivable Processor** screen is set to **o(verwrite)**.

Valid entries	Usage
1 to **255**	Enter the desired interval in seconds. Default is **30**.

Reliable Protocol

Use this field to indicate whether you want to use a reliable protocol over this link. This field is only administrable if the **Enable** field on the SURVIVABLE PROCESSOR - IP-SERVICES page of the **Survivable Processor** screen is set to **o(verwrite)**.

Valid entries	Usage
y/n	Enter **y** to indicate that you want to Use reliable protocol if the adjunct on the far end of the link supports it. Default is **y**.

Service Type

This field is display-only and corresponds to the **Service Type** entry on the SURVIVABLE PROCESSOR - IP-SERVICES page of the **Survivable Processor** screen.

Session Connect Message Cntr

The Session Connect Message counter indicates the number of times Communication Manager tries to establish a connection with the far-end adjunct.

Valid entries	Usage
1 to **5**	Enter the desired number of connection attempts.

SPDU Cntr

The Session Protocol Data Unit (SPDU) counter indicates the number of times Communication Manager transmits a unit of protocol data before generating an error.

Valid entries	Usage
1 to **5**	Enter the desired number of transmission attempts.

System Capacity

The **System Capacity** screen (command **`display capacity`**) is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431. Detailed system capacity information can be found in *System Capacities Table for Avaya Communication Manager on Avaya Media Servers*, 03-300511.

System Configuration

The **System Configuration** screen shows all the boards on your system that are available for connecting telephones. You can see the board number, board type, circuit-pack type, and status of each board's ports.

Figure 540: System Configuration screen

```
                         SYSTEM CONFIGURATION
Board                                                 Assigned Ports
Number     Board Type       Code    Vintage u=unassigned t=tti p=psa

01A05     DIGITAL LINE       TN754B 000002  01 u  03 u  05 u  07 08
01A06     ANALOG LINE        TN742  000010  01 02 03 04 u  u  u  u
01B05     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C04     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C05     DIGITAL LINE       TN2224 000004  01 u  u  04 u  u  07 08
                                            u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C06     HYBRID LINE        TN762B 000004  01 02 u  u  u  u  u  u
01C10     DIGITAL LINE       TN754  000004  u  u  u  u  u  u  u  u
```

System Parameters Call Coverage/Call Forwarding

This screen sets the system-wide parameters for call coverage and call forwarding.

Field descriptions for page 1

Figure 541: System-Parameters — Call Coverage/Call Forwarding screen

```
 change system-parameters coverage-forwarding                     page 1 of x

         SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
           Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
         Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
                           Coverage - Caller Response Interval (seconds): _
     Threshold for Blocking Off-Net Redirection of Incoming Trunks Calls: 1
                           Location for Covered and Forwarded Calls: called
COVERAGE
                                         Keep Held SBA at Coverage Point? _
        External Coverage Treatment for Transferred Incoming Trunk Calls? _
         Immediate Redirection on Receipt of PROGRESS Inband Information? _
                                               Maintain SBA At Principal? _
              QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? _
                                            Station Hunt Before Coverage? n

FORWARDING
                                                   Call Forward Override? _
                                               Coverage After Forwarding? _
```

CALL COVERAGE / FORWARDING PARAMETERS

Coverage - Caller Response Interval (seconds)

The time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point. The calling party either can hang up, use Leave Word Calling, or press the **Go to Cover** button during this time interval.

Valid entries	Usage
0 to **10**	Enter the time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point.

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at a local coverage point before a call redirects to the next coverage point
- the number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated

Valid entries	Usage
1 to **99**	Enter the desired number of rings. **Note:** When ringing local destinations (i.e., in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Location for Covered and Forwarded Calls

Valid entries	Usage
called	Default. • If the called party is registered or in-service, coverage and forwarding use the called party's physical phone's location number. • If the called party is AWOH (x-port) or unregistered, coverage and forwarding use the location number that corresponds to the ARS FAC column on the Locations (change locations) screen. • When the forwarding or coverage destination is to UDP instead of to an external destination starting with the ARS FAC, routing is always based on the caller's physical phone's location regardless of how this field is administered.
caller	Coverage and forwarding use the caller's physical phone's location number.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at an off-net coverage point before a call is redirected to the next coverage point
- the number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.

Valid entries	Usage
1 to **99**	Enter the desired number of rings. **Note:** When ringing local destinations (i.e., in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Note:

When ringing local destinations (i.e., in an office environment), a short interval often is appropriate because the intended party either is near the telephone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the telephone. If the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

This field applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.

Valid entries	Usage
1 to **7**	The number of allowed calls to be routed off-net before blocking commences.
n (all)	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal can be redirected off-net.

COVERAGE

External Coverage Treatment for Transferred Incoming Trunk Calls

This field governs how an transferred incoming trunk call is handled if the call redirects to coverage.

Valid entries	Usage
y	Enter **y** to allow external coverage treatment for incoming trunk calls that redirect to coverage.
n	Enter **n** to allow internal coverage treatment for incoming trunk calls that redirect to coverage

Immediate Redirection on Receipt of PROGRESS Inband Information

This field appears only if one of the following is true:

- The **Coverage of Calls Redirected Off-Net Enabled** field on the **System Parameters Coverage/Forwarding** screen is **y**.
- The **Value-Added Avaya (VALU)** field on the System Parameters Customer-Options (Optional Features), Page 6, screen is **y**.

This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities. Some cellular telephone providers send an ISDN PROGRESS message with the **Progress Indicator** field set to ‘inband information’ when a cellular telephone is unavailable to receive a call. In these circumstances, the message indicates that an announcement is being played to the originating party and we should move the call immediately to the next available coverage point.

However, a PROGRESS message with a Progress Indicator of ‘inband information’ might be received for reasons other than an unavailable cellular telephone. In this case, you probably do not want to redirect the call to the next coverage point.

There is no way for Avaya Communication Manager to know the actual intent of such a PROGRESS message, yet you might choose how the system should handle this message. It is essentially an educated, but blind, choice and you should be aware that there will be instances when this choice leads to inappropriate call handling.

Avaya Communication Manager queries this field on receipt of a qualifying PROGRESS message and acts according to your instruction on how to treat it.

As a guide, users in European countries following the ETSI standard and redirecting to GSM cellular telephones might want to consider setting this field to **y**.

In the United States, PROGRESS messages with the **Progress Indicator** field set to ‘inband information’ are sent for a variety of reasons not associated with unavailable cellular telephones and you should set this field to **n**.

Valid entries	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point.
n	Do not immediately redirect an off-net coverage/forwarded call to the next coverage point.

Keep Held SBA at Coverage Point

This field governs how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid entries	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but might enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

Maintain SBA At Principal

Allows a user to maintain a simulated bridged appearance when a call redirects to coverage.

Valid entries	Usage
y	Enter **y** to maintain a simulated bridged appearance (SBA) on the principal's telephone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	When set to **n**, no SBA is maintained on the principal's telephone. DCS with rerouting will be attempted, and if successful, the principal will lose the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

This field appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Basic Supplementary Services** and **Supplementary Services with Rerouting** fields are both set to **y**.

This field specifies whether, with both QSIG Diversion with Rerouting and QSIG VALU turned on, the Coverage After Forwarding option on the **Station** screen will work for a user for calls that go to remote coverage. Normally, with QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG server on which the forwarding destination resides. In this case, the forwarded call cannot return to coverage for the user who originally received the call, even when the remote destination is busy or does not answer.

However, you can enter **y** in this field to have QSIG VALU call coverage take precedence. In this case, if the **Coverage After Forwarding** option on the **Station** screen is enabled for a user, then QSIG Diversion with Rerouting is not be attempted.

Valid entries	Usage
y/n	Enter **y** to allow Coverage After Forwarding to work when it is activated on a user's **Station** screen and Diversion with Rerouting is also turned on.

Station Hunt Before Coverage

This field allows you to choose whether a call to a busy station performs station hunting before going to coverage.

Valid entries	Usage
y/n	Enter **y** to use Station Hunt Before Coverage.

FORWARDING

Call Forward Override

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid entries	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.
n	Directs the system to forward calls to a station even when they are from the forwarded-to party.

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

Field descriptions for page 2

Figure 542: System-Parameters Coverage-Forwarding screen

```
change system-parameters coverage-forwarding                      page 2 of x

           SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage Of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                       Ignore Network Answer Supervision? y
                      Disable call classifier for CCRON over ISDN trunks? n
                      Disable call classifier for CCRON over SIP trunks?  n
```

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

When the system redirects a call off-net at the final coverage point in a coverage path, the system can apply no further coverage treatment even if the call is unanswered. The only reason for activating answer detection on such a call is to maintain the simulated bridged appearance (SBA) on the principal's telephone that allows the principal to answer or bridge onto the call. However, when the system monitors the call through use of a call classifier port, there is an inherent cut-through delay following the detection of answer at the far end. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Valid entries	Usage
y	Directs the system to maintain a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Allows the system to drop the SBA on the principal's telephone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Coverage Of Calls Redirected Off-Net Enabled

This field allows you to enable/disable the Coverage of Calls Redirected Off-Net (CCRON) feature. This field provides a quick means of disabling this feature if the demand on the call classifier port resources degrades other services provided by Communication Manager.

To set this field to **y**, first set the **Coverage Of Calls Redirected Off-net** field on the **System Parameters Customer-Options (Optional Features)** screen to **y**. The **Coverage of Calls Redirected Off-Net** field on this screen must be **y** to administer this field.

Valid entries	Usage
y	Avaya Communication Manager monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Avaya Communication Manager does not monitor off-net coverage/ forwarded calls. No further coverage treatment is provided if such calls are unanswered.

Disable call classifier for CCRON over ISDN trunks

When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and assumes precedences over ISDN trunk signalling. This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Disable call classifier for CCRON over SIP Enablement Services (SES) trunks

This field provides a customer the means of directing Communication Manager to dispense with the call classifier on interworked calls and rely on the SIP Enablement Services (SES) trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Ignore Network Answer Supervision

This field appears only if the **Coverage of Calls Redirected Off-Net Enabled** field on this screen is **y**.

CCRON might use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that might indicate an answered state. However, some customers pay the operating company to provide network answer supervision on their trunks and desire that CCRON not discard that information. This service can be preserved by setting this field to **n**.

On the other hand, beware when you tandem a call over a tie trunk through another server node from where it redirects to the public network over non-ISDN facilities. If the trunk on the far-end node sends a timed answer supervision, that might get tandemed back to the originating node as a network answer. In such a scenario, the originating server interprets the call as answered, leading to some undesirable behavior. To avoid these calls from mistakenly be treated as answered, set this field to **y**. An unfortunate consequence is that a short cut-through delay that is inherent to call classification is introduced when the call is answered.

Valid entries	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered.
n	Treat network answer supervision as a true answer.

System Parameters CDR (Call Detail Recording)

See **CDR System Parameters**.

System Parameters Country-Options

This screen implements parameters associated with certain international (including North American) call characteristics. You cannot change this screen. See your Avaya technical support representative if you want to modify any of the values here.

The following table shows the country codes that are used in Communication Manager. The Country Code is used by various fields and screens throughout the system.

Country code table

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
1	United States, Canada, Korea, India	300v peak to peak, < 200v peak to ground; < 70 Hz; < 5s on > 1s off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off
2	Australia, New Zealand	75 +/- 20 VRMS superimposed on 48 V dc at 14.5 to 55 Hz with cadence 400ms on, 200ms off, 400ms on, 2000ms off New Zealand: Ringing voltage at the customer's premises not less than 38 V rms (25Hz) on top of 50V d.c; 20 Hz; 400ms on, 200ms off, 400ms on, 2000ms off
3	Japan	75 VRMS(75-10VRMS <= x <= 75+8VRMS), 15-20 Hz and cadence of 1second on and 2 seconds off is required
4	Italy	20 to 50 Hz, 26 to 80 Volts rms superimposed on 48 V dc, 1 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
5	Netherlands	25 Hz, 35 to 90 Volts rms superimposed on 66 V dc, 1 sec on, 4 sec off. Note that 50 Hz is recommended, and another cadence may be 0.4 sec on, 0.2 sec off, 0.4 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
6	Singapore	75V at 24Hz with a cadence of 0.4 seconds on, 0.2 seconds off, 0.4 seconds on and 2.0 seconds off.
7	Mexico	25 Hz, 70 +/- 20 Vrms superimposed on 48Vdc Cadence 1 sec on, 4 sec off, flashhook is 100 ms
		1 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
8	Belgium, Luxembourg, Korea	25 Hz, 25 to 75 Volts rms superimposed on 48 V dc, 1 sec on, 3 sec off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
9	Saudi Arabia	
10	United Kingdom	U.K.: 15 to 26.25 Hz, 25 to 100 Volts rms superimposed on 48 V dc, 0.35 on, 0.22 off then start in at any point in: 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off. Note 1: 48v DC may be present during the whole cadence or may be confined to silent periods. Note 2: Some exchanges provide a facility known as immediate ring; in this case an initial burst of ringing 20 msec to 1 sec in length immediately precedes switching to any point in the normal ringing cycle. Ireland: 25 Hz, 30 to 90 Volts rms superimposed on 50 V dc, 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off another possible cadence is 0.375 sec on, 0.250 sec off, 0.375 sec on, 2 sec off. ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
11	Spain	20 to 30 Hz, 35 to 75 Volts rms superimposed on 48 V dc, 1 to 1.5 sec on, 3 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
12	France	50 Hz, 28 to 90 Volts rms superimposed on 0.45 to 54 V dc, 1.5 sec on, 3.5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
		2 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
13	Germany	Germany: 25 Hz, 32 to 75 Volts rms superimposed on 0 to 85 V dc, 1 sec on, 4 sec off Austria: 40 to 55 Hz, 25 to 60 Volts rms superimposed on 20 to 60 V dc, 1 sec on, 5 sec off +/- 20% ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
14	Czech Republic, Slovakia	
15	Russia (CIS)	25+\-2 Hz, 95+\-5 Volts eff, local call cadence: first ring 0.3-4.5 sec then 1 second on 4 seconds Off toll automatic cadence 1 sec On 2 sec Off toll operator: manual sending
16	Argentina	25Hz; 75 Vrms superimposed on 48 Vdc; 1s on 4s off
17	Greece	
18	China	25Hz +/- 3Hz; 75 +/- 15 Vrms; Harmonic Distortion <= 10%; 1 sec ON, 4 secs OFF
19	Hong Kong	75 +/- 20 VRMS superimposed on -40 to -48 V dc at 25 Hz +/- 10% with cadence 0.4 s on, 0.2 s off, 0.4 s on, 3.0 s off
20	Thailand	
21	Macedonia	
22	Poland	
23	Brazil	25Hz +/-2.5Hz; minimum of 40 Vrms; 1s on, 4s off for equipment supporting up to six trunks only otherwise 25Hz +/-2.5Hz; minimum of 70+/-15 Vrms at a continuous emitting condition under no load, overlapping a DC level.
		3 of 4

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
24	Nordic	Finland: 25 Hz, 35 to 75 Volts rms superimposed on 44 to 58 V dc, 1 sec on, 4 sec off 25 Hz, 40 to 120 Volts rms superimposed on 44 to 56 V dc, 0,75 on, 7,5 off +/- 20 % 25 Hz, 28 to 90 Volts rms superimposed on 24 to 60 V dc, 1 sec on, 4 sec off 25 and 50 Hz, 30 to 90 Volts rms superimposed on 33 to 60 V dc, 1 sec off, 5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
25	South Africa	
		4 of 4

Field descriptions for page 1

Figure 543: System Parameters Country-Options screen

```
change system-parameters country-options                      Page 1 of x

                        SYSTEM PARAMETERS COUNTRY-OPTIONS

                       Set Layer 1 timer T1 to 30 seconds? n
                                    Display Character Set? Ukrainian
                              Directory Search Sort Order: Cyrillic
                                     Howler Tone After Busy? n
                     Disconnect on No Answer by Call Type? n
 Enable Busy Tone Disconnect for Analog Loop-start Trunks? n

TONE DETECTION PARAMETERS
        Tone Detection Mode: 5       dial tone validation timer (sec): 60
          Interdigit Pause: short
```

Dial Tone Validation Timer (sec)

This field appears only when **Tone Detection Mode** is equal to **4** or **5**. (Valid with TN420C or later Tone Detector circuit pack.).

Valid entries	Usage
0 to **6375** in increments of 25	Displays number of milliseconds that the dial tone validation routine will use to sample transmissions.

Directory Search Sort Order

This field appears only when **Display Character Set** on the **System Parameters Country-Options** screen is set to **Cyrillic** or **Ukrainian**.

Tip:

You can toggle to the unadministered value for a single search session by first pressing the pound key (#) on your telephone dial pad. Subsequent sessions return to the administered value.

Valid entries	Usage
Cyrillic	Cyrillic Collation is used for integrated directory name search and result sorting. This is the default value.
Roman	Eurofont Latin Collation is used for directory name search and result sorting. The letters to be searched in the specified order for dial pad button presses are defined in the row for each key.

Disconnect on No Answer by Call Type

Drops outgoing trunk calls (except DCS and AAR) that users leave unanswered too long.

Valid entries	Usage
y/n	Enables the system to disconnect calls that are not answered.

Display Character Set

The value in this field determines the character set used for all name values that do not have an ASCII-only restriction.

Note that the setting for this field can affect the Display Language field on the **Station** screen for Unicode-capable stations. Specifically, if **Display Character Set** is set to **katakana**, the **Display Language** field for 4624 sets will allow an entry for **Unicode**, which is required for

Toshiba SIP Telephone (TSP) sets. If **Display Character Set** is not set to **katakana**, then Toshiba SIP telephones will not operate correctly in Japan.

WARNING:
Changing the value in this field might cause some telephones to perform improperly, and will cause non-ASCII data in non-native names to display incorrectly on telephones. To correct this, you must remove the non-native names of previously administered station(s) and re-administer them, together with any display messages that have been administered, to use non-ASCII characters.

Valid entries	Usage
Cyrillic **Katakana** **Roman** **Ukrainian**	Indicate the enhanced character set to display. See "Telephone Display" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information. **Note: Cyrillic**, **Roman**, and **Ukrainian** map to the Eurofont character set. For **Katakana**, the Optrex font is used. If a Communication Manager server uses non-English in any name field, characters on a BRI station are not displayed correctly.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

This field allows Communication Manager to recognize a busy tone from the central office as a disconnect signal.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to disconnect the trunk when a busy tone is received from the central office.

Howler After Busy

Plays howler tone when users leave their analog telephone off-hook too long.

Valid entries	Usage
y/n	Enables howler tone.

Set Layer 1 timer T1 to 30 seconds

Valid entries	Usage
y/n	Specifies whether the Layer 1 timer is set to 30 seconds.

TONE DETECTION PARAMETERS

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored. (Valid with TN420C or later Tone Detector circuit pack.)

Valid entries	Usage
short	5 to 30ms
long	20 to 40ms

Tone Detection Mode

The country code specifies the type of tone detection used on a TN420B (or later) tone-detection circuit pack.

Valid entries	Usage
1	Precise Italian tone detection algorithm
2	Precise Australian tone detection algorithm
3	Precise UK tone detection algorithm
4	Imprecise normal broadband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
5	Imprecise wideband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
6	

System Parameters Customer-Options (Optional Features)

This screen shows you which optional features are enabled for your system, as determined by the installed license file. The fields on this screen are populated by the license file, and are display only. If you have any questions about disabling or enabling one of these features, contact your Avaya representative.

Field descriptions for page 1

Figure 544: System Parameters Customer-Options (Optional Features) screen

```
display system-parameters customer-options                      Page   1 of  x
                                OPTIONAL FEATURES

G3 Version: V12  123456789012                 Software Package: Standard
  Location: 2                              RFA System ID (SID):
  Platform: 2                              RFA Module ID (MID): 123456

                                                             USED
                                Platform Maximum Ports: 300   174
                                      Maximum Stations: 300   174
                              Maximum XMOBILE Stations: 30    28
                    Maximum Off-PBX Telephones - EC500: 1200  0
                    Maximum Off-PBX Telephones -   OPS: 1200  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0



        (NOTE: You must logoff & login to effect the permission changes.)
```

G3 Version

Identifies the version of Avaya Communication Manager being used.

Location

Indicates the location of this Avaya S8XXX Server. 1 indicates Canada or the United States. 2 indicates any other location, and allows the use of International Consolidation circuit packs and telephones.

Maximum Off-PBX Telephones - EC500

Valid entries	Usage
0 to license max	Stations that are administered for any Extension to Cellular (EC500/ CSP) application count against this limit. Default is **0**. This field is administrable when H.323, IDSN-PRI, ISDN-BRI Trunks or Multifrequency Signaling is enabled on the **System Parameters Customer-Options (Optional Features)** screen (`display system-parameters customer-options`). The “license max” value is defined as follows: ● On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - OPS

Valid entries	Usage
0 to license max	Stations that are administered for any SIP Extension to Cellular/OPS application count against this limit. Default is **0**. This field is administrable when SIP is enabled on the **System Parameters Customer-Options (Optional Features)** screen (`display system-parameters customer-options`). The “license max” upper limit is: ● On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - PBFMC

Valid entries	Usage
0 to license max	Number of stations administered for Public Fixed-Mobile Convergence. Each station is allowed only one PBFMC application. Default is **0**. This field is administrable when H.323, IDSN-PRI, ISDN-BRI Trunks or Multifrequency Signaling is enabled on the **System Parameters Customer-Options (Optional Features)** screen (**`display system-parameters customer-options`**). The “license max” value is defined as follows: ● On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - PVFMC

Valid entries	Usage
0 to license max	Number of stations administered for Private Fixed-Mobile Convergence. Each station is allowed only one PVFMC application. Default is **0**. This field is administrable when SIP Trunks are enabled on the **System Parameters Customer-Options (Optional Features)** screen (**`display system-parameters customer-options`**). The “license max” upper limit is: ● On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP Enablement Services (SES) trunks. ● On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - SCCAN

Valid entries	Usage
0 to license max	The “license max” value is defined as follows: ● SCCAN is only available on Linux systems. The upper limit is the maximum number of administrable stations. ● Stations that are administered for any Extension to Cellular/OPS application count against this limit. Default is **0**.

Maximum Stations

Displays the maximum number of stations allowed in the system. This feature is set based on the Communication Manager license file. Default is **0**.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

A display-only field that identifies, via a number mapping, the platform being used for a specific customer. Valid values and server types are:

Platform number	Server type
6	S87XX Server
8	S87XX Server
12	S8500 Server
14	S87XX ESS Server
15	S8500 ESS server

Platform Maximum Ports

Number of ports active, per contract.

Software Package

Indicates whether the software package license is **Standard** or **Enterprise**.

Used

Shows the actual current usage as compared to the system maximum for each field.

Field descriptions for page 2

Figure 545: System Parameters Customer-Options (Optional Features) screen

```
display system-parameters customer-options                          page 2 of x

                              OPTIONAL FEATURES

IP PORT CAPACITIES                                                 USED
                       Maximum Administered H.323 Trunks:
                          Maximum Administered IP Trunks: 100   96
             Maximum Concurrently Registered IP Stations: 10    10
               Maximum Administered Remote Office Trunks: 0     0
  Maximum Concurrently Registered Remote Office Stations: 0     0
                Maximum Concurrently Registered IP eCons: 0     0
                          Maximum Video Capable Stations: 0     0
                     Maximum Video Capable IP Softphones: 0     0
                         Maximum Administered SIP Trunks: 500   25
    Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
     Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                               Maximum TN2501 VAL Boards: 1     0
                      Maximum G250/G350/G700 VAL Sources: 0     0
             Maximum TN2602 Boards with 80 VoIP Channels: 20    12
            Maximum TN2602 Boards with 320 VoIP Channels: 4     3

     Maximum Number of Expanded Meet-me Conference Ports: 0      0

           (NOTE: You must logoff & login to effect the permission changes.)
```

IP PORT CAPACITIES

Maximum Administered Ad-hoc Video Conferencing Ports

Defines the number of ad-hoc ports allowed for the system; one for each simultaneous active conference port. The maximum number of ad-hoc video conferencing ports allowed is the sum of the maximum allowed IP trunks and the maximum allowed SIP trunks on your system. Default is **O**. The **IP Trunks** field on the **System Parameters Customer Options** screen must also be set to **y**.

Maximum Administered IP Trunks

Defines limits of the number of IP trunks administered.

Maximum Administered Remote Office Trunks

Defines limits of the number of IP endpoints based on the endpoint. Use the smaller of this number or the number based on the **MAXIMUM IP REGISTRATIONS BY PRODUCT ID** page of this screen.

Maximum Administered SIP Trunks

Defines limits on the number of SIP Enablement Services (SES) trunks administered.

Maximum Concurrently Registered IP eCons

Specifies the maximum number of IP SoftConsoles that can be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that can be registered at one time. This field accepts 6,000 concurrently registered IP stations for the S87XX series servers, and 3,000 for S8500 servers.

Maximum G250/G350/G700 VAL Sources

Specifies the maximum number of VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Shows the number of DS1 circuit packs that can have echo cancellation.

Maximum Number of Expanded Meet-me Conference Ports

Displays the license-file based value of the system maximum for the number of Expanded Meet-me Conference ports.

Maximum TN2501 VAL Boards

This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Valid entries	Usage
0 to **10** (S87XX Servers) **0** to **5** (DEFINITY CSI, and S8300 Servers)	• For values greater than 1, the **Val Full 1-Hour Capacity** field on page 4 of the **System Parameters Customer-Options (Optional Features)** screen must be set to **y**. • This field updates the **System Limit** field on the **System Capacity** report.

Maximum TN2602 Boards with 80 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 80 VoIP channels. The value is based on the value in the Avaya Communication Manager license file.The **USED** value is the total number of TN2602AP boards in the system administered with 80 VoIP channels. Default is **0**.

Maximum TN2602 Boards with 320 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 320 VoIP channels. The value is based on the value in the Avaya Communication Manager license file. The **USED** value is the total number of TN2602AP boards in the system administered with 320 VoIP channels. Default is **0**.

Maximum Video Capable Stations

Specifies the maximum number of stations that are video-capable. The maximum number depends on the type of system.

Maximum Video Capable IP Softphones

Specifies the maximum number of IP Softphones that are video-capable. The maximum number depends on the type of system.

Used

For each item with a capacity listed, the **USED** value is the actual number of units currently in use.

Field descriptions for page 3

Figure 546: System Parameters Customer-Options (Optional Features) screen

```
display system-parameters customer-options                         page 3 of x

                         OPTIONAL FEATURES

     Abbreviated Dialing Enhanced List?             Audible Message Waiting?
         Access Security Gateway (ASG)?                 Authorization Codes?
         Analog Trunk Incoming Call ID?                          CAS Branch?
   A/D Grp/Sys List Dialing Start at 01?                           CAS Main?
  Answer Supervision by Call Classifier?                  Change COR by FAC?
                                   ARS?   Computer Telephony Adjunct Links?
                   ARS/AAR Partitioning?    Cvg Of Calls Redirected Off-net?
            ARS/AAR Dialing without FAC?                       DCS (Basic)?
            ASAI Link Core Capabilities?                  DCS Call Coverage?
            ASAI Link Plus Capabilities?                 DCS with Rerouting?
         Async. Transfer Mode (ATM) PNC?
Async. Transfer Mode (ATM) Trunking?        Digital Loss Plan Modification?
                ATM WAN Spare Processor?                           DS1 MSP?
                                 ATMS?              DS1 Echo Cancellation?
                  Attendant Vectoring?
```

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. You dial an abbreviated code or depress an assigned button. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Allows you to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1. This allows Abbreviated Dialing under Avaya Communication Manager to operate like it did with the DEFINITY G2 system.

Analog Trunk Incoming Call ID

This field allows collection and display the name and number of an incoming call information on analog trunks.

Answer Supervision by Call Classifier

This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This field is set to **y** if the system contains a call-classifier circuit pack.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into 8 user groups within a single server running Avaya Communication Manager. Can establish individual routing treatment for each group.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a feature access code (FAC).

CAUTION:
Contact your Avaya technical support representative to discuss details of this feature before enabling it.

ASAI Link Core Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

Note:

ASAI Link Core Capabilities only applies to links administered as type **asai**. This field was previously named **ASAI Interface**.

If the **ASAI Link Core Capabilities** field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Control
- Domain Control
- Event Notification
- Single Step Conference
- Request Feature
- II Digits
- Set Value
- Value Query

For more information, see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

ASAI Link Plus Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. If the **ASAI Link Plus Capabilities** field is administered to **y**, then the following ASAI capability groups are enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Note:

ASAI Link Plus Capabilities only applies to links administered as type **asai**.

For more information, see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Asynch. Transfer Mode (ATM) PNC

ATM PNC can be enabled only if:

- all prior fiber-link administration has been removed

- all "switch-node" and "dup-switch-node" carrier types have been removed

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Can only be enabled if the **ISDN-PRI** field is set to **y**. Enables circuit emulation service (CES).

ATM WAN Spare Processor

An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional. Cannot be set to **y** if the **Async. Transfer Mode (ATM) Trunking** field is **n**.

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

Attendant Vectoring

Allows you to use attendant vectoring. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**.

Audible Message Waiting

Provides audible message waiting.

Authorization Codes

Permits you to selectively specify levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Provides Centralized Attendant Service - Branch. See **CAS Main** for more information. Cannot be set to **y** if the **Attendant Vectoring** is **y** and **Centralized Attendant** on the QSIG OPTIONAL FEATURES page of the **System Parameters Customer Options** screen is **y**.

CAS Main

Provides multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Avaya Communication Manager location. The main Avaya Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call. Cannot be set to **y** if the **Centralized Attendant** and **CAS Branch** fields are **y**.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a telephone by using a feature access code (FAC). Cannot be set to **y** if the **Tenant Partitioning** field is **y**.

Computer Telephony Adjunct Links

Provides linkage between Avaya Communication Manager and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

Note:

Computer Telephony Adjunct Links only applies to links administered as type **adjlk**. This field was previously named **ASAI Proprietary Adjunct Links**.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of servers.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities. Cannot be set to **y** if the **ISDN PRI** field is **n**.

Digital Loss Plan Modification

Allows you to customize the digital loss and digital tone plans.

DS1 MSP

Provides the ability to change fields on **DS1 Circuit Pack** screen without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

Field descriptions for page 4

Figure 547: System Parameters Customer-Options (Optional Features) screen

```
display system-parameters customer-options                      Page   4 of  x
                                OPTIONAL FEATURES

  Emergency Access to Attendant? y                             IP Stations? y
          Enable 'dadmin' Login? y
          Enhanced Conferencing? y                       ISDN Feature Plus? y
                 Enhanced EC500? y      ISDN/SIP Network Call Redirection? y
   Enterprise Survivable Server? y
      Enterprise Wide Licensing? y                         ISDN-BRI Trunks? y
             ESS Administration? y                                ISDN-PRI? y
         Extended Cvg/Fwd Admin? y              Local Survivable Processor? y
    External Device Alarm Admin? y                     Malicious Call Trace? y
         Extended Cvg/Fwd Admin? y   Mode Code for Centralized Voice Mail? y
    External Device Alarm Admin? y
 Five Port Networks Max per MCC? y                 Multifrequency Signaling? y
               Flexible Billing? y Multimedia Appl. Server Interface(MASI)? y
  Forced Entry of Account Codes? y       Multimedia Call Handling (Basic)? y
     Global Call Classification? y    Multimedia Call Handling (Enhanced)? y
           Hospitality (Basic)? y              Multimedia IP SIP Trunking? y
Hospitality (G3V3 Enhancements)? y
                     IP Trunks? y

          IP Attendant Consoles? y

           (NOTE: You must logoff & login to effect the permission changes.)
```

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya Communication Manager or dialed by users.

Enable 'dadmin' Login

Provides business partners the ability to install, administer, and maintain S87XX Servers and DEFINITY switches. The dadmin login has access to all the same commands as other logins with the exception of `Go` and **`WP`**. `Go` is used for **`go tcm`** and **`go debug`** as well as **`go server`**. WP is for writing memory.

Enhanced Conferencing

Enhanced Conferencing allows the customer to use the Meet-me Conference, Expanded Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and the No Hold Conference features. Must be **y** to enable the Enhanced Conferencing features.

Enhanced EC500

Enables Extension to Cellular for administration. “EC500” refers to the Extension to Cellular feature. When this field is set to **y**, all screens under the **`off-pbx-telephone`** commands are available.

Enterprise Survivable Server

This display-only field is activated through the license file. When this field is set to **y**, this server is an Enterprise Survivable Server (ESS). For more information about ESS, see *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

ESS Administration

This display-only field enables administration of Enterprise Survivable Servers (ESS) on the **System Parameters - ESS** screen. For more information about ESS, see *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. See your Avaya representative for more information.

Five Port Networks Max Per MCC

Available only for S87XX Series Fiber-PNC. Allows system administrator to create five port networks in a multi-carrier cabinet. If there are any cabinets with more than two PNs assigned, this field cannot be set to **n**.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, etc.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode). Cannot be set to **y** if the **Hospitality (Basic)** field is **n**.

Note:
Standard hospitality features are included in basic system software.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be **y** for IP telephones.

IP Trunks

Controls permission to administer H.323 trunks. Must be **y** for IP trunks.

ISDN Feature Plus

Provides ISDN Feature Plus signaling. This option is enabled when either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field is **y**.

ISDN/SIP Network Call Redirection

Administrable if the **ISDN-PRI** or **ISDN-BRI Trunk** field is **y**. Network Call Redirection (NCR) redirects an incoming ISDN call from a server running Avaya Communication Manager to another PSTN endpoint. It is used in Call Centers with Best Service Routing and Lookahead Interflow.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager. If enabled, can add isdn trunk groups and the following screens are accessible:

- **network-facilities**
- **private-numbering**
- **public-unknown- numbering**

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be **y** for IP and SIP trunks.

Local Survivable Processor

This display-only field indicates that the server is a Local Survivable Processor (LSP). When this field is set to **y**, the LSP server is configured to provide standby call processing in case the primary server is unavailable.

Malicious Call Trace

Provides ability to retrieve certain information related to a malicious call.

Mode Code for Centralized Voice Mail

This feature provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for a screen of number signaling used between Communication Manager and the central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Multimedia IP SIP Trunking

If enabled, extends applicability of the H.323 video station licensing/control to all non-ip-softphones.

Field descriptions for page 5

Figure 548: System Parameters Customer-Options (Optional Features) screen

```
display system-parameters customer-options                          page 5 of x

                                OPTIONAL FEATURES

                 Multinational Locations?           Station and Trunk MSP? n
Multiple Level Precedence and Preemption?      Station as Virtual Extension? n
                      Multiple Locations?
                                               System Management Data Transfer? n

         Personal Station Access (PSA)? y
                       Posted Messages? n             Tenant Partitioning? n
                       PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                  Port Network Support? y             Time of Day Routing? y
             Processor and System MSP? n             Uniform Dialing Plan? y
                   Private Networking? y     Usage Allocation Enhancements? y
                   Processor Ethernet? y       TN2501 VAL Maximum Capacity? y

                        Remote Office? n              Wideband Switching? y
         Restrict Call Forward Off Net? y                       Wireless? n
                 Secondary Data Module? y
```

Multinational Locations

The Multinational Locations feature provides you with the ability to use a single Enterprise Communication Server (ECS) with stations, port networks, remote offices, or gateways in multiple countries. With this feature enabled, you can administer location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Allows you to establish numbering plans and time zone and daylight savings plans that are specific for each cabinet in a port network.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Posted Messages

Supports users being able to post messages, which they select from among a set of as many as 30 (15 fixed, 15 administrable), to be shown on display telephones.

PNC Duplication

If set to **y**, the **Enable Operation of PNC (Port Network Connectivity) Duplication** field appears on the **System Parameters Duplication** screen. The **Enable Operation of PNC Duplication** field is set with the **Enable Operation of SPE (Switch Processing Element) Duplication** field to provide non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Indicates that the server is operating as a stand-alone Internal Communications Controller (ICC) when set to **n** and is used to disable traditional port networking. Set to **y** to indicate that traditional Avaya DEFINITY port networks are in use.

Private Networking

Upgrades PNA or ETN software RTU purchased with earlier systems.

Processor Ethernet

Appears only on S8300, S8400, and S8500 Servers. Used to enable use of the Ethernet card resident in the processor cabinet for use by the DEFINITY Call Processing software in place of a Control Lan (C-LAN) card (located in a port network). The Processor Ethernet interface is always enabled for S87XX Servers. For more information, see Setting up Processor Ethernet on page 593.

Processor and System MSP

Allows the customer administrator or technician to maintain processor and system circuit packs.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

The system can monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides ability to use any data module as a secondary data module.

Station and Trunk MSP

Provides the customer administrator or technician to maintain station and trunk circuit packs.

Station as Virtual Extension

Allows **virtual** to be entered in the **Type** field of the **Station** screen, which allows multiple virtual extensions to be mapped to a single physical analog telephone. The user can also administer a specific ringing pattern for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

System Management Data Transfer

Indicates Communication Manager is accessible by Network Administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with **X** in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port. Must be set to **y** for Automatic Customer Telephone Rearrangement.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week. You can take advantage of lower calling rates during specific times.

TN2501 VAL Maximum Capacity

If this is enabled, you have the Enhanced offer, which allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs.

Uniform Dialing Plan

Provides 3- to 7-digit Uniform Dial Plan (UDP) and 1 to 7 digit steering. Also allows you to use Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Provides for assigning ISDN-PRI or ISDN-BRI Services/Features for Usage Allocation Plans. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Wideband Switching

Provides wideband data software for switching video or high-speed data. You can aggregate DSO channels up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for wireless applications in certain Network Systems sales. You can purchase it from Avaya Network Wireless Systems.

Field descriptions for Call Center Optional Features

Figure 549: Call Center Optional Features screen

```
  display system-parameters customer-options                        page 6 of x

                        CALL CENTER OPTIONAL FEATURES

                          Call Center Release:

                               ACD? y    PASTE (Display PBX Data on Phone)? n
                      BCMS (Basic)? y                          Reason Codes? n
                                                     Service Level Maximizer? y
         BCMS/VuStats Service Level? n            Service Observing (Basic)? y
                 Business Advocate? n    Service Observing (Remote/By FAC)? n
                   Call Work Codes? y             Service Observing (VDNs)?
      DTMF Feedback Signals For VRU? y                            Timed ACW?
                  Dynamic Advocate? n                     Vectoring (Basic)? y
       Expert Agent Selection (EAS)? y                Vectoring (Prompting)? y
                           EAS-PHD? n            Vectoring (G3V4 Enhanced)?
                  Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? n
              Least Occupied Agent?      Vectoring (G3V4 Advanced Routing)? n
         Lookahead Interflow (LAI)?                       Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n    Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                 Vectoring (Holidays)?
```

ACD

Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills. Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Cannot be set to **n** if the **Call Work Codes** field is **y**.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Allows you to set up hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be considered acceptable.

Business Advocate

Software that provides an integrated set of advanced features to optimize call center performance. If set to **n**, the **Least Occupied Agent** field displays. For information on Business Advocate, contact your Account Executive.

Call Center Release

Displays the call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers. Cannot be set to **y** if the **ACD** field is **n**.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to VRUs.

Dynamic Advocate

Software that provides an integrated set of advanced features to optimize call center performance.

EAS-PHD

Increases the number of skills an agent can log in to from four to 20. Increases the number of agent skill preference levels from two to 16.

Expert Agent Selection (EAS)

Provides skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See **Multiple Call Handling**.

Least Occupied Agent

Appears only if the **Business Advocate** field is **n**. Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations. Cannot be set to **y** if the **Vectoring (Basic)** field is **n**.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced. Cannot be set to **y** if the **ACD** field is **n** and the **Forced ACD Calls** field is **y**.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met. SLM is used with Expert Agent Selection (EAS), and without Business Advocate. Call Center Release must be 12 or later.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature’s access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's calls.

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (Basic)

Provides basic call vectoring capability.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing.

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (CINFO)

Provides the ability to collect ced and cdpd from the network for vector routing. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Vectoring (Best Service Routing)

Enables the Best Service Routing feature. Through special vector commands, Best Service Routing allows you to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (Holidays)

Enables the Holiday Vectoring feature. It simplifies vector writing for holidays.

Field descriptions for Call Center Optional Features

Figure 550: Call Center Optional Features screen

```
display system-parameters customer-options                     Page 7 of x
                       CALL CENTER OPTIONAL FEATURES

     VDN of Origin Announcement? n                               VuStats? n
         VDN Return Destination? n             VuStats (G3V4 Enhanced)? n








                                     Used
            Logged-In ACD Agents: 500
       Logged-In Advocate Agents: 500
   Logged-In IP Softphone Agents: 500
```

Logged-In ACD Agents

Number of ACD Agents contracted for. This field limits the number of logged-in ACD agents to a number no more than the maximum purchased. The value of this field indicates the total of ACD agents that can be logged-in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

Appears when the **Business Advocate** field is **y**. Number of Business Advocate Agents contracted for.

The total number of logged-in Business Advocate agents must be equal to or less than the number allowed in the **Logged-In ACD Agents** field. The number of logged-in Business Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

Number of IP Softphone Agents contracted for. This field limits the number of logged-in IP Softphone agents to a number no more than the maximum purchased. The value of this field indicates the total of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Allows you to present BCMS statistics on telephone displays.

VuStats (G3V4 Enhanced)

Allows you to use the G3V4 VuStats enhancements including historical data and thresholds.

Field descriptions for QSIG Optional Features

Figure 551: QSIG Optional Features screen

```
display system-parameters customer-options                      Page 8 of x
                            QSIG OPTIONAL FEATURES

                                         Basic Call Setup? n
                             Basic Supplementary Services? n
                                    Centralized Attendant? n
                                    Interworking with DCS? n
                    Supplementary Services with Rerouting? n
                            Transfer into QSIG Voice Mail? n
                                       Value-Added (VALU)? n
```

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**.

Basic Supplementary Services

To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**. Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Centralized Attendant

Can be enabled only if the **Supplementary Services with Rerouting** field is **y**. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**. Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines. If this field is **y**, the **IAS** fields on the **Console Parameters** screen do not display.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled server in a network and a QSIG-enabled server:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

This field cannot be set to **y** if the **DCS (Basic)** field is **n**.

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Can be enabled only if the **Basic Supplementary Services** field is **y** and either the **ISDN-PRI Trunk** or **ISDN-BRI Trunk** field is **y**. Allows transfer directly into the voice-mail box on the voice-mail system when a QSIG link connects Avaya Communication Manager and the voice-mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting. See *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504, for more information.

Field descriptions for ASAI

Figure 552: ASAI Features screen when the ASAI Link Plus Capabilities field is y

```
change system-parameters customer options                      Page 8 of X

                                  ASAI FEATURES

                                 CTI Stations? n
                                Phantom Calls? n



                            ASAI PROPRIETARY FEATURES

                                 Agent States? n
```

Agent States

Appears when the **Computer Telephony Adjunct Links** field is **y**. The **Agent States** field provides proprietary information used by Avaya applications. For more information, contact your Avaya technical support representative.

Note:

The **Agent States** field only applies to links administered as type **adjlk**. This field was previously named **Proprietary Applications**.

CTI Stations

Appears when the **ASAI Link Plus Capabilities** field is **y**. This field needs to be enabled for any application (using a link of Type ASAI) that uses a CTI station to receive calls.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Phantom Calls

Appears when the **ASAI Link Plus Capabilities** field is **y**. Enables phantom calls. The **Phantom Calls** field only applies to links administered as type ASAI.

For more information see the *Avaya Communication Manager ASAI Technical Reference*, 555-230-220.

Field descriptions for Maximum IP Registrations by Product ID

Figure 553: Maximum IP Registrations by Product ID screen

```
                                                               Page 9 of x
                            MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit     Product ID_Rel. Limit     Product ID_Rel. Limit
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
```

Limit

Maximum number of IP registrations allowed.

Valid entries	Usage
1000 or **5000**, depending on your server configuration	Maximum number of IP registrations allowed. For Avaya R300 Remote Office Communicator, defaults to the maximum allowed value for the **Concurrently Registered Remote Office Stations** on page 1 of this screen.

Product ID

Identifies the product using the IP (internet protocol) registration.

Valid entries*	Usage
Avaya_IR	Interactive Response product
IP_Agent	IP Agents
IP_eCons	SoftConsole IP attendant
IP_Phone	IP Telephones
IP_ROMax	R300 Remote Office telephones
IP_Soft	IP Softphones
*These are just a few examples of valid Product IDs. The valid Product IDs for your system are controlled by the license file.	

Rel

Release number of the IP endpoint.

Valid entries	Usage
0 to **99** or blank	Release number of the IP endpoint

System Parameters - Duplication

Use the **System Parameters Duplication** screen to enable PNC or IPSI duplication.

Field descriptions for page 1

Figure 554: System Parameters - Duplication screen

```
change system-parameters duplication                         Page 1 of x

                DUPLICATION RELATED SYSTEM PARAMETERS

 Enable Operation of PNC Duplication? y
Enable Operation of IPSI Duplication? y
```

Enable Operation of IPSI Duplication

Use this field to enable IPSI duplication.

Note:

This field is set to **n(o)** when either the TN8412AP or TN2312BP circuit pack is used in an S8400 configuration. This is because TN8412/TN2312 duplication is not supported in Phase 1 of the S8400. Duplication may be offered in the future.

Valid entries	Usage
y/n	Enter **y** to enable IPSI duplication.

Enable Operation of PNC Duplication

Valid entries	Usage
y/n	Enter **y** to enable PNC duplication. Appears when **PNC Duplication** is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

System Parameters - ESS

Use the **System Parameters ESS** screen to administer Enterprise Survivable Servers. This screen consists of 7 pages. On pages 1 through 5 you can administer up to 63 ESS clusters, page 6 is for administering Port Network Communities, and page 7 is for administering the no service timer and scheduling the Auto Return feature.

Note:

Beginning with Communication Manager 3.1, node names are used in place of IP addresses for ESS servers on pages 1 through 5 of the **System Parameters ESS** screen.

For more information about ESS, see *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428. For more information on Processor Ethernet, see Setting up Processor Ethernet on page 593.

Field descriptions for page 1 through 5

Figure 555: System Parameters - ESS screen

```
change system-parameters ess                                  Page 1 of x

               ENTERPRISE SURVIVABLE SERVER INFORMATION
Cl   Plat    Server A             Server B            Pri Com  Sys  Loc  Loc
ID   Type    ID     Node Name     ID     Node Name    Scr      Prf  Prf  Only
__________________________________________________________________________

                            MAIN SERVERS
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___

                   ENTERPRISE SURVIVABLE SERVERS
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
___ _______ ___ _______________ ___ _______________ ___ ___ ___  ___  ___
```

CLID

Valid entries	Usage
1 to **999** or blank	Enter the Cluster ID (the Module ID from the Communication Manager license file) for the ESS server. The Module ID appears in the license file. **Note:** If you do not know the Module ID for the ESS server, use the **`statuslicense -v`** command. The Module ID displays in the **RFA Module ID** field.

Plat Type

Valid entries	Usage
simplex **duplex**	Enter the platform type. Enter **simplex** if this server is an S8400 or S8500 Server. Enter **duplex** if this is a S87XX Server. If **simplex** is entered, you cannot enter data for the B-side server.

Server A ID

Valid entries	Usage
1 to **256** or blank	Enter the Server ID (SVID) of the S8400 or S8500 Server, or the SVID of the A-side of the S87XX Server. If this field is blank, all other entries on the line display default values.

Server A Node Name

Valid entries	Usage
valid node name administered on the **IP Node Names** screen	Enter the node name for the S8400 or S8500 Server or the A-side of the S87XX Server.

Server B ID

Valid entries	Usage
1 to **256** or blank	S87XX Servers only. Enter the Server ID (SVID) of the B-side of the S87XX Server.

Server B Node Name

Valid entries	Usage
valid node name administered on the **IP Node Names** screen	S87XX Servers only. Enter the node name for the B-side S87XX Server.

Pri Scr

A priority value is used to distinguish between ESS servers with the same preference settings, ESS servers with no preference setting, or ESS servers that are not in the same community as the IPSI. The Priority Score can be set from one to 100 points. The default value is **1**. The value of the priority score and the selected preference, combine to determine the position of the ESS server on the IPSI's Priority Score List. The IPSI will failover to the ESS server at the top of the list with the highest priority score.

Valid entries	Usage
1 to **100**	Enter the Priority Score for this ESS server. Default is **1**.

Com

A community is a virtual group consisting of an ESS server and one or more Port Networks. Assigning an ESS server to a community associates the ESS server with the IPSI(s) in the Port Network(s) for that community. The Port Networks are assigned to Communities on page 6 of the **System Parameters ESS** screen. The association effects how the ESS server is prioritized for the IPSI in that community, if the ESS server is administered with a **Local Preferred** or **Local Only** preference.

Note:

It is possible to administer an ESS server as having no preferences and just a priority score. If all ESS servers where administered in this fashion, the IPSI would prioritize each ESS server based on its priority score only.

Valid entries	Usage
1 to **64**	Enter the Network community number in this field. This field cannot be blank.

Sys Prf

System Preferred servers will have a higher value than any other **Local Preferred** server independent of community or administered priority value. If multiple **System Preferred** servers are administered, the server with the highest administered priority value will have the top priority on an IPSI's list.

Valid entries	Usage
y/n	Use this option when the goal is to keep as much of the system network intact as possible, allowing one ESS server to replace the Main server. If this field is set to **y**, then **Local Preferred** and **Local Only** default to **n** and cannot be changed. If this field is **n**, then **Local Preferred** and **Local Only** can be either **y** or **n**. Default is **y**.

Loc Prf

After the **System Preferred** preference, the **Local Preferred** preference has the second highest value within an IPSI community. When multiple **Local Preferred** servers are administered within the same community, the priority score is used to determine which server will have the higher priority in the IPSI's list.

Valid entries	Usage
y/n	Use this option when you want the ESS server to accept the request for service from IPSIs co-located in the same geographical region, WAN/LAN segment, district, or business unit. Default is **n**.

Loc Only

A Local Only server only advertises to the IPSI within its community. The **Local Only** preference has no value. If a **Local Preferred** server (outside its administered community), advertised to an IPSI in the same community as a **Local Only** server, the priority score of each server would determine its ranking on the IPSI's priority list. Servers set to **Local Only** do have the option of also using the **Local Preferred** setting to increase its priority on a local IPSI's list.

Valid entries	Usage
y/n	Use this option when you want the ESS server to accept the request for service from an IPSI, only if the IPSIs is located in the ESS server's same community. Default is **n**.

Field descriptions for page 6

Use this page to enter the community assignments for each Port Network. Assigning Port Networks to a community associates the Port Network with an ESS server administered with the Local Preferred or Local Only preference. To have the ESS server and the Port Networks in the same community, the community number of the ESS server and the community number for each Port Network must match.

Figure 556: System Parameters - ESS screen - page 6

```
change system-parameters ess                                  Page 6 of x

            COMMUNITY ASSIGNMENTS FOR PORT NETWORKS
PN Community PN Community PN Community PN Community PN Community
------------- ------------- ------------- ------------- -------------
 1: 1         14: 1      27: 1        40: 1        53: 1
 2: 1         15: 1      28: 1        41: 1        54: 1
 3: 1         16: 1      29: 1        42: 1        55: 1
 4: 1         17: 1      30: 1        43: 1        56: 1
 5: 1         18: 1      31: 1        44: 1        57: 1
 6: 1         19: 1      32: 1        45: 1        58: 1
 7: 1         20: 1      33: 1        46: 1        59: 1
 8: 1         21: 1      34: 1        47: 1        60: 1
 9: 1         22: 1      35: 1        48: 1        61: 1
10: 1         23: 1      36: 1        49: 1        62: 1
11: 1         24: 1      37: 1        50: 1        63: 1
12: 1         25: 1      38: 1        51: 1        64: 1
13: 1         26: 1      39: 1        52: 1

Enter number between 1 and 64
```

PN

Valid entries	Usage
1 to **64**	Displays the port network.

Community

Valid entries	Usage
1 to **64**	Enter the Network Community number you want to associate with this port network. **Note:** If the port network is administered in the system, the default community is **1** and administrable with a value between **1** and **64**. If the port network is not administered in the system, the community value is **1** and not administrable.

Field descriptions for page 7

Use this page to schedule the Auto Return feature and to set the no service timer.

Figure 557: System Parameters - ESS screen - page 7

```
change system-parameters ess                                    Page 7 of x

             ENTERPRISE SURVIVABLE SERVER OPTIONAL FEATURES

FAILOVER PARAMETERS                          FALLBACK PARAMETERS
No Service Time Out Interval: 5____                    Auto Return: scheduled
                                                               Day: Thursday
                                                              Time: 23:30

                                           IPSI Connection up time:
```

Auto Return

The Auto Return functionality is used to schedule a day and time for all Port Networks to return to the control of the Main server after a failover occurs.The schedule can be set up to seven days prior the its activation.

Valid entries	Usage
y(es)	When the value is set to **y(es)**, the **IPSI Connection up time** field appears. When **Auto Return** is set to **y(es)**, the port networks can automatically return to the main server after the value set in the **IPSI Connection up time** expires.
n(o)	Auto Return is disabled. When the value is set to **n(o)**, the port networks cannot automatically return to the control of the main server. No additional fields appear when the value is set to **n(o)**.
s(cheduled)	Auto Return is enabled. When set to **s**, the **Day** and **Time** fields appear. Schedule a day and time to return the port networks to the control of the main server. The schedule can be set up to seven days prior to its activation. • **Day**: Enter the day of the week • **Time**: Enter the time of day in a 24 hour (military) format

Day

Valid entries	Usage
Monday through **Sunday**	Enter the day of the week.

IPSI Connection up time

Valid entries	Usage
3 to **120** minutes	Enter the number of minutes that the IPSI will wait to return to the main server after communication with the main server is restored.

No Service Time Out Interval

Valid entries	Usage
3 to **15**	Enter the time, in minutes, that the IPSIs will wait before requesting service from the highest ESS server on its priority list. Default is **5** minutes.

Time

Valid entries	Usage
00:00 to **23:59**	Enter the time of day in a 24 hour (military) format.

System Parameters - Features

See **Feature-Related System Parameters**.

System Parameters - IP Options

See **IP-Options System Parameters**.

System Parameters - Maintenance

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

System Parameters Media Gateway Automatic Recovery Rule

This screen is used to define rules for returning a fragmented network, where a number of H.248 Media Gateways are being serviced by one or more Local Survivable Processors (LSPs), to the primary Avaya S8XXX Server in an automated fashion. The system displays a different warning message and/or time window grid depending on the option selected for the **Migrate H.248 MG to primary** field. The following figures show the screens that appear for each option.

For more information on Auto Fallback for H.248 Gateways, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Field descriptions for page 1

Figure 558: System Parameters Media Gateway Automatic Recovery Rule screen (immediately)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   immediately
Minimum time of network stability: 3

WARNING: The MG shall be migrated at the first possible opportunity. The MG may be
migrated with a number of active calls. These calls shall have their talk paths
preserved, but no additional call processing of features shall be honored. The
user must hang up to regain access to all features.





Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 559: Media Gateway Automatic Recovery Rule Time Entry screen (0-active-calls)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   0-active-calls
Minimum time of network stability: 3

WARNING: The MG shall only be migrated when there are no active calls.






Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 560: Media Gateway Automatic Recovery Rule Time Entry screen (time-day-window)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   time-day-window
Minimum time of network stability: 3

WARNING: The MG may be migrated with a number of active calls. These calls shall
have their talk paths preserved, but no additional call processing of features
shall be honored. The user must hang up in order to regain access to all features.
Valid registrations shall only be accepted during these intervals.


                                        Time of Day
Day of Week  00                             12                              23
Sunday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Monday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Tuesday      __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Wednesday    __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Thursday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Friday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Saturday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 561: Media Gateway Automatic Recovery Rule Time Entry screen (time-window OR 0-active-calls)

```
change system-parameters mg-recovery-rule 1

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name: Rule 1
Migrate H.248 MG to primary:   time-window-OR-0-active-calls
Minimum time of network stability: 3

WARNING: The MG shall be migrated ANY time when there are no active calls OR the
MG may be migrated with a number of active calls when a registration is received
during the specified intervals. These calls shall have their talk paths
preserved, but no additional call processing of features shall be honored.


                                        Time of Day
Day of Week  00                              12                              23
Sunday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Monday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Tuesday      __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Wednesday    __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Thursday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Friday       __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __
Saturday     __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __ __

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Migrate H.248 MG to primary

Use this field to indicate auto-fallback preferences. For each option the system displays a unique warning message and/or time window grid.

You must specify an **x** or **X** for each hour during which you want to permit the return migration. If you do not want to permit a given hour, then they leave it blank. This method helps with overlapping time issues between days of the week. You can specify as many intervals as you wish.

Valid entries	Usage
immediately	The first media gateway registration that comes from the media gateway is honored, regardless of call count or time of day. this is the default.
0-active calls	The first media gateway registration reporting “0 active calls” is honored.
	1 of 2

Valid entries	Usage
time-day-window	A valid registration message received during any part of this interval is honored. When this option is selected the system displays a grid for defining desired hours/days for the time window.
time-window-OR-0 -active-calls	A valid registration is accepted anytime, when a 0 active call count is reported OR if a valid registration with any call count is received during the specified time/day intervals. When this option is selected the system displays a grid for defining desired hours/days for the time window.
	2 of 2

Minimum time of network stability

Use this field to administer the time interval for stability in the H.248 link before auto-fallback is attempted.

Valid entries	Usage
3 to **15**	Enter the number of minutes before auto-fallback is attempted. Default is **3**.

Recovery Rule Number

Valid entries	Usage
1 to server maximum	Enter the number of the recovery rule.

Rule Name

Use this field for an optional text name for the rule, as an aid in associating rules with media gateways.

Valid entries	Usage
Alpha-numeric characters	Enter a name for this recovery rule.

System Parameters - Mode Code

See **Mode Code Related System Parameters**.

System Parameters - Multifrequency Signaling

See **Multifrequency-Signaling-Related Parameters**.

System Parameters OCM Call Classification

This screen enters the tone characteristics for your country for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. If you cannot access this screen, contact your Avaya technical support representative.

This screen appears when **Global Call Classification** field on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**, or when the **Enable Busy Tone Disconnect for Analog loop-start Trunks** field on the **System Parameters Country Options** screen is set to **y**. This screen defines the busy tone and cadence and can be administered with up to 4 on and off steps, which is four valid cycles to determine busy tone.

Avaya recommends that you use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is considered BTD signal.

Field descriptions for page 1

Figure 562: System Parameters OCM Call Classification screen

```
                                                        Page 1 of x
                  SYSTEM PARAMETERS OCM CALL CLASSIFICATION


TONE DETECTION PARAMETERS
  Global Classifier Adjustment (dB): ___
              USA Default Algorithm? _n_
                  USA SIT Algorithm? ___
```

Global Classifier Adjustment (dB)

Enter a number to specify the dB loss adjustment.

Valid entries	Usage
0 to **15**	**0** is the least and **15** the most adjustment.

USA Default Algorithm

Valid entries	Usage
y/n	To use the default United States tone detection, set this field to **y**. If you enter **n**, the **US Special Information Tones (SIT) Algorithm** field appears.

USA SIT Algorithm

Valid entries	Usage
y	To use the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name "intercept" as if they were SIT VACANT, and treats tones with the administered tone name "information" as if they were SIT UNKNOWN.

Field descriptions for page 2

Figure 563: System Parameters OCM Call Classification screen

```
                                                              Page 2 of x
                  SYSTEM PARAMETERS OCM CALL CLASSIFICATION

Tone Name  Instance       Tone      Cadence   Duration     Duration
                       Continuous  Step      Minimum      Maximum
_________   _____         ___      1.  on     _____        _____
                                   2.  off    _____        _____
                                   3.  on     _____        _____
                                   4.  off    _____        _____
                                   5.  on     _____        _____
                                   6.  off    _____        _____
                                   7.  on     _____        _____
                                   8.  off    _____        _____
```

Cadence Step

A display-only field identifying the number of each tone cadence step and indicating whether the tone is on or off during this cadence step.

Duration Maximum

Specifies the upper limit in milliseconds of the tone duration.

Note:

On the **Feature-Related System Parameters** screen, set the **Off-Premises Tone Detect Timeout Interval** field to its maximum value.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Duration Minimum

Specifies the lower limit in milliseconds (msec) of the tone duration.

Valid entries	Usage
75 to **6375**	Enter in increments of 25 msec.

Instance

Enter the instance number of the tone. If the system identifies a tone that matches the characteristics defined on more than one page of this screen the system applies the tone definition from the earlier page.

Valid entries	Usage
1 to **8**	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid entries	Usage
y	Indicates a continuous tone. If you enter **y**, you cannot enter data in the **Duration Minimum** or **Duration Maximum** fields.
n	Indicates a non-continuous tone.

Tone Name

This field is required for tone definition outside of the U.S. and Canada.

If the **Global Call Classification** field on the **System Parameters Customer-Options (Optional Features)** screen is **n**, only busy can be entered into this **Tone Name** field. If **Busy Tone Disconnect** is enabled, only busy can be entered into this field.

Valid entries	Usage
busy **information** **intercept** **reorder** **ringback**	Enter the name of the tone that you are adding or modifying. Enter **busy** for **Busy Tone Disconnect**.

System Parameters - SCCAN

Field descriptions for page 1

Figure 564: SCCAN-Related System Parameters screen

```
change system-parameters sccan                                     Page 1 of x
                      SCCAN - RELATED SYSTEM PARAMETERS

                MM(WSM) Route Pattern: ________
                          H1 Handover: ________
                          H2 Handover: ________
                         Announcement: ________
             Special Digit Conversion? ________
```

MM (WSM) Route Pattern

Enter a route pattern number that is SCCAN-enabled. Partition route pattern indexes, RHNPA indexes, deny, or nodes are not allowed.

Valid entries	Usage
blank	Default value. If this field is left blank, the feature is turned off. To enable this feature, you must enter an acceptable value. This is the default.
digits	Right-click on the field on the SAT screen to see valid entries for your system.

H1 Handover

Valid entries	Usage
unassigned extension	The primary handover number called to facilitate handover of a cellular call to the WAN or WLAN. Depending on whether the user is entering or exiting the Enterprise space, Communication Manager replaces the active call with the new call made using the hand-off H1 or H2 number.

H2 Handover

Valid entries	Usage
unassigned extension	A secondary handover number used when no acknowledgement is received from the H1 Handover number.

Announcement

Valid entries	Usage
assigned announcement extension	Enter the extension of the announcement you want to play during call handin or handout.

Special Digit Conversion

This field allows a user to call a cellular telephone number and get the same treatment as calling an extension that is running Communication Manager.

Valid entries	Usage
y	ARS checks the dialed string to determine if the dialed string is a SCCAN telephone number. If the number is a SCCAN telephone number, the cellular telephone number is replaced with the extension number that the cellular telephone is mapped to.
n	The feature is turned off. This is the default.

System Parameters - Security

See Security-Related System Parameters.

Telecommuting Access

This screen allows the System Administrator to administer the extension which allows remote users to use the feature.

Field descriptions for page 1

Figure 565: Telecommuting Access screen

```
add telecommuting-access
                          TELECOMMUTING ACCESS

                 Telecommuting Access Extension: ________
```

Telecommuting Access Extension

This only allows remote access to the Telecommuting Access feature.

Valid entries	Usage
Unassigned extension of 1 to 13 digits, or blank	Enter an extension that conforms to your system's dial plan and is not assigned to any other system object.

Related Topics

See Configuring Avaya Communication Manager for Telecommuting on page 429 for information about setting up telecommuting.

Tenant

This screen defines tenants to the system. If your server running Communication Manager uses tenant partitioning, see "Tenant Partitioning" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Field descriptions for page 1

Figure 566: Tenant screen

```
change tenant n                                                     Page 1 of x
                           Tenant n

               Tenant Description: ________________________________________
                  Attendant Group: 1
            Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____
                Night Destination: _____
                     Music Source: 1
          Attendant Vectoring VDN:

DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                       Attendant Originated Calls: external

                        COS Group: 1
```

Attendant Group

This required information relates a tenant to an attendant group.

Note:
The default for the system is that all attendant groups exist. However, the attendant group will be empty if no consoles are assigned to it.

Valid entries	Usage
1 to **128**	See *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207, for your system's range of allowable attendant group numbers.

Attendant Vectoring VDN

This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Attendant Vectoring** field is **y** and the **Tenant Partitioning** field is **n**. Enter the assigned Attendant VDN extension or blank. When set to **y**, the **VDN** and **Call Vector** screens display.

COS Group

This field appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **Tenant Partitioning** field is **y**. Use this field to assign this tenant to a Class of Service group.

Valid entries	Usage
1 to **100**	Enter the Class of Service group to which this tenant is assigned.

Ext Alert Port (TAAS)

Enter Trunk Answer Any Station (**TAAS**) alert port information, if any. The port type and the object type must be consistent, and the port can be assigned to only one tenant.

Valid entries	Usage
A valid port address or X **01** to **03** (DEFINITY CSI) or **1** to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth character are the slot number
01 to **04** (Analog TIE trunks) **01** to **31**	Six and seventh characters are the circuit number
1 to **80** (DEFINITYCSI) or **1** to **250** (S87XX/S8300 Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Ext Alert (TAAS) Extension

This field appears only if you entered an **x** in the **Ext Alert Port (TAAS)** field. A system installer can then use the Terminal Translation Initialization (TTI) feature from a telephone plugged into any port to assign this extension number to that port. Doing so makes that port the external alert TAAS port.

Valid entries	Usage
A valid extension	Assign an extension as the external alert TAAS extension.

Music Source

Valid entries	Usage
1 to **20**	Enter the music/tone source for this partition. These sources are defined on the **Music Sources** screen.

Night Destination

Valid entries	Usage
A valid extension	Enter the night service station extension, if you want night service for this tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Tenant Description

Valid entries	Usage
40 alpha-numeric characters or blank	You can leave the description field blank, but future administration will be easier if you provide descriptive information.

DISTINCTIVE AUDIBLE ALERTING

The following Distinctive Audible Alerting fields appear when **Tenant Partitioning** on the **System Parameters Customer Options** screen is **y**. Use these fields to administer distinctive ring patterns per tenant.

Attendant Originated Calls

This field appears when Tenant Partitioning on the **System Parameters Customer Options** screen is **y**.

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls. Default is **external**.

Distinctive Audible Alerting (Internal, External, Priority)

This field appears when Tenant Partitioning on the **System Parameters Customer Options** screen is **y**.

This is also known as Distinctive Ringing. Enter the number of rings for **Internal**, **External**, and **Priority** calls. For virtual stations, this applies to the mapped-to physical telephone. Defaults are as follows:

- **1**: Internal calls
- **2**:External and attendant calls
- **3**:Priority calls

Note:

SIP Enablement Services (SES) messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid entries	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

Field descriptions for page 2

Figure 567: Tenant screen

```
change tenant n                                                Page 1 of x
                                   Tenant n


CALLING PERMISSION (Enter y to grant permission to call specified Tenant)


 1? y  11? n  21? n  31? n  41? n  51? n  61? n  71? n  81? n  91? n
 2? n  12? n  22? n  32? n  42? n  52? n  62? n  72? n  82? n  92? n
 3? n  13? n  23? n  33? n  43? n  53? n  63? n  73? n  83? n  93? n
 4? n  14? n  24? n  34? n  44? n  54? n  64? n  74? n  84? n  94? n
 5? n  15? n  25? n  35? n  45? n  55? n  65? n  75? n  85? n  95? n
 6? n  16? n  26? n  36? n  46? n  56? n  66? n  76? n  86? n  96? n
 7? n  17? n  27? n  37? n  47? n  57? n  67? n  77? n  87? n  97? n
 8? n  18? y  28? n  38? n  48? n  58? n  68? n  78? n  88? n  98? n
 9? n  19? n  29? n  39? n  49? n  59? n  69? n  79? n  89? n  99? n
10? n  20? n  30? n  40? n  50? n  60? n  70? n  80? n  90? n  100? n
```

Calling permissions

The system default allows each tenant to call only itself and Tenant 1. If you want to change that, you can do that on this screen.

Valid entries	Usage
y/n	Enter **y** to establish calling permission between the tenant number that you entered on the command line and any other tenant.

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Terminal Parameters

This screen administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, 8434D, and 2420/2410 telephones. Only authorized Avaya personnel can administer this screen.

Note:

With the Multinational Locations feature enabled, you can administer terminal parameters per location, rather than system-wide.

Field descriptions for page 1

Figure 568: 603/302 Terminal Parameters screen

```
change terminal-parameters                                       Page 1 of x
                      302/603/606-TYPE TERMINAL PARAMETERS

          Base Parameter Set: 1             Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

   OPTIONS*
                   Display Mode: _*               DLI Voltage Level: _________*


   PRIMARY LEVELS*
            Voice Transmit (dB): _____*         Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
```

Figure 569: 6400/607A1/4600/2420 Type Terminal Parameters screen

```
change terminal-parameters                                          Page 2 of x

                  6400/607A1/4600/2420-TYPE TERMINAL PARAMETERS

          Base Parameter Set: 1         Customize Parameters? y
     Note: Location-parameters forms assign terminal parameter sets.
     Note: LEVELS do not apply to the 4600 terminals.*

    OPTIONS*
                    Display Mode:_*       Handset Expander Enabled?
           Volume for DCP Types:_*
            Volume for IP Types:_*
    PRIMARY LEVELS*
             Voice Transmit (dB):__*          Voice Sidetone (dB):__*
              Voice Receive (dB):__*      Touch Tone Sidetone (dB):__*
        Touch Tone Transmit (dB):__*


    BUILT-IN SPEAKER LEVELS*
             Voice Transmit (dB):__*           Voice Receive (dB):__*
                                          Touch Tone Sidetone (dB):__*
    6402 BUILT-IN SPEAKER LEVELS*
              Voice Receive (dB):__*      Touch Tone Sidetone (dB):__*
```

Figure 570: 8400-Series Terminal Parameters screen

```
change terminal-parameters                                           Page 3 of x
                           8400-TYPE TERMINAL PARAMETERS

          Base Parameter Set: __         Customize Parameters? _
     Note: Location-parameters forms assign terminal parameter sets.

OPTIONS*
                   Display Mode: _*              DLI Voltage Level: _________*

   PRIMARY LEVELS*
            Voice Transmit (dB): _____*         Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*    Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
   ADJUNCT LEVELS*+
            Voice Transmit (dB): _____*         Voice Receive (dB): _____*
            Voice Sidetone (dB): _____*   Touch Tone Sidetone (dB): _____*
   BUILT-IN SPEAKER LEVELS
            Voice Transmit (dB): _____*         Voice Receive (dB): _____*
                                          Touch Tone Sidetone (dB): _____*

   8403 BUILT-IN SPEAKER LEVELS
             Voice Receive (dB): _____*   Touch Tone Sidetone (dB): _____*
```

Base Parameter Set

Determines which default set of telephone options and levels will be used. This field corresponds to the country codes. For the country code listing, see the Country code table on page 1579.

Customize Parameters

Indicates whether the administrator wishes to change one or more of the default parameters.

Note:

Beginning with the May 2004 2.1 Release of Communication Manager, when the **Customize Parameters** field on the **Terminal Parameters n** screen is set to **y**, all Base Parameter Set default values display in the parameter fields. You must change the values in fields for which the default is not desired. To change just a few parameters back to default values, temporarily set the **Customize Parameters** field on the **Terminal Parameters n** screen to **n**, but do not submit the screen (do not press **Enter**). Make note of the default values for the specific fields you want to change, then set the Customize Parameters field back to **y**, and enter the default values in the fields.

Valid entries	Usage
y	If this field is **y** (yes), the **Option** and **Level** fields appear and all fields can be edited.
n	If this field is **n** (no), the system uses all default parameters associated with the **Base Parameter Set** field and all fields are display-only.

OPTIONS

Display Mode

Determines how the #) and ~ characters appear on the telephone's display.

Valid entries	Usage
1	If this field is set to **1**, the # and ~ do not change.
2	If this field is set to **2**, the telephone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust.

Handset Expander Enabled

Determines whether the telephone will reduce noise on the handset.

Valid entries	Usage
y	If the field is **y**, the telephone reduces background noise.

Primary levels

The system displays the default setting from the Base Parameter Set for all fields. Also, these fields all require the same input; valid entries are from **-44.0** db through **+14.0** db in 0.5 increments (for example, -44.0, -43.5, -43.0 and son on).

Volume for DCP Types

This field allows the DCP telephone volume to be adjusted while the call is in progress.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

This field allows the IP telephone volume to be adjusted while the call is in progress.

Note:

If you use this field, Avaya recommends that you not change any values in the **PRIMARY LEVELS** or **BUILT-IN SPEAKER LEVELS** areas.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

PRIMARY LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Touch Tone Transmit (dB) —

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the telephone.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the telephone.

Note:

You cannot administer all five of the **Primary Level** fields to +14.0 dB. If you attempt to submit the **Terminal Parameters** screen with all **Primary Levels** set to +14.0 dB, you receive an error message.

ADJUNCT LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

Terminating Extension Group

This screen defines a Terminating Extension Group (TEG). Any telephone can be assigned as a TEG member; however, only a multi-appearance telephone can be assigned a **TEG** button with associated status lamp. The **TEG** button allows the telephone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member's COR. The members could all be termination restricted but still receive calls if the group is not restricted.

The system allows for as many as 32 TEGs with up to 4 members each. An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Field descriptions for page 1

Figure 571: Terminating Extension Group screen

```
change term-ext-group 1                                             Page 1 of x
                       TERMINATING EXTENSION GROUP

                      123456789012345678901234567                1234567890123
        Group Number: 1                         Group Extension: 40999
          Group Name: TERMINATING EXT. GROUP 1    Coverage Path: t77
       Security Code:                                       COR: 1
                                                             TN: 1
ISDN/SIP Caller Disp:                             LWC Reception: spe
          AUDIX Name:

GROUP MEMBER ASSIGNMENTS
    Ext             Name
    1234567890123   123456789012345678901234567
 1: 41153
 2: 41910           Station 41910 on ST2
 3: 41504           Gry Mrkt x41504 4a1803
 4: 41750           st2 4a1802
```

AUDIX Name

Name of the AUDIX machine as it appears in the **IP Node Names** screen.

Valid entries	Usage
Audix machine description	Unique identifiers for adjunct equipment.

COR

Valid entries	Usage
0 to **995**	Enter the class of restriction (COR) number that reflects the desired restrictions.

Coverage Path

Enter a number for the call coverage path for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid entries	Usage
1 to **9999**	path number
t1 to **t999**	time of day table
blank	

Group Extension

Enter the extension of the terminating extension group.

Valid entries	Usage
1 to 7 digits	Unused extension number (cannot be a VDN extension). Do not leave blank.

Group Name

Enter the name used to identify the terminating extension group.

Group Number

A display-only field when the screen is accessed using an administration command such as `add` or `change`.

ISDN Caller Disp

This field is required if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**.

Valid entries	Usage
grp-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
mbr-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
blank	If the **ISDN-PRI** or **ISDN-BRI Trunks** field is **n**, leave blank.

LWC Reception

Defines the source for Leave Word Calling (LWC) messages.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX.
spe	If LWC is attempted, the messages are stored in the system processing element (spe).
none	If LWC is attempted, the messages are not stored.

Security Code

Valid entries	Usage
3 to 8 digit security code	This code is used for the Demand Print feature.

TN

Valid entries	Usage
1 to **100**	Enter the Tenant Partition number.

GROUP MEMBER ASSIGNMENTS

Ext

Enter the extension number (cannot be a VDN extension) assigned to a station.

Valid entries	Usage
1 to 7 digits	An extension number of 1 to 7 digits.

Name

This display-only field shows the name assigned to the preceding extension number when the TEG member's telephone is administered.

TFTP Server

The **Trivial File Transfer Protocol** screen allows specification of the TFTP server that Avaya Communication Manager uses to get download files.

Field descriptions for page 1

Figure 572: TFTP Server Configuration screen

```
change tftp-server                                            Page 1 of x
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

Filename in Memory

A display-only field showing the name of the file currently in Communication Manager memory.

File Size

A display-only field showing the number of bytes transferred.

File Status

A display-only field showing Download In Progress, Download Failed, File Not Found, or Download Completed.

File to Retrieve

Valid entries	Usage
up to 32 alpha-numeric, case sensitive, characters	Enter the name of the file you are going to retrieve using up to 32 alpha-numeric, case sensitive, characters for identification.

Local Node Name

The local node name must be a valid entry from the **IP Node Names** screen. The node must be assigned to a CLAN IP interface or **procr** (processor CLAN).

Valid entries	Usage
1 to 15 characters	Valid entry from the **IP Node Names** screen.
procr	Processor CLAN for S8300/S87XX Servers

TFTP Server Node Name

Valid entries	Usage
1 to 15 characters	The TFTP server node name must be a valid entry from the **IP Node Names** screen.

TFTP Server Port

Valid entries	Usage
1 to **64500**	Enter a number for the remote TCP port.

Time of Day Coverage Table

This screen allows up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Field descriptions for page 1

Figure 573: Time of Day Coverage Table screen

```
change coverage time-of-day n
                   TIME OF DAY COVERAGE TABLE n___

           Act   CVG     Act   CVG     Act   CVG     Act   CVG     Act   CVG
           Time  PATH    Time  PATH    Time  PATH    Time  PATH    Time  PATH

  Sun      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Mon      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Tue      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Wed      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Thu      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Fri      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
  Sat      00:00 ____    __:__ ____    __:__ ____    __:__ ____    __:__ ____
```

Act Time

Specify the activation time of the coverage path administered in the next **CVG PATH** field. Enter the information in 24-hour time format.

Valid entries	Usage
00:01 to **23:59**	If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Enter the coverage path number.

.

Valid entries	Usage
1 to **9999** or blank	For the S87XX Series IP-PNC

Time of Day Coverage Table

A display-only field when the screen is accessed using an administration command. Specifies the Time of Day Coverage Table number. Up to 999 can be administered.

Time of Day Routing Plan

Use this screen to set up Time of Day Routing Plans. You can route AAR and ARS calls based on the time of day each call is made. You can design up to 8 Time of Day Routing Plans, each scheduled to change up to 6 times a day for each day in the week.

Match the Time of Day Routing Plan PGN# with the **PGN#** field on the Partition Routing Table for the route pattern you want to use.

Note:

Automatic Route Selection (ARS) or **Private Networking**, **AAR/ARS Partitioning**, and **Time of Day Routing** must be enabled on the **System Parameters Customer-Options (Optional Features)** screen before you can use Time of Day Routing.

Field descriptions for page 1

Figure 574: Time Of Day Routing Plan screen

```
change time-of-day
                                  TIME OF DAY ROUTING PLAN ___              Page 1 of x
      Act     PGN    Act    PGN    Act     PGN    Act     PGN    Act     PGN    Act     PGN
      Time     #     Time    #     Time     #     Time     #     Time     #     Time     #
Sun   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Mon   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Tue   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Wed   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Thu   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Fri   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
Sat   00:00    1    __:__   _     __:__    _     __:__    _     __:__    _     __:__    _
```

Act Time

Specifies the time of day the route pattern (identified by PGN) begins.

Valid entries	Usage
00:00 to **23:59**	Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

Identifies the route pattern for activation time listed.

Valid entries	Usage
1 to **8**	Enter a PGN that matches the PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day.

Time of Day Routing Plan

Displays the Time of Day Routing Plan number (1 through 8).

Time of Day Station Lock Table

Use this screen to lock stations automatically by a time of day schedule.

Field descriptions for page 1

Figure 575: Time of Day Coverage Table screen

```
change time-of-day station-lock table 1                     page 1 of x
                     TIME OF DAY Station Lock Table 1

     Table Active? y                   Manual Unlock allowed? y


       INTERVAL 1          INTERVAL 2                INTERVAL 3
       Begin End           Begin End                 Begin End
       Time  Time          Time  Time                Time  Time
  Sun  00:00 00:00         00:00 00:00               00:00 00:00
  Mon  00:00 00:00         00:00 00:00               00:00 00:00
  Tue  00:00 00:00         00:00 00:00               00:00 00:00
  Wed  00:00 00:00         00:00 00:00               00:00 00:00
  Thu  00:00 00:00         00:00 00:00               00:00 00:00
  Fri  00:00 00:00         00:00 00:00               00:00 00:00
  Sat  00:00 00:00         00:00 00:00               00:00 00:00
```

Interval (1, 2, 3)

Use these fields to indicate the TOD Station Lock Interval. There are seven rows of entries for 7 days of the week, each row starting with a fixed day entry. The first row starts with Sunday (Sun). The administration will impose validation of overlapping intervals or invalid blank entries.

Valid entries	Usage
0 to **23** or blank for hours and **0** to **59** or blank for minutes	Enter the desired TOD Station Lock Intervals.

Manual unlock allowed

Use this field to indicate if the TOD Station Lock Interval can be deactivated by the manual Station Lock sequence.

Valid entries	Usage
y/n	When set to **y**, the user can manually unlock the TOD-locked station using either a **sta-lock** button or a Feature Access Code followed by an SSC. When set to **n**, the user cannot unlock the station. Default is **n**.

Table Active

Use this field to indicate if this Time-Of-Day-Lock Table is activated or deactivated. Enter n to turn off TOD Station lock for all stations associated to this table. Valid entries are **y**(es), **n**(o). Default is **n**.

Valid entries	Usage
y	Enter **y** to turn on TOD Station lock for all stations associated to this table.
n	Enter **n** to turn off TOD Station lock for all stations associated to this table. Default is **n**.

Toll Analysis

Note:

The **Toll List** field on this screen does not interact with or relate to the ARS Toll Table.

This screen associates dialed strings to the system's Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. You can force users to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize system security, Avaya recommends that toll calling areas be restricted as much as possible through the use of the **RCL (Restricted Call List)** and **Toll List** fields on this screen.

Field descriptions for page 1

Figure 576: Toll Analysis screen

```
change toll n                                                   Page 1 of x
                                TOLL ANALYSIS                  Percent Full:_
                               Location:
                         Total        Toll  CDR  <--Unrestricted Call List-->
    Dialed String       Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _    _  _  _  _  _  _  _  _  _  _
```

CDR FEAC

Valid entries	Usage
x	Enter **x** to require an account code from a call whose facility COR requires a Forced Entry of Account Code.

Dialed String

Valid entries	Usage
digits **0** to **9** (up to 18 characters)	Enter the dialed string you want Avaya Communication Manager to analyze.
***, x, X**	wildcard characters

Location

Display-only field.

Valid entries	Usage
1 to **64**	Defines the server's location for this Toll Analysis Table. On the **System Parameters Customer-Options (Optional Features)** screen, the **ARS** field and the **Multiple Locations** field must be set to **y** for values other than **all** to appear.
all	Indicates that this Toll Analysis Table is the default for all port network (cabinet) locations.

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this dialed string.

Min

Valid entries	Usage
1 to **Max**	Enter the minimum number of user-dialed digits the system collects to match to this dialed string.

Percent Full

Display-only field showing the percentage (**0** to **100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

RCL

Enter **x** to assign the Dialed String to the Restricted Call List (RCL).

Valid entries	Usage
x	All entries of **x** and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR. A call attempt from a facility whose COR is marked as being associated with the RCL and whose dialed string matches a RCL dialed string field will be denied. The caller receives intercept treatment.

Toll List

Valid entries	Usage
x	Enter **x** to assign the Dialed String to the Toll List.

Dialed String	Min	Max	Toll List
0	1	23	x
1	4	23	x
20	10	10	x
21	10	10	x
30	10	10	x
			1 of 2

Dialed String	Min	Max	Toll List
31	10	10	x
40	10	10	x
41	10	10	x
50	10	10	x
51	10	10	x
60	10	10	x
61	10	10	x
70	10	10	x
71	10	10	x
80	10	10	x
81	10	10	x
90	10	10	x
91	10	10	x
			2 of 2

Unrestricted Call List

Valid entries	Usage
x	Enter **x** to assign the dialed string to one of the system's Unrestricted Call Lists (UCL).

Tone Generation

The **Tone Generation** screen allows you to administer the tone characteristics that parties on a call hear under various circumstances.

Note:

With the Multinational Locations feature enabled, tone generation can be administered per location, rather than system-wide.

Field descriptions for page 1

Figure 577: Tone Generation screen

```
change tone-generation 2                                        Page 1 of  X
                             TONE GENERATION 2

                                                Base Tone Generator Set: 1
          440Hz PBX-dial Tone? n              440Hz Secondary-dial Tone? n
```

440Hz PBX-dial Tone

Specifies whether the switch (primary) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

440Hz Secondary-dial Tone

Specifies whether the Secondary (CO) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in **Base Tone Generation Set**.

Base Tone Generator Set

The country code identifies the base tone generation set to be used. For information on the appropriate tone-generation hardware to use in a specific country, see *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207.

Valid entries	Usage
1 to **25**	See the Country code table at the beginning of the **System-Parameters Country-Code** screen description.

Field descriptions for page 2

Figure 578: Tone Generation screen

```
change tone-generation                                          Page   2 of  X
                              TONE GENERATION CUSTOMIZED TONES

        Tone Name         Cadence        Tone
                            Step    (Frequency/Level)
         Hold                1:      480/-17.25            Duration(msec): 1000
                             2:      goto                            Step:
                             3:
                             4:
                             5:
                             6:
                             7:
                             8:
                             9:
                            10:
                            11:
                            12:
                            13:
                            14:
                            15:
```

Cadence Step

Display-only fields that identify the number of each tone cadence step.

Valid entries	Usage
1 to **15**	Identifies the number of each tone cadence step.

Duration (msec)

Valid entries	Usage
50 to **12750**, in increments of 50	Enter the duration of this step in the tone sequence.

Step

This field appears when you enter **goto** in the **Tone/Frequency Level** field.

Valid entries	Usage
Cadence step	.Enter the number of the cadence step for this **goto** command.

Tone (Frequency/Level)

Valid entries	Usage
silence	An entry of **silence** means no tone. A final step of **silence** with an infinite duration will be added internally to any tone sequence that does not end in a **goto**.
	1 of 2

Valid entries	Usage
goto	An entry of **goto** means to repeat all or part of the sequence, beginning at the specified cadence step.
350/-17.25 **350+425/-4.0** **350+440/-13.75** **375+425/-15.0** **404/-11.0** **404/-16.0** **404+425/-11.0** **404+450/-11.0** **425/-4.0** **425/-11.0** **425/-17.25** **440/-17.25** **440+480/-19.0** **450/-10** **480/-17.25** **480+620/-24.0** **525/-11.0** **620/-17.25** **697/-8.5** **770/-8.5** **852/-8.5** **941/-8.5** **1000/0.0** **1000/+3.0** **1004/0.0** **1004/-16.0** **1209/-7.5** **1336/-7.5** **1400/-11.0** **1477/-7.5** **1633/-7.5** **2025/-12.1** **2100/-12.1** **2225/-12.1** **2804/-16.0**	Specifies the frequency and level of the tone.
	2 of 2

Tone Name

Each entry uses one of the keywords below to indicate which of the individually administrable tones this screen modifies.

Valid entries	Usage
blank	If this field is blank, all entries are ignored in the corresponding **Tone (Frequency/Level)** field.
1-call-wait **2-call-wait** **3-call-wait** **busy** **busy-verify** **call-wait-ringback** **conference** **confirmation** **disable-dial** **hold** **hold-recall** **immed-ringback** **intercept** **intrusion** **mntr/rec-warning** **PBX-dial** **recall-dial** **recall-dont-ans** **redirect** **reorder** **rep-confirmation** **reset-shift** **ringback** **secondary-dial** **special-dial** **whisper-page** **zip**	

Note:

For information on setting the Caller Response Interval before a call goes to coverage (when the value for this field is **redirect**), see "Caller Response Interval" in the Call Coverage section of *Feature Description and Implementation for Avaya Communication Manage*r, 555-245-205.

Trunk Group

Use the **Trunk Group** screen to set basic characteristics for every type of trunk group and to assign ports to the group. This section lists and describes all the fields you might see on the screen. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in the **Group Type** field might significantly change the content and appearance of the **Trunk Group** screen.

For more information on administering trunk groups, see Managing Trunks on page 477.

Note:

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Field descriptions for page 1

The figure below is only an example, and is intended to show most of the fields that might appear on page 1 of the **Trunk Group** screen. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Figure 579: Trunk Group screen - page 1

```
add trunk-group next                                                   Page 1 of x
                                   TRUNK GROUP

Group Number: 8                    Group Type: co CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1 TN: 1 TAC:
   Direction: two-way       Outgoing Display? n
 Dial Access? n            Busy Threshold: 255   Night Service: 1234567890123
Queue Length: 0          Country: 1       Incoming Destination: 1234567890123
   Comm Type: voice                 Auth Code? n    Digit Absorption List:
    Prefix-1? y                   Trunk Flash? n         Toll Restricted? y

         Trunk Type:
```

Analog Gain

Use this field to reduce the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume. Experiment to find the best setting for your situation. This field appears if the **Country** field is **15** and the **Trunk Type (in/out)** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

Valid entries	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. Don’t change this setting unless the trunk group’s sound quality is unacceptable.

Auth Code

This field affects the level of security for tandemed outgoing calls at your server running Communication Manager. This field appears if the **Direction** field is **incoming** or **two-way**, and it can only be **y** if the **Authorization Codes** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
y/n	Enter **y** to require callers to enter an authorization code in order to tandem a call through an AAR or ARS route pattern. The code will be required even if the facility restriction level of the incoming trunk group is normally sufficient to send the call out over the route pattern.

BCC

Generalized Route Selection uses the BCC to select the appropriate facilities for routing voice and data calls. Far-end tandem servers/switches also use the BCC to select outgoing routing facilities with equivalent BCC classes.The entry in the **Bearer Capability Class** field is used to select the appropriate facilities for incoming ISDN calls. Avaya Communication Manager compares the entry in the **BCC** field to the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities. For example, a call with BCC 4 will only be connected through facilities that support 64 kbps data transmission.

The **Bearer Capability Class** field appears when all of the following are true:

- Either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field on the **System Parameters Customer-Options (Optional Features)** screen is **y**.
- The **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**.
- The **Comm Type** field is **data**, **avd**, or **rbavd**.

Valid entries	Usage
0	For voice and voice-grade data
1	For 56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
4	For 64 kbps data on unrestricted channels

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Valid entries	Usage
1 to **255** (S87XX Series IP-PNC)	Enter the number of trunks that must be busy in order to light the warning lamp on the Attendant Console. For example, if there are 30 trunks in the group and you want to alert the attendant whenever 25 or more are in use, enter **25**.

CDR Reports

Valid entries	Usage
y	All outgoing calls on this trunk group will generate call detail records. If the **Record Outgoing Calls Only** field on the **CDR System Parameters** screen is **n**, then incoming calls on this trunk group will also generate call detail records.
n	Calls over this trunk group will not generate call detail records.
r (ring-intvl)	CDR records will be generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: ● Abandoned calls: The system creates a record with a condition code of "H," indicating the time until the call was abandoned. ● Answered calls: The system creates a record with a condition code of "G," indicating the interval from start of ring to answer. ● Calls to busy stations: The system creates a record with a condition code of "I," indicating a recorded interval of 0.

Note:

For ISDN trunk groups, the **Charge Advice** field affects CDR information. For CO, DIOD, FX, and WATS trunk groups, the **PPM** field affects CDR information.

CESID I Digits Sent

This field appears when **Group Type** is **cama**. For emergency 911 service, Communication Manager might send Caller's Emergency Service Identification (CESID) information to the central office or E911 tandem server/switch. This digit string is part of the E911 signaling protocol.

Valid entries	Usage
1 to 3 digits	Determine the correct entry for this field by talking to your E911 provider.

Comm Type

Use this field to define whether the trunk group carries voice, data, or both.

Note:
Comm Types of **avd**, **rbavd** and **data** require trunk member ports on a DS1 circuit pack.

Valid entries	Usage
avd	Enter **avd** for applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving end server discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating ends insert a modem pool for any calls when **Comm Type** is **avd**. The **Signaling Mode** field on the **DS1 Circuit Pack** screen must be set for either **common-chan** or **CAS** signaling.
data	Enter **data** only when all calls across the trunk group originate and terminate at Avaya Communication Manager digital data endpoints. Public networks don't support **data**: supported by Avaya's DCP protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The **Signaling Mode** field on the **DS1 Circuit Pack** screen might be set to **robbed-bit** or **common-chan**.
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The **Signaling Mode** field on the **DS1 Circuit Pack** screen must be set to **robbed-bit** unless mixed mode signaling is allowed on the DS1 circuit pack. In that case, the **Signaling Mode** field might be **isdn-ext** or **isdn-pri**.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use **voice**.

COR

Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed. For details on using COR and FRLs, see *Avaya Toll Fraud and Security Handbook*, 555-025-600.

Valid entries	Usage
0 to **995**	Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.

Tip:

Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

CO Type

This field appears when the **Country** field is **14** and is used only by trunk group members administered on a TN464D vintage 2 or later DS1 circuit pack.

Valid entries	Usage
analog	This field specifies whether the trunk group is connected to analog or digital facilities at the central office.
digital	

Country

This field is administered at installation and sets numerous parameters to appropriate values for the public network in which the server running Communication Manager operates. For example, the value of this field, with the values of the **Trunk Termination** and the **Trunk Gain** fields, determines the input and trans-hybrid balance impedance requirements for ports on TN465B, TN2146, and TN2147 circuit packs.

This field appears for the trunk groups that connect Avaya Communication Manager to a central office in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

CAUTION:

Do not change this field. If you have questions, contact your Avaya technical support representative.

Note:

For DID trunk types, country code **19** is not accepted in the **Trunk Group** screen in Communication Manager. This will be supported at a later date.

Valid entries	Usage
1 to **25**	Set at installation. For a list of country codes, see the Country code table on page 1579.
11	If the **Country** field is **11**, Avaya Communication Manager is administered for Public Network Call Priority (Call Retention and Re-ring).
	1 of 2

Valid entries	Usage
14	If the **Country** field is **14** and the **Group Type** is **DID** or **DIOD**, the **CO Type** field appears.
15	If the **Country** field is **15**, Communication Manager is administered for Public Network Call Priority (Intrusion and Re-ring). Also, the **Protocol Type** field appears for **Group Type DID** or **DIOD**.
18	If the **Country** field is **18**, Avaya Communication Manager can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Re-ring).
23	If the **Country** field is **23** and **Group Type** field is either **CO** or **DID**, Communication Manager is administered for Block Collect Calls.
	2 of 2

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

SECURITY ALERT:
Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and aren't restricted by facility restriction levels. For security, you might want to leave the field set to n unless you need dial access to test the trunk group.

Valid entries	Usage
y	Allows users to access the trunk group by dialing its access code.
n	Does not allow users to access the trunk group by dialing its access code. Attendants can still select this trunk group with a **Trunk Group Select** button.

Digit Absorption List

This field assigns a digit absorption list, when used, to a trunk group that terminates at a step-by-step central office.

Valid entries	Usage
0 to **4** or blank	Enter the number of the digit absorption list this trunk group should use.

Note:

In a DCS network, DCS features that use the **remote-tgs** button (on telephones at a remote end) do not work when the incoming trunk group at your end deletes or inserts digits on incoming calls. The **remote-tgs** button on a remote server/ switch, for example, tries to dial a TAC on your switch. If your end adds or deletes digits, it defeats this operation. If you need to manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Direction

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the **Administrable Timers** page. This field appears for all trunk groups except DID and CPE.

Valid entries	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing.
two-way	Enter two-way for Network Call Redirection.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a unique name that provides information about the trunk group. Do not use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: Qwest Local; Sprint Toll, etc. **Note:** The **Group Name** field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701. For more information on Unicode language display, see Administering Unicode display on page 203. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Group Number

This field displays the group number assigned when the trunk group was added.

Note:
For trunk groups connecting two Avaya S8XXX Servers in Distributed Communication System networks, Avaya recommends that you assign the same group number on both servers.

Group Type

Enter the type of trunk group. The fields that are displayed and available might change according to the trunk group type selected. Busy-out the trunk group before you change the group type. Release the trunk group after you make the change. For more information about busying out and releasing trunk groups, see your system's maintenance documentation.

For more information about ISDN trunk group screens, see ISDN Trunk Group.

Valid entries	Usage
Access	Use access trunks to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main server. Allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks are used in private networks. APLT trunks allow inband ANI.
CAMA	CAMA trunks route emergency calls to the local community's Enhanced 911 systems.
CO	CO trunks typically connect Communication Manager to the local central office, but they can also connect adjuncts such as external paging systems and data modules.
CPE	Use CPE trunks to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Use DID trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point. Allows Inband ANI.
DIOD	DIOD trunks are two-way trunks that transmit dialed digits in both directions. In North America, use tie trunks for applications that require two-way transmission of dialed digits. Allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. DMI-BOS trunks allow inband ANI.
	1 of 2

Valid entries	Usage
FX	An FX trunk is essentially a CO trunk that connects your server running Communication Manager directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.
ISDN	Use ISDN trunks when you need digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Use **ISDN** for Network Call Transfer. **Note:** You cannot enter **ISDN** unless the **ISDN-PRI** field, the **ISDN-BRI Trunks** field, or both have been enabled on the **System Parameters Customer-Options (Optional Features)** screen.
RLT	Release Link trunks work with Centralized Attendant Service in a private network.
SIP	Use **SIP** trunks to connect a server running Communication Manager to a SIP Enablement Services (SES) home server, or to connect two Communication Manager servers. **Note:** The Automatic CallBack, Priority Calling, and Whisper Page features currently do not work correctly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Tandem trunks connect tandem nodes in a private network. Tandem trunks allow inband ANI.
Tie	Use tie trunks to connect a server running Communication Manager to a central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls, and allow inband ANI.
WATS	Use WATS trunks to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. Outgoing WATS service allows calls to certain areas ("WATS bands") for a flat monthly charge. Incoming WATS trunks allow you to offer toll-free calling to customers and employees.
	2 of 2

Incoming Destination

Use this field to set the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination you enter here is also the default night service destination unless you enter a different destination in the **Night Service** field. Appears when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type** (in/out) field is not **auto/...**.
valid extension number	Calls go to the extension you enter. You can enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it is active. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

ITC

The Generalized Route Selection feature, part of the automatic routing technology used in Avaya Communication Manager, compares the line coding of available digital facilities and selects appropriate routes for voice and data calls. The **Information Transfer Capability** field appears when the **Comm Type** field is **data**, **avd**, or **rbavd** and the **BCC** field is not **0**.

Valid entries	Usage
rest(ricted)	Restricted trunks use **ami-basic** or **ami-zcs** line coding and can carry only restricted calls.
unre(stricted)	Unrestricted trunks use **b8zs**, **hdb3,** or **cmi** line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC can have only unrestricted trunks as members.

Night Service

This field sets the destination to which incoming calls go when **Night Service** is in operation. If a **Night** field on the **Group Member Assignments** page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:

Whenever possible, use a night service destination on your switch: otherwise some features won't work correctly, even over a DCS network.

Valid entries	Usage
blank	Leave this field blank if the **Trunk Type (in/out)** field is not auto/….
An extension number (can be a VDN)	Enter the extension of your night service destination. **Note:** If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Number of Members

This field appears only when **Group Type** is **sip**.

Valid entries	Usage
1 to **255**	Type the number of SIP Enablement Services (SES) trunks that are members of the trunk group. All members of an SES trunk group will have the same characteristics. **Note:** Member pages for SES trunk groups are completed automatically based on this entry and are not individually administrable.

Outgoing Display

This field allows display telephones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information might be useful to you when you're trying to diagnose trunking problems.

Valid entries	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Prefix-1

Use this field for outgoing and two-way trunk groups handling long distance service. This field appears for CO, FX, and DIOD trunk groups.

Valid entries	Usage
y/n	Enter **y** to add the prefix "1" to the beginning of the digit string for outgoing calls. Do not enter **y** for trunk groups in AAR or ARS route patterns.

Protocol Type

This field specifies the type of line signaling protocol used for DID and DIOD trunk groups. This field appears when the **Country** field is **15** and is used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack. For a list of country codes, see the Country code table on page 1579.

Valid entries	Usage
inloc (Incoming local)	Enter the protocol the central office is using for this trunk group. Only the inloc protocol provides ANI.
intol (Incoming toll)	

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, Communication Manager will call the extension that placed the original call. The caller will hear 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: Communication Manager remembers the number the caller dialed and automatically completes the call.

This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
0 to **100**	Enter the number of outgoing calls that you want to be held waiting when all trunks are busy.
0	Enter **0** for DCS trunks.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined Facility Type of **0** (feature) or **1** (service) on the **Network Facilities** screen is allowed.

As many as 10 ISDN trunk groups can have this field administered as **cbc** (for Avaya DEFINITY Server CSI).

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
	1 of 2

Valid entries	Usage
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be set to **y**.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.
	2 of 2

Signaling Group

The screen displays this field only when the value of the entry in the **Group Type** field is **sip**.

Valid entries	Usage
1 to **650**	Type the number of the SIP Enablement Services (SES) signaling group associated with this trunk group in the **Group Number** field on the Signaling Group screen.

TAC

Enter the trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

Valid entries	Usage
1 to 4 digit number	Enter any number that fits the format for trunk access codes or dial access codes defined in your dial plan.
*****, **#**	* and # can be used as the first character in a TAC.

TN

Valid entries	Usage
1 to **100** (S87XX Series IP-PNC)	Enter a Tenant Partition number to assign this trunk group to the partition.

Tip:

Double-check your entry. If you accidentally enter an unassigned tenant partition number, the system will accept the entry but no calls will go to the trunk group.

Toll Restricted

Valid entries	Usage
y	Enter **y** to prevent toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.
n	Enter **n** if the field is automatic or if you don’t want to restrict access.

Tip:

To find out what kind of line coding a trunk group member uses, check the **Line Coding** field on the **DS1 Circuit Pack** screen for the DS1 port to which the member is assigned.

Trunk Flash

Trunk Flash enables multifunction telephones on Avaya Communication Manager to access central office customized services that are provided by servers at the far-end or Central Office (CO). These central office customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the Communication Manager telephone on an active trunk call.

Valid entries	Usage
y/n	Enter **y** to allow trunk flash.

Trunk Signaling Type

This field controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field also controls the signaling used by members in public network digital trunk groups. This field displays if the **Group Type** field is **access**, **aplt**, **rlt**, **tandem**, or **tie**. Entries in this field affect which timers appear on the **Administrable Timers** page.

Valid entries	Usage
cont (continuous) **dis** (discontinuous)	E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling you select on Communication Manager must match the signaling type administered on the far-end server. Use these values only when all trunk group members are assigned to ports on a TN464F, TN2464, or TN2140 circuit pack. Entering one of these values causes the **Send Release Ack**, **Receive Release Ack**, and **Send Answer Supervision** fields to appear. See Trunk Type (in/out) on page 1687 for more information.
Use the following entries for tie trunks in Main-Satellite/Tributary networks. Each entry defines a function of the trunk group in the network. Use these values only when all trunk group members are assigned to a TN497 circuit pack.	
tgu (for outgoing trunks)	Enter **tgu** at the main server running Communication Manager to administer a tie trunk group connected to a satellite server. (This same group should be administered as **tge** at the satellite.)
tge (for incoming trunks)	Enter **tge** at a satellite server to administer a tie trunk group connected to the main server running Communication Manager. (This same group should be administered as **tgu** at the main server.)
	1 of 2

Valid entries	Usage
tgi (for internal trunks)	Enter **tgi** at to administer a two-way tie trunk group between 2 satellites or between the main server and a satellite. (This trunk group should be administered as **tgi** on both servers.)
DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. Use the following entries for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.	
cont **pulsed** **discont**	Enter **cont** for continuous E&M signaling. Enter **pulsed** for pulsed E&M signaling. Leave blank forR2 signaling. Hungary uses discontinuous E&M signaling when this field is **dis**. Brazil E&M trunks use continuous and pulsed E&M.
	2 of 2

Trunk Type (in/out)

Use this field to control the seizure and start-dial signaling used on this trunk group. The setting of the **Trunk Signaling Type** field can affect the entries allowed in this field. In addition, settings might differ for incoming and outgoing trunks.

Valid entries	Usage
auto **cont** **delay** **disc** **immed** **wink**	There are numerous valid entries for this field: use the online help in Communication Manager to view all the possible combinations. Here are what the elements used in those combinations mean: ● auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. ● cont — Continuous signaling is used with Italian E&M tie trunks. The server/switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer. See Trunk Hunt on page 1707 for more information. ● delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits. ● disc —Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The Avaya S8XXX Server can seize a trunk by sending a single, short signal for the duration specified by the **Normal Outgoing Seize Send** field. However, with the Three-Way Seizure option, the calling end can also send routing information to the called end by sending one or a series of brief seizure signals. ● wink — The sending server or switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it's ready to receive the digits. ● immed — The sending server or switch sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Tip:

When incoming trunks use the setting immed/immed, the far-end server seizes the trunk and sends digits without waiting for acknowledgment from the receiving end. When traffic is heavy, the receiving server or switch might not immediately attach a Touch Tone Receiver to a call and therefore lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end server or switch to avoid this problem.

Note:

The value in this field affects the appearance of the **Incoming Partial Dial (sec)** field on the **Administrable Timer** page.

Version

Use this field to adjust the signaling on multi-country CO trunk circuit packs. Entries in this field adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country. The field appears only for CO, FX, and WATS trunk groups when the **Country** field is **5**, **16**, or **23**.

Valid entries	Usage
If the **Country** field is **5**, the **Version** field only controls TN2147 ports.	
a	Enter **a** to use standard signaling for the Netherlands public network.
b	Enter **b** to use country 1 (U.S.) signaling. The value **b** is appropriate if Communication Manager is connected to a central office using an Ericcson AXE-10 switch.
If the **Country** field is **16** or **23**, the **Version** field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.	
a	Enter **a** to set input impedance to 600 Ohms.
b	Enter **b** to set input impedance to 900 Ohms. The value **b** is appropriate in Brazil.

This field appears when the "out" side of the entry in the **Trunk Type (in/out)** field is **…/wink** or **…/delay** and the **Group Type** is **tie**, **access**, **aplt**, **dmi-bos**, **rlt**, or **tandem**. The setting in this field only affects trunks administered to ports on TN760C (vintage 4 or later), TN767, TN464C (or later), and TN2242 circuit packs. If the trunk group also contains trunks assigned to ports on other types of circuit packs, those trunks are unaffected.

Valid entries	Usage
300 to **5000** in increments of 50	In general, Avaya recommends that you not change this field. If you do, remember that timing on your server running Communication Manager must be compatible with the timing on the far-end server.

Field descriptions for page 2

The figure below is only an example, and is intended to show the fields that might appear on page 2 of the **Trunk Group** screen for one particular trunk type. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Figure 580: Trunk Group screen (page 2)

```
add trunk-group next                                              Page   2 of  x
      Group Type: co                     Trunk Type:

TRUNK PARAMETERS

    Outgoing Dial Type: tone                               Cut-Through? n
     Trunk Termination: rc                         Incoming Dial Type: tone
                                              Disconnect Timing(msec): 500


           Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none
     Analog Loss Group: 6                          Digital Loss Group: 11
                                  Trunk Gain: high

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 10          Receive Answer Supervision? n
          Administer Timers? y
```

Administer Timers

This field is displayed for all trunk group types except **cpe**, **h.323**, and **sip**.

Valid entries	Usage
y/n	Enter **y** to allow administration of timers on this trunk group. For **Group Type isdn**, the default value is **n**. For all other trunk group types, the default is **y**.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plan.

Answer Supervision Timeout

If the **Receive Answer Supervision** field is **n**, use this field to set the answer supervision timer for outgoing and two-way trunks. During a cut-through operation, timing begins after each outgoing digit is sent by Communication Manager and timing ceases after the far-end sends answer supervision. If the timer expires, Communication Manager acts as if it had received answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Valid entries	Usage
0 to **250**	Enter the number of seconds you want Communication Manager to wait before it acts as though answer supervision has been received from the far-end. Set this field to **0** if **Receive Answer Supervision** is **y**.

Note:

This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers. To control answer supervision sent by DS1 firmware, set the **Outgoing End of Dial (sec)** field on the **Administrable Timers** page of the **Trunk Group** screen.

Auto Guard

This field appears if the **Group Type** field is **co** or **fx**. This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports consider a trunk defective if no dial tone is detected on an outgoing call, and the **Outpulse Without Tone** field is **n** on the **Feature-Related System Parameters** screen.

Valid entries	Usage
y/n	Enter **y** to prevent repeated seizures of a defective trunk. Communication Manager will do a maintenance busy-out on these trunks.

Bit Rate

This field specifies the baud rate to be used by pooled modems. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
300	Enter the speed of the fastest modem that will use this trunk group.
1200	
2400	
4800	
9600	
19200	

Call Still Held

This field appears if the **Group Type** field is **co** or **fx**. This field is used when the receiving end server initiates the disconnection of incoming calls. It effectively extends the Incoming Glare Guard timer by 140 seconds. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the Country Code field is **2**.

Valid entries	Usage
y/n	Enter **y** to prevent glare by delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call.

Cut-Through

This field appears when the **Outgoing Dial Type** field is either **rotary** or **tone**.

SECURITY ALERT:
Entering **y** in this field will reduce your ability to prevent toll fraud.

Valid entries	Usage
y	Enter **y** to allow users to get dial tone directly from the central office. Outgoing calls over this trunk group will bypass AAR/ARS (if you're using it) and any of your administered restrictions (such as COR or FRL).
n	Enter **n** and the user will receive switch dial tone. Instead of digits being sent to the central office, they will be collected and checked against administered restrictions. If no restrictions apply, the digits are sent to the central office.

Cyclical Hunt

When a call is offered to a trunk group, Communication Manager searches for an available trunk. This field, which appears when the **Direction** field is **two-way** and the **Trunk Type** field is **loop-start**, controls the starting point of this search.

You can change this field from **n** to **y** at any time. To change from **y** to **n**, however, all the trunks in the group must be idle or busied out.

Valid entries	Usage
y	Enter **y** to have Communication Manager start its search from the last trunk seized. This method is faster, and thus better suited for high-traffic trunk groups.
n	Enter **n** to have Communication Manager start each search at member 1 (the first trunk administered on the Group Member Assignments page).

Dial Detection

Applies only to TN2199 ports. The **Country** field must be **15**.

Valid entries	Usage
A-wire	Indicate whether digit pulses are detected by observing the A-wire (default) or the B-wire only.
B-wire	

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **19**	Shows the index into the loss plan and tone plan.

Digits

This field is used with the **Digit Treatment** field, and its meaning depends on the entry in that field. If the **Digit Treatment** field is **absorption**, this field specifies *how many* digits are deleted. If the **Digit Treatment** field is **insertion**, this field identifies the *specific digits* that are added.

Valid entries	Usage
1 to **5**	Enter the number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	Enter the actual digits to be added (inserted).
blank	This field can be blank only if the **Digit Treatment** field is blank.

Digit Treatment

Use this field to modify an incoming digit string (as on DID and tie trunks, for example) by adding or deleting digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Valid entries	Usage
blank	The incoming digit string is not changed.
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.

If you enter absorption or insertion, then you must enter a value in the **Digits** field.

Disconnect Supervision-In

This field indicates whether Avaya Communication Manager receives disconnect supervision for incoming calls over this trunk group. It appears when the **Direction** field is **incoming** or **two-way**. (If the **Direction** field is **outgoing**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the **Feature-Related System Parameters** screen must be **y**).

If a user connects 2 trunks through conference or transfer, and neither far-end server on the resulting connection provides disconnect supervision, the trunks involved will not be released because Avaya Communication Manager cannot detect the end of the call. Communication Manager will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly might cause trunks to become unusable until the problem is detected and the trunks are reset.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end server/switch sends a release signal when the calling party releases an incoming call, and you want to make the far-end server/switch responsible for releasing the trunk. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on incoming calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

In general, U.S. central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with your network services provider.

Disconnect Supervision-Out

This field indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. It appears when the **Direction** field is either **outgoing** or **two-way**. (If the **Direction** field is **incoming**, Communication Manager internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the **Trunk-to-Trunk Transfer** field on the **Feature-Related System Parameters** screen must be **y**). If a user connects 2 trunks through conference or transfer, and neither far-end server/switch on the resulting connection provides disconnect supervision, the trunks involved won't be released because Communication Manager can't detect the end of the call. Communication Manager will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly might cause trunks to become unusable until the problem is detected and the trunks are reset.

Also, remember that Avaya Communication Manager must receive *answer* supervision on outgoing analog CO, FX, WATS, Tie, Tandem, and Access trunks before it will recognize a disconnect signal. If this trunk group does not receive *answer* supervision from the far-end server/switch, and you enter **y** in this field, Avaya Communication Manager will internally set the field to **n**.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end sends a release signal when the called party releases a call an outgoing call, and you want to make the far-end responsible for releasing the trunk. The **Answer Supervision Timeout** field must be **0** and the **Receive Answer Supervision** field must be **y** for the switch to recognize a **y** entry. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end server/switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on outgoing calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

Do not set this field to **y** unless you are certain that the far-end server/switch will provide answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with your network services provider.

Disconnect Timing (msec)

This field specifies the minimum time in milliseconds that the central office or far-end server requires to recognize that your end has disconnected from a call. This timer does not affect ports on a circuit pack that uses the administrable Incoming Disconnect and Outgoing Disconnect timers; in fact, settings on those two timers override this field.

Valid entries	Usage
140 to **2550** ms in increments of 10	The default of **500** is an industry standard and you shouldn't change it. If you set this field too high, your server/switch won't disconnect sometimes when it should; too low, and it will disconnect when it shouldn't.

Disconnect Type

This field indicates which side or user controls the disconnect, where **A** refers to the calling party and **B** refers to the called party. Appears only if the **Country** field is **15** and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

This applies *only* to the TN2199 port.

Valid entries	Usage
AandB	Both parties control the disconnect.
AorB	Either party controls the disconnect.

Drop Treatment

This field only applies to DID trunks. It determines what the calling party hears when the called party terminates an incoming call.

Valid entries	Usage
intercept	Select one. For security reasons, it’s better to apply a tone: silence could provide an opening for hackers.
busy	
silence	

Note:

In Italy, the **Drop Treatment** field must be administered as **intercept** for all DID trunk groups.

Duplex

This field specifies whether a two-way trunk group allows simultaneous transmission in both directions. This field appears when the **Comm Type** field is **avd** or **rbavd**. It also appears if the **Comm Type** field is **data**, but only if the **ISDN-PRI** field is enabled on the **System Parameters Customer-Options (Optional Features)** screen.

Note:

Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid entries	Usage
full	Enter **full** in most cases: this allows simultaneous two-way transmission, which is most efficient.
half	Enter **half** to support only one transmission direction at a time.

End-to-End Signaling

This field appears when the **Group Type** field is **cpe** (customer-provided equipment trunk groups). Auxiliary equipment such as paging equipment and music sources might be connected to Avaya Communication Manager by auxiliary trunks. Communication Manager might send DTMF signals (touch tones) to these devices. This field sets the duration of these tones.

Valid entries	Usage
60 to **360** ms in increments of 10	Use this field to set the duration in milliseconds of the touch-tone signal that is sent to the connected equipment.

Note:

For trunks that do not receive real answer supervision, a "connect" Event report is sent when the Answer Supervision Timeout occurs.

Expected Digits

Note:

Set this field to **blank** if the **Digit Treatment** field is set to **insert** and the **Digits** field contains a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table.

Valid entries	Usage
1 to **18**	Enter the number of digits that the far-end server sends for an incoming connection. If your end is absorbing digits on this trunk group, the entry in this field must be larger than the entry in the **Digits** field.
	If you leave this field blank, you cannot administer digit absorption.

Extended Loop Range

This field appears only for a **DID** trunk group and is used only with the TN459A circuit pack.

Valid entries	Usage
y/n	Enter **y** or **n** depending on the distance between the central office and your server. If greater than the required distance, then the field should be **y**.

Format

This field appears if the **Send Calling Number** field is **y** or **r** or the **Send Connected Number** field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. **Unknown** indicates the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown. **Private** indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the **Numbering - Private Format** screen. An entry of **unk-pvt** also determines the **Type of Number** from the **Numbering - Private Format** screen, but the **Numbering Plan Indicator** is unknown.

Group Type

Displays the type of trunk group selected for this field on page 1 of the **Trunk Group** screen. For details, see the field description for the page 1 **Group Type** field.

Incoming Calling Number - Delete

The Incoming **Calling Number - Delete**, **Insert**, and **Format** fields are the administrable fields for the Calling Line Identification Prefix feature. They appear when the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
1 to **15**, **all**, or blank	Enter the number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.

Incoming Calling Number - Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The **Numbering Format** field on page 2 of this screen applies to calls originated from this server running Communication Manager.

If this field is blank, Avaya Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the **Format** field is blank in this case, the value defaults to **pub-unk**.

If the **Format** field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to "unk-unk" encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

Note:

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Incoming Calling Number - Insert

Valid entries	Usage
Enter up to 15 characters (0 to 9), **all**, or blank	Enter up to specific digits, if any, to add to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. Usually, you should match what your central office provides. This field appears when **Group Type** is **Access**, **APLT**, **DID**, **DIOD**, **DMI-BOS**, **FX**, **RLT**, **Tandem**, or **WATS**. It also appears for **Tie** trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touchtone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. Also, if you're using the Inband ANI feature, enter **tone**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone telephones. Though the tone entry supports rotary dialing as well, it's inefficient to reserve touch tone registers for calls that don't use DTMF.
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the **System Parameters Customer-Options (Optional Features)** screen in order for you to enter **mf** here. You cannot enter **mf** if the **Used for DCS** field (field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **mf**.

Incoming Dial Tone

Indicates whether or not your server running Communication Manager will give dial tone in response to far-end seizures of the trunk group.

Valid entries	Usage
y	Enter **y** if the incoming trunk group transmits digits. For example, you would enter **y** for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Enter **n** for trunks that aren't sending digits, such as tandem or incoming CO trunks.

Incoming Rotary Timeout (sec)

Call setup at central offices that still use older switching equipment, such as step-by-step technology, is considerably longer then at central offices with more modern servers. If you're receiving digits with incoming calls from a central office that uses less efficient switching technology, your server needs to allow more time to ensure it receives all the incoming digits. When the **Incoming Dial Type** field is **rotary**, use this field to set the maximum time your end will wait to receive all incoming digits from the far-end switch.

Valid entries	Usage
5 to **99** or blank	If the system is connected to a step-by-step central office, or any CO using older switching technology, enter at least **18** seconds; if not, enter at least **5** seconds.

Line Length

This field appears only when the **Group Type** field is **tie** and the **Trunk Signaling Type** field is **tge**, **tgi**, or **tgu**.

Valid entries	Usage
short	Indicate the line length.
long	

Note:
Unless one or more trunk members are administered, the administered value is not saved when you submit the screen (press **Enter**).

Outgoing Dial Type

This field sets the method used to transmit digits for an outgoing call. Usually, you should match what your central office provides. This field appears for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. It also appears for Tie trunk groups when the **Trunk Signaling Type** field is blank, **cont**, or **dis**.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touchtone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone telephones. If you have a full touch tone system internally and a connection to a central office that only supports rotary dialing, for example, it would be appropriate to enter **rotary**.
r1mf	Enter **r1mf** for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter **r1mf** to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. **Group type** field must be set to **co**.
mf	Enter **mf** if the **Trunk Signaling Type** field is blank. The **Multifrequency Signaling** field must be enabled on the **System Parameters Customer-Options (Optional Features)** screen in order for you to enter **mf** here. You cannot enter **mf** if the **Used for DCS** field (Field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
automatic	Enter **automatic** for tie trunks if the **Trunk Signaling Type** field is blank. This provides "cut-through" operation to outgoing callers who dial a trunk access code, connecting them directly to central office dial tone and bypassing any toll restrictions administered on Communication Manager.

Preferred Minimum Session Refresh Interval (sec)

Appears when **Group Type** is **sip**, and **SCCAN** is **n**. This field sets the session refresh timer value of an SES session for non-SCCAN applications. The timer starts once an SES session established. Avaya Communication Manager then sends a session refresh request as a Re-INVITE or UPDATE after every timer interval. In this way, an ongoing session is maintained. If a session refresh request is not received before the interval passes, the session terminates.

Valid entries	Usage
90 to **1800**	Administer the desired number of seconds for the session refresh interval. Default is **600** seconds.

Receive Answer Supervision

Use this field to specify whether the network provides answer supervision for a trunk group.

For Outbound Call Management applications, set this field to **y** for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Valid entries	Usage
y	Enter **y** if the network provides answer supervision. Set the **Answer Supervision Timeout** field to **0**.
n	Enter **n** if the network does not provide answer supervision, and set the **Answer Supervision Timeout** field. Also enter **n** for incoming trunk groups.

Note:

When you set this field to **y**, the **Outgoing End of Dial (sec)** field is not displayed. The firmware timeout on circuit packs controlled by the **Outgoing End of Dial (sec)** field is automatically set to **0**.

Receive Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** if Communication Manager will receive a release acknowledgment in response to a forward or backward release signal.

Redirect on OPTIM failure

This field is a timer that determines how long to wait for OPTIM to intercede before the call is redirected. Redirect on OPTIM failure is sometimes known as ROOF.

Valid entries	Usage
250 to **32000 milliseconds**	See Off-PBX documentation for details on this field.

SCCAN

This field appears when the **Group Type** field is **sip** and **Enhanced EC500** on the **System Parameters Customer-Options (Optional Features)** screen is set to **y**. When this field is set to **y**, the non-SCCAN-associated fields are hidden.

Valid entries	Usage
y/n	Enter **y** to indicate that this trunk group provides support for incoming SCCAN calls. Default is **n**.

Send Answer Supervision

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager signal the calling server when an incoming call is answered. You can only set this field to **y** if the **Direction** field is **incoming** or **two-way**.

Send Release Ack

This field appears when the **Trunk Signaling Type** field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** to send a release acknowledgment in response to a forward or backward release signal.

Sig Bit Inversion

When transmission facilities use bit-oriented signaling (such as CAS), 2 bits are used to transmit seizure and release signals for calls. Called the A-bit and the B-bit, their meaning can vary. For example, in the A-bit a "1" might mean on-hook and a "0" might mean off-hook. The entry in the **Country Protocol** field on the **DS1 Circuit Pack** screen sets the default meaning of these bits.

For trunk ports on TN2242 and TN464B and later circuit packs, this field allows you to invert the A- and B-bits as necessary so that the far-end server/switch can understand seizure and release signals from Communication Manager. If the far-end server, such as a central office, on this trunk group interprets the A- and B-bits differently from the default, you might need to invert one or both bits — to change "1" to "0" and vice-versa in the A-bit, for example.

Valid entries	Usage
A	For the TN464B and later circuit packs, indicate which bits, if any, should be inverted.
B	
A&B	
none	
A and **none**	For the Japanese 2Mbit trunk circuit pack, indicate which bits, if any, should be inverted.

Supplementary Service Protocol

Appears only when **Group Type** is **ISDN**.

Valid entries	Usage
a	Allows ASAI Flexible Billing. AT&T, Bellcore, Nortel. When the **Country Code** field on the **DS1** screen is **1A**, SSA selects AT&T custom supplementary services. When the **Country Code** field on the **DS1** screen is **1B**, SSA selects Bellcore Supplementary Services. When the **Country Code** field on the **DS1** screen is **1C**, SSA selects Nortel Proprietary Supplementary Services.
b	QSIG; also used for SBS signaling trunk groups when full QSIG functionality is needed.
c	ETSI Use c protocol for Network Call Deflection.
d	ECMA QSIG
e	Allows ASAI Flexible Billing. Allows DCS with rerouting. **DCS with Rerouting** must be **y**, and the **Used for DCS** field on the **Trunk Group** screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **ISDN-PRI** or **ISDN-BRI** field is **y** or the **Used for DCS** field is **y**. Use **g** protocol for Network Call Transfer.

Synchronization

This field determines whether the trunk group will use synchronous or asynchronous communications. This field appears if:

- the **Group Type** field is **dmi-bos** or **isdn**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats** and the **Comm Type** field is **avd** or **rbavd**, or
- the **Group Type** field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**, the **Comm Type** field is **data**, and the **ISDN-PRI** field or the **ISDN-BRI Trunks** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen.

Valid entries	Usage
async	Do not change this field without the assistance of Avaya or your network service provider.
sync	

Trunk Gain

This field specifies the amplification applied to the trunks in this group. With the values of the **Trunk Termination** and **Country** fields, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO and DID circuit packs are set automatically to high

Valid entries	Usage
high	Enter **high** if users complain of low volume.
low	Enter **low** if users complain of squeal or feedback.

Trunk Hunt

Avaya Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed

.

Valid entries	Usage
ascend	Enter to enable a linear trunk hunt search from the lowest to highest numbered channels.
cyclical	Enter to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group.
descend	Enter for a linear trunk hunt search from the highest to lowest numbered channels.

Note:

The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the **Wideband Support** field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Trunk Termination

This field adjusts the impedance of the trunk group for optimal transmission quality. Check with your service provider if you're not sure of the distance to your central office.

Valid entries	Usage
600ohm	Enter **600ohm** when the distance to the central office or the server at the other end of the trunk is less than 3,000 feet.
rc	Enter **rc** when the distance to the central office or the server at the other end of the trunk is more than 3,000 feet.

Trunk Type

Use this field to control the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and *must* match the corresponding setting on the far-end server or switch. This field appears for CO, DID, FX, and WATS trunk groups.

Procedures in Tips for working with trunk groups on page 477 give specific suggestions for signaling to use with different types of trunk groups.

Valid entries	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, try to use loop-start signaling only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto **auto/delay** **auto/immed** **auto/wink**	The term before the slash tells Communication Manager how and when it will receive incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it’s ready to receive the digits. • wink — The sending server running Communication Manager does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it’s ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don’t know what type of connection they’re using. To use these entries, the **Country** field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Unicode Name

Appears when the **Group Type** field is **sip**. The value for this field is only examined for calls to SIP Enablement Services (SES) stations over an SES trunk group and is used to determine whether to send Name1 (legacy name) or Name2 (Unicode name). Note that Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Valid entries	Usage
y	Enter **y** to indicate the use of Unicode Name.
n	Type **n** to use the table with UTF-8 format and so might contain Asian language names. Note that fifteen UTF-8 characters can take up to 45 bytes. Also, legacy names support Roman, Cyrillic, Ukrainian, and Katakana characters.

Wink Timer (msec)

This field appears when one of the "**wink**" options is entered in the **Trunk Type** field. This field allows you to reduce the risk of glare by controlling part of call setup. Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds. While some circuit packs are hard-coded to allow the full 5 seconds in both cases, other circuit packs allow you reduce the allowed start time and duration, thus reducing the window in which glare could occur.

Unlike other fields on this screen, the **Wink Timer** field therefore controls 2 different variables. What your entry does depends on the outgoing value in the **Trunk Type (in/out)** field.

Setting of the Trunk Type (in/out) field	What the Wink Timer field sets
…/wink	Maximum duration of the wink signal (wait-for-wink-to-end)
…/delay	Maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start)

Field descriptions for page 3

The figure below is only an example, and is intended to show most of the fields that might appear on page 3 of the **Trunk Group** screen. This example might not show all fields, or might show fields that normally do not appear together; it is not intended to reflect a specific trunk group type. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Figure 581: Trunk Group screen (page 3)

```
add trunk-group next                                            Page 3 of x
                              TRUNK FEATURES
           ACA Assignment? _          Measured: ____      Wideband Support? _
 Long Holding Time(hours): _                            Maintenance Tests? _
  Short Holding Time(sec): _  Data Restriction? _     NCA-TSC Trunk Member: _
  Short Holding Threshold: __        Send Name: _      Send Calling Number: _
             Used for DCS? _                          Send EMU Visitor CPN?

Suppress # Outpulsing? _   Numbering Format: _______
Outgoing Channel ID Encoding: _________     UUI IE Treatment: ______
                                        Maximum Size of UUI IE Contents: ___
                                              Replace Restricted Numbers? _
                                             Replace Unavailable Numbers? _
                                                   Send Connected Number? _
                                               Hold/Unhold Notifications? _

        Send UUI IE? _
          Send UCID? _                BRS Reply-best DISC Cause Value: __
                                                   Ds1 Echo Cancellation? _

                                         US NI Delayed Calling Name Update? _
Show ANSWERED BY on Display? y
                          Network (Japan) Needs Connect Before Disconnect? _

Time (sec) to Drop Call on No Answer:_
Outgoing ANI:_
R2 MFC Signaling:_

DSN Term? n       Precedence Incoming _______   Precedence Outgoing _______
Used Only for Paging?                          Voice Paging Timeout (sec)? 10
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Abandoned Call Search

Use this field when the **Trunk Type** field is **ground-start**. Abandoned Call Search is designed to work with analog ground-start CO trunks that do not provide disconnect supervision. Your central office must support Abandoned Call Search for the feature to work properly. If your central office provides disconnect supervision, you do not need to use the Abandoned Call Search feature.

Valid entries	Usage
y/n	Enter **y** if this trunk group will conduct an Abandoned Call Search to identify ghost calls.

ACA Assignment

Valid entries	Usage
y/n	Enter **y** if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If **y** is entered, complete the **Long Holding Time**, **Short Holding Time**, and **Short Holding Threshold** fields.

Charge Conversion

Avaya Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. If there is no value in this field, Avaya Communication Manager displays the number of charge units without converting it to currency. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to **64**, **500**	Enter the value of a charge unit in terms of the currency you use.

Charge Type

Entries in this field are text strings you use to describe charges related to a telephone call. This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 7 characters (embedded spaces count as characters) blank	Enter the words or characters you want to appear on telephone displays after the charge amount. Most likely you will use either the currency symbol or the charge type, bur not both.

Connected to CO

This field appears when the **Group Type** field is **tie**.

Valid entries	Usage
y/n	Enter **y** to allow overlap sending to a Central Office.

Currency Symbol

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**.

Valid entries	Usage
1 to 3 characters (leading and embedded spaces count as characters) or blank	Enter the symbol you want to appear on telephone displays before the charge amount.

Data Restriction

If **y**, whisper page is denied on this trunk.

Valid entries	Usage
y/n	Enter **y** to prevent features from generating tones on a data call that would cause erroneous data transmission.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the **Direction** field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the **Charge Advice** field is *not* **none**. Chose the appropriate representation for a decimal point as it will appear on telephone displays. Entering **comma** or **period** in this field divides the charge value by 100.

Note:
If the received charge contains no decimals, no decimal point is displayed (i.e., the administered decimal point is ignored for charge information received with no decimals). On an upgrade from a QSIG trunk group with the **Decimal Point** field administered as **none**, the field defaults to **period**.

Valid entries	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point.
none	No decimal point is displayed.

DS1 Echo Cancellation

Reduces voice call echo.

Note:
Changes to the **DS1 Echo Cancellation** field do not take effect until one of the following occurs:

- Port is busied-out/released
- Trunk group is busied-out/released
- SAT command test trunk group is performed
- Periodic maintenance runs

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation on a per port basis.

DSN Term

Valid entries	Usage
y/n	Use the **DSN Term** field to identify the trunk group as a DSN termination telephone. The default is **n**.

Format

The screen displays this field if the **Send Calling Number** field is either **y** or **r**, or the **Send Connected Number** field is either **y** or **r**. The **Numbering Format** field specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**.

Valid entries	Usage
Public	Indicates that the number plan according to CCITT Recommendation E.164 is used and that the **Type of Number** is national. This is the default entry for SIP Enablement Services (SES) trunks.
Unknown	Indicates that the **Numbering Plan Indicator** is unknown and that the **Type of Number** is unknown.
Private	Indicates the **Numbering Plan Indicator** is PNP and the **Type of Number** is determined from the **Numbering - Private Format** screen.
unk-pvt	Also determines the **Type of Number** from the **Numbering - Private Format** screen, but the **Numbering Plan Indicator** is unknown.

Glare Handling

This field determines what Communication Manager will do when glare occurs. This field appears when the **Direction** field is **two-way** and the outgoing side of the **Trunk Type** field is either **…/wink** or **…/delay**.

If you enter **control** or **backoff**, and ports for the trunk group are not capable of detecting glare, warnings are generated. The following circuit packs can detect glare: TN767 (all releases), TN760C (or later releases), and TN464C (or later releases).

Valid entries	Usage
control	Communication Manager will seize the trunk and proceed with call setup. The other switch will find another trunk.
backoff	The other server or switch will seize the trunk and proceed with call setup. Your server/switch will find another trunk.
none	

Hold/Unhold Notifications

Appears only when the **Group Type** field is **isdn**. Use this field to indicate whether or not hold/unhold messages are sent over the isdn trunk when a user places a call on hold/unhold.

Valid entries	Usage
y/n	Enter **y** to enable sending of hold/unhold messages over this isdn trunk. Default is **y**.

Incoming Tone (DTMF) ANI

This field appears only when the **Incoming Dial Type** field is **tone**. Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the INTUITY AUDIX and ASAI interfaces, and displayed on the telephone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Valid entries	Usage
ANI*DNIS	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The telephone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The telephone displays Call from 555-3800.
no	

Internal Alert

Valid entries	Usage
y/n	Enter **y** if internal ringing and coverage will be used for incoming calls.

Long Holding Time (hours)

This field appears only when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **10**	Enter the length of time (in hours) that the system will consider as being a long holding time. If you enter **0**, the system will not consider long holding calls.

Maintenance Tests

Appears when the **Group Type** field is **aplt**, **isdn**, **sip**, or **tie**.

Valid entries	Usage
y/n	Enter **y** if hourly maintenance tests will be made on this trunk group. One or more trunk members must be administered for this entry to be saved.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the **BCMS (Basic)** or the **VuStats** field is **y** on the **System Parameters Customer-Options (Optional Features)** screen. If the **ATM** field is set to **y** on the **System Parameters Customer-Options (Optional Features)** screen, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when **ATM** is **y**, **none** appears.

Valid entries	Usage
internal	Enter **internal** if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.
external	Enter **external** to send the data to the CMS.
both	Enter **both** to collect data internally and to send it to the CMS.
none	Enter **none** if trunk group measurement reports are not required.

MF Tariff Free

This field appears for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the **Incoming Dial Type** field is **mf** or the **Group Type** field is **tie**, the **Trunk Signaling Type** field is blank, **cont**, or **dis**, and the **Incoming Dial Type** field is **mf**.

Valid entries	Usage
y/n	Enter **y** to make Communication Manager generate an MFC Tariff-Free Backward Signal (administered on the **Multifrequency-Signaling-Related-System-Parameters** screen) during call setup instead of the "free" signal. This aids CO billing.

Network Call Redirection

Valid entries	Usage
deflect	Use to allow Network Call Deflection.
ANSI-transfer	Use to allow Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Use to allow Network Call Transfer for MCI DMS 250 switches.
telcordia-tbct	Use to allow Network Call Transfer for Lucent 5ESS or Nortel DMS100 switches.

Outgoing ANI

If this trunk group is used for an outgoing call with ANI, the entry in this field overrides the normal ANI. The ANI is sent exactly as administered, except for the normal truncation to 7 digits for Russian ANI. This ANI override works both for calls originated in Avaya Communication Manager and calls tandemed through it. This field appears for CO, DIOD, FX, and WATS trunk groups.

Valid entries	Usage
1 to 15 digits	Enter the digit string to be sent in place of normal ANI.
blank	Leave this field blank to allow ANI to work normally.

Path Replacement Method

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, the **Path Replacement with Retention** is **n**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the **Path Replacement with Retention** field is **y**. Path replacement fails (and the original trunk group is retained) if the **Path Replacement with Retention** field is **n**.

Path Replacement with Retention

Appears when the following fields are set on the **Trunk Group** screen: trunk **Group Type** is **ISDN**, **Supplementary Service Protocol** is **b** or **e**, and the **Supplementary Services with Rerouting** field or the **DCS with Rerouting** field on the System Parameters Customer-Options (Optional Features) screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the **Path Replacement Method** field.

PBX ID

Appears when the **Used for DCS** field is **y**. This field identifies the remote switch in the network with which the trunk will communicate on a DCS signaling link.

Valid entries	Usage
1 to **63** or blank	Enter the ID of the switch at the other end of this trunk.

Per Call CPN Blocking Code

For Access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, **#**	Can be used as the first digit

Per Call CPN Unblocking Code

For access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number or blank	
*****, **#**	Can be used as the first digit

Precedence Incoming

The **Precedence Incoming** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is received as digits (rotary pulses).
dtmf (a-d)	Precedence level is received as DTMF signals (touchtones).

Precedence Outgoing

The **Precedence Outgoing** field defines whether the precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as digits (rotary pulses) or as DTMF signals (touchtones). Appears when the **DSN Term** field is **y** and **Group Type** is **tie**.

Valid entries	Usage
digit	Precedence level is sent as digits (rotary pulses).
dtmf (a-d)	Precedence level is sent as DTMF signals (touchtones).

R2 MFC Signaling

Appears when, on the **System Parameters Customer-Options (Optional Features) screen**, **Multinational Locations** is **y**, and on the **Trunk Group** screen, **Outgoing Dial Type** is **mf**.

Valid entries	Usage
1 to **8**	Enter the MFC signaling parameters set used by this trunk group.

Receive Analog Incoming Call ID

15 characters of name and number information associated with an incoming call on analog trunks (ICLID, or incoming call line identification information) is stored and displays. This field appears for CO, DID, and DIOD trunk groups when the **Analog Trunk Incoming Call ID** field on the **System Parameters Customer-Options (Optional Features)** screen is **y** and the **Direction** field is **incoming** or **two-way**.

Valid entries	Usage
Bellcore	Used to collect ICLID information in the U.S.
NTT	Used to collect ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.
V23-Bell	Enter V23-Bell for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Replace Unavailable Numbers

Appears when the **Group Type** field is **isdn** or **sip**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI/PRI, H.323, and SIP Enablement Services (SES) trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Request Category

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
y/n	Enter **y** if Communication Manager should request a call category from the central office.

Seize When Maintenance Busy

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. It indicates whether this server generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end server or switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end server or switch from reporting problems with a trunk that has been removed from service on your end. This field's setting does not affect the behavior of the far-end server or switch; it controls the behavior of your server and defines the expected far-end behavior.

For DIOD trunks using TN464F (or later release) or TN2464, displays only when the **Group Type** field is **diod** and the **Trunk Signaling Type** field is **pulsed**, **cont**, or **dis**.

Valid entries	Usage
near-end	Enter **near-end** if Communication Manager generates an outgoing seizure when a trunk is maintenance busied, but the far-end server or switch does not. The seizure is maintained until the maintenance busyout is released.
far-end	Enter **far-end** if the far-end server or switch generates an outgoing seizure when a trunk is maintenance busied, but this server running Communication Manager does not.
both-ends	Enter **both-ends** if both this server running Communication Manager and the far-end server or switch generate an outgoing seizure when a trunk is maintenance busied.

If a server generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end server or switch, perhaps leading to a far-end maintenance busy out, unless the far-end server or switch is administered to expect this behavior.

If the administered value of this field is either **far-end** or **both-ends**, any abnormally long incoming seizure (including failure to drop from a completed call) is assumed to be the result of a far-end maintenance busy condition. Note that this assumption might be incorrect, since the abnormally long seizure might actually be due to failure of the trunk circuit. Regardless of the cause of the abnormally long seizure, your server running Communication Manager does the following things:

1. Generates a warning alarm indicating that the trunk is assumed to be maintenance busy at the far-end
2. Removes the trunk from service
3. Keeps the trunk out of service until a far-end disconnect is received

Allowable values depend on the entry in the **Direction** field: check the online help in Avaya Communication Manager.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number to be sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the calling number is sent "presentation restricted."

Note:

The **ISDN Numbering - Public/Unknown Format** screen can override the **Send Calling Number** field entry for any administrable block of extensions.

Send Called/Busy/Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **y**. Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Called/Busy/Connected Number** field must be set to **y** in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Send Connected Number

Appears if the **QSIG Value-Added** field on the **Trunk Group** screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the **ISDN Numbering - Public/Unknown Format** screen is accessed to construct the actual number sent, or the **ISDN Numbering - Private** screen (based on the **Numbering Format** field) is used. If the value is **r**, the connected number is sent "presentation restricted." The **Send Connected Number** field must be set to **y** in order for the

Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination on the **DS1 Circuit Pack** screen, you should not administer the **Send Connected Number** field to **r** (restricted) on the **ISDN Trunk Group** screen, as this causes display problems.

The **ISDN Numbering - Public/Unknown Format** screen overrides the **Send Connected Number** field entry for any administrable block of extensions.

Send EMU Visitor CPN

Use this field to control which calling party identification (extension of the primary telephone or extension of the visited telephone) is used when a call is made from a visited telephone. If you want to use the calling party information of the primary telephone, set this field to **n**.

There are areas where public network trunks disallow a call if the calling party information is invalid. In this case, there can be instances where the extension of the primary telephone is considered invalid and the extension of the visited telephone must be used. To use the extension of the visited telephone, set the **Send EMU Visitor CPN** field to **y**. For more information on Enterprise user Mobility, see Setting Up Enterprise Mobility User on page 171.

Valid entries	Usage
y	Sends calling party identification information on the extension of the EMU user’s telephone.
n	Sends calling party identification information on the primary telephone.

Send Name

Appears only when the **Group Type** field is **isdn** or **sip**. Specifies whether the calling/ connected/called/busy party’s administered name is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by Avaya Communication Manager, but will be marked "presentation restricted." This value is valid only if the **Supplementary Service Protocol** field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the **Supplementary Service Protocol** field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted.

Note:

If name information is not administered for the calling station or the connected/ called/busy station, the system sends the extension number in place of the name.

Short Holding Threshold

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk. Enter **0** for no short holding calls.

Short Holding Time (seconds)

This field appears when the **ACA Assignment** field is **y**.

Valid entries	Usage
0 to **160**	Enter the length of time (in seconds) that the system considers as being a short holding time. If **0** is entered, the system will not consider short holding calls.

Show ANSWERED BY on Display

This field appears when the **Group Type** field is **isdn pri/bri** or **sip**. Use this field to administer whether or not the words "ANSWERED BY" are displayed in addition to the connected telephone number on calls over this trunk.

Note:
Based on display language settings for stations, "ANSWERED BY" is translated into and displayed in the appropriate language.

Valid entries	Usage
y	When set to **y**, the words "ANSWERED BY" are displayed in addition to the connected telephone number. This is the default.
n	When set to **n**, only the connected telephone number is displayed. This might be preferred when outgoing calls are over a trunk that might be redirected.

Shuttle

This field appears when the **Group Type** field is **co**, **fx**, or **wats**, the **Country** field is **15**, and the **Outgoing Dial Type** field is **rotary**. It can be administered on TN464D (or later release) or TN2199 circuit packs.

Valid entries	Usage
y/n	Enter **y** to enable MF shuttle signaling.

Signaling Group

This field displays the group number assigned when the group was added.

Start B Signal

This field appears when the **Country** field is **15** and the **Shuttle** field is **y**. Enter **1** to **3** to indicate which B-signal should be used to start a call. The value administered in this field must be coordinated with your central office. See Start Position on page 1725.

Valid entries	Usage
1	Start calls with signal B1 (first digit)
2	Start calls with signal B2 (next digit)
3	Start calls with signal B3 (previous digit)

Start Position

The value administered in this field must be coordinated with your central office. This field appears when the **Country** field is **15** and the **Shuttle** field is **y**.

Valid entries	Usage
1 to **9**	Indicate which digit in the digit string is considered to be the "previously sent" digit (see Start B Signal on page 1725).

Suppress # Outpulsing

Valid entries	Usage
y/n	Enter **y** if end-to-end signaling begins with (and includes) "#". The final "#" is suppressed in cases where the system would normally outpulse it. This field should be **y** when the Central Office (for example, rotary) or any other facility treats "#" as an error.

Time (sec) to Drop Call on No Answer

This field appears if the **Group Type** field is **co** or **diod** and the **Outgoing Dial Type** field is **mf**, or if the **Group Type** field is **co**, **diod**, **fx**, or **wats** and the **Country** field is **15**.

Valid entries	Usage
0 to **1200**	Enter the duration (in seconds) Communication Manager should wait for outgoing calls to be answered. If the call is not answered in the specified number of seconds, the call drops. If this field is **0**, the timer is not set and no calls drop.

Used for DCS

Valid entries	Usage
y/n	Enter **y** if this trunk group will send and receive messages on a DCS signaling link.

Note:

This field cannot be activated if the trunk group number is greater than 255 or if the Trunk Access code is more than 3-digits long.

If this field is **y**, you can administer ISDN-BRI trunk groups unless the **DCS Signaling** field is **d-chan**.

Used Only for Paging

This field appears when the **Group Type** field on the **Trunk Group** screen is **wats**, and the **Port Network Support** field on the **System Parameters Customer Options** screen is **n**.

Valid entries	Usage
y/n	Enter **y** to designate this trunk for paging use. Default is **n**.

Voice Paging Timeout (sec)

This field appears when **Used Only for Paging** is **y**.

Valid entries	Usage
10 to **600**	Enter the number of seconds before a paged trunk call drops. Default is **10**.

Field descriptions for Administrable Timers page

This screen might not appear for all trunk group types. The figure below is only an example, and is intended to show most of the fields that might appear on this page of the **Trunk Group** screen. This example might not show all fields, or might show fields that normally do not appear together. Your own screen might vary from this example according to specific field and system settings. The list of field descriptions that follows the figure is in alphabetical order for quick reference. This list is intended to be comprehensive, and might include information on fields that are not shown in the example. The field descriptions identify fields that are specific to particular trunk group types.

Note:

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Figure 582: Administrable Timers for Trunk Group screen

```
add trunk-group next                                          Page  3 of  x
                           ADMINISTRABLE TIMERS
               Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                                Outgoing Dial Guard(msec): _____
  Incoming Glare Guard(msec): _____            Outgoing Glare Guard(msec): _____

                                   Outgoing Rotary Dial Interdigit (msec): ____
       Ringing Monitor(msec): _____                   Incoming Seizure(msec): ____
   Outgoing End of Dial(sec): ___          Outgoing Seizure Response(sec): ___
 Programmed Dial Pause(msec): _____            Disconnect Signal Error(sec): ___
          Flash Length(msec): ____
                                                     Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ____     Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Answer Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 and TN2199 ports receive this timer.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of the answer signal pulse.

Busy Tone Disconnect

The field appears when **Enable Busy Tone Disconnect for Analog loop-start Trunks** is **y** on the **System Parameters Country-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow Communication Manager to recognize a busy tone signal as a disconnect on this trunk group.

Cama Outgoing Dial Guard (msec)

This field appears when **Group Type** is **cama** (the trunk group type used for emergency 911 service).

Valid entries	Usage
25 to **6375** in increments of 25	Enter the minimum interval between the seizure acknowledgment on the receiving server or switch and the outpulsing of digits by this server.

Cama Wink Start Time (msec)

This field appears when **Group Type** is **cama**.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Disconnect Signal Error (sec)

This field appears for ground-start trunk groups.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager will wait to receive a disconnect signal from the far-end after the local party (a telephone or tie trunk) goes on-hook. If the timer expires, Communication Manager assumes a disconnect failure and takes appropriate action, such as creating an error message.

Flash Length (msec)

This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), (TN753 if Country is **23**), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of a flash signal generated toward the central office.

Glare

This field is only administrable if the **Trunk Type** field is **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **100** in increments of 10	Enter the minimum acceptable interval (in msec) between the moment your server running Communication Manager sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Incoming Dial Guard (msec)

Valid entries	Usage
10 to **2550** in increments of 10	Enter the minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. Communication Manager will not accept digits before this timer expires. This timer is never sent to TN429 ports.

Incoming Disconnect (msec)

The field appears only when the **Direction** field is **incoming** or **two-way** and the **Trunk Type** field is either blank or **cont**.

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an incoming call. Avaya Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use **600**.

Incoming Disconnect Send (msec)

This field is only administrable if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the backward release signal your server running Communication Manager sends at the end of an incoming call.

Incoming Glare Guard (msec)

This field only appears when the trunk group **Direction** field is **two-way**.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the trunk is released.

Incoming Incomplete Dial Alarm (sec)

Only the TN436 (all), TN459 (all), TN464C (or later), TN767, TN2140, TN2146, TN2184, TN2199, and TN2242 circuit packs use this timer.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum acceptable interval between an incoming seizure and receipt of all digits. Intervals greater than this limit generate an inline error.

Incoming Partial Dial (sec)

This timer appears only if the **Incoming Dial Type** field is **rotary**.

Valid entries	Usage
5 to **255** in increments of 1	Enter the maximum time allowed between incoming rotary digits.

Note:
This timer is never sent to TN429 ports.

Incoming Seizure (msec)

This field appears when the **Direction** field is **incoming** or **two-way**, and, when applicable, the **Trunk Type** field is **cont**. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the shortest incoming seizure signal your server running Communication Manager can recognize. For ICLID, set this field to 120. The field cannot be blank.

Normal Outgoing Seize Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the signal your server running Communication Manager sends for an outgoing seizure.

Outgoing Dial Guard (msec)

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. This field cannot be blank. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after they lift the receiver, but before the timer expires, is not outpulsed until the timer expires.

Outgoing Disconnect (msec)

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an outgoing call. Communication Manager will not recognize shorter disconnect signals. This field cannot be blank. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. For Brazil pulsed E&M signaling, use **600**.

Outgoing Disconnect Send (msec)

This field is administrable only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
100 to **9900** in increments of 100	Enter the duration of the forward release signal your server running Communication Manager sends at the end of outgoing calls.

Outgoing End of Dial (sec)

This field controls firmware answer supervision timers on circuit packs that have them. It appears when the **Direction** field is **outgoing** or **two-way** and the **Receive Answer Supervision** field is **n**. If the **Receive Answer Supervision** field is **y**, this field does not appear and the firmware timer on the appropriate circuit pack is automatically disabled.

Note:

The **Answer Supervision Timeout** field on the **Trunk Group** screen provides timed answer supervision for circuit packs without administrable timers. Since trunk groups might contain ports on more than one circuit pack, it's possible you might need to use both timers with the same trunk group. If so, set the **Outgoing End of Dial** field and the **Answer Supervision Timeout** field to the same value.

During a cut-through operation, timing begins after Communication Manager sends each outgoing digit and ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision.

Valid entries	Usage
1 to **254** in increments of 1	Enter the maximum time, in seconds, that Communication Manager will wait to receive answer supervision for outgoing calls on the ports controlled by firmware timers. For Brazil pulsed E&M signaling, use **40**.

Outgoing Glare Guard (msec)

This field only appears for **outgoing** and **two-way** trunk groups.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the outgoing trunk is released.

Outgoing Last Digit (sec)

This field is only administrable if the **Trunk Type** field is **dis** or **cont** and the trunk group **Direction** field is **two-way** or **outgoing**. Only TN497 and TN2140 ports receive this timer.

Valid entries	Usage
1 to **40** in increments of 1	Enter the maximum time that Communication Manager will wait for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Outgoing Rotary Dial Interdigit (msec)

This field only appears when:

1. the trunk group **Group Type** field is **access**, **aplt**, **co**, **diod**, **dmi-bos**, **fx**, **rlt**, **tandem**, or **wats** and the **Outgoing Dial Type** field is **rotary**.
2. the **Group Type** field is **tie**, the **Trunk Type** field is blank, **cont**, or **dis**, and the **Outgoing Dial Type** field is **rotary**.
3. the **Group Type** field is **tie**, and the **Trunk Type** field is **tge**, **tgi**, or **tru** (the **Outgoing Dial Type** field does not appear but is implied to be **rotary**).

Valid entries	Usage
150 to **2550** in increments of 10	Enter the minimum time between outpulsed digits on outgoing rotary trunks.

Outgoing Seizure (msec)

Appears when the **Country** field is **15**, the **Direction** field is **outgoing** or **two-way**, and the **Trunk Type** field is **2-wire-ac**, **2-wire-dc**, or **3-wire**. This timer is sent only to the TN2199 circuit pack.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the outgoing seizure signal.

Outgoing Seizure Response (sec)

This timer is sent to the TN438B, TN439, TN447, TN458, TN464B (or later), TN465B (or later), TN767, TN2140, TN2147, TN2184, TN2199, and TN2242 circuit packs.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that Communication Manager should wait after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received in this time, a seizure failure response is uplinked. For Brazil pulsed E&M signaling, use **255**.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

Release Ack Send (msec)

After your server running Communication Manager receives a forward release signal, it must send a forward release acknowledgment signal. This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the signal your server running Communication Manager sends for a forward release acknowledgment.

Ringing Monitor (msec)

This timer is sent to TN464C (or later), TN767, TN438 (all), TN447, TN465 (all), TN2138, TN2147, TN2184, and TN2199 CO circuit packs.

Valid entries	Usage
200 to **51000** in increments of 200	Enter the minimum time Communication Manager requires to determine if a trunk disconnects. The field cannot be blank. If the ringing signal disappears for a duration longer than the time specified in this field, Communication Manager assumes the call has been disconnected.

Seize Ack Delay (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **120** in increments of 10	Enter the maximum interval your server running Communication Manager will wait after receipt of an incoming seizure to send seizure acknowledgment.

Seize Ack Send (msec)

This field appears only if the **Trunk Type** field is **dis** and the trunk group **Direction** field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the seizure acknowledgment signal your server running Communication Manager sends in response to an incoming seizure.

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

The field appears only for a co, fx, or wats trunk group.

Valid entries	Usage
y/n	Enter **y** if you want to send the incoming disconnect and outgoing disconnect timer values to the trunk group ports that are on a TN465 board.

END TO END SIGNALING

Pause (msec)

THis field is administrable only if the **Trunk Type** field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid telephone.

Tone (msec)

This field appears only if the **Trunk Type** field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid telephone is pressed.

OUTPULSING INFORMATION

Break (msec)

Valid entries	Usage
Enter the duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If PPS field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5.	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

Frequency

This field identifies the PPM pulse frequency, or frequencies, sent by the public network. It appears if the **Direction** field is **outgoing** or **two-way** and **PPM** is **y**. Circuit packs can detect up to three different frequencies, (12kHz, 16kHz, and 50Hz), plus two frequency combinations, (50Hz/12kHz and 50Hz/16kHz). This field controls TN465B, TN2138, and TN2184 circuit packs.

Valid entries	Usage
12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **12k** is administered, the circuit pack will be set to detect 12kHz.
16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **16k** is administered, the circuit pack will be set to detect 16kHz.
	1 of 2

Valid entries	Usage
50	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50** is administered, the circuit pack will be set to detect 16kHz.
50/12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/12k** is administered, the circuit pack will be set to detect 12kHz.
50/16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/16k** is administered, the circuit pack will be set to detect 16kHz.
	2 of 2

Make (msec)

Valid entries	Usage
Enter the duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.	
20 to **80** in increments of 5	If the **PPS** field is **10**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 100.
10 to **40** in increments of 5	If the **PPS** field is **20**, the sum of the **Make (msec)** and **Break (msec)** fields must equal 50.

PPM

For CO, DIOD, FX, PCOL, and WATS trunks. This field appears when the **Direction** field is **outgoing** or **two-way**.

Valid entries	Usage
y/n	Enter **y** if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. If this field is **y**, the **Frequency** field appears.

PPS

Valid entries	Usage
10 **20**	Enter the rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group. **Note:** The TN439, TN458, TN497, TN747Bv12 (or later), and TN767 circuit packs send rotary pulses at 10 pps only.

Field descriptions for ATMS Thresholds page

This screen appears when the **Direction** field on page 1 is **outgoing** or **two-way** and the **ATMS** field is **y** on the **Feature-Related System Parameters** screen.

The figure below shows a common configuration for the **ATMS Thresholds** page of the **Trunk Group** screen. This screen is only an example, and the fields shown below might change or disappear according to specific field settings.

Note:

This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Figure 583: ATMS Thresholds screen

```
add trunk-group next                                                  Page 4 of x
                              ATMS THRESHOLDS
TTL Type: _________                 Far End Test No: __________________
  TTL Vendor: ______________            TTL Contact: ______________________
Trunk Vendor: ____________            Trunk Contact: ___________________
Trunk Length: _____
                         MARGINAL         UNACCEPTABLE
                         Min    Max          Min    Max
           1004 Hz Loss: __     __           __     __

                         -Dev  +Dev          -Dev   +Dev
            404 Hz Loss:  _     _             _      _
           2804 Hz Loss:  _     _             _      _

Maximum C Message Noise:    __                   __
Maximum C Notched Noise:    __                   __
         Minimum SRL-HI:    __                   __
         Minimum SRL-LO:    __                   __
            Minimum ERL:    __                   __

      Allow ATMS Busyout, Error Logging and Alarming? _
  Maximum Percentage of Trunks Which Can Be Removed from Service by ATMS: __
```

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Far-End Test No.

Valid entries	Usage
1 to 16 digits	Enter the access number dialed to reach the terminating test line (TTL).

Trunk Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the trunk vendor who can be contacted in the event of problems with the trunks.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information might be useful,

Valid entries	Usage
Use this field to record the length of the trunk group in kilometers or miles.	
0 to 4 digits followed by **k**	Shows the length in kilometers.
0 to 4 digits followed **m**	Shows the length in miles.

Trunk Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor providing service over this trunk group (the company to notify in the event of problems with the trunks in this trunk group).

TTL Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line.

TTL Type

Specifies the type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and thus which threshold values need to be administered.

Valid entries	Usage
105-w-rl	105 with return loss
105-wo-rl	105 without return loss
high-lts	high-level tone source
low-lts	low-level tone source
100	100 type
102	102 type

The following table explains the differences between types of terminating test lines:

Type TTL	Description	Example
105-w-rl	Full range of 18 measurements or some defaults for return loss used (56A)	TN771B, ZLC12 and SN261B circuit packs and new 56A mini-responder
105-wo-rl	Cannot return default values for far-end return loss	Older 56A mini-responder
high-level-tone	Sends a fixed sequence of tones at 0 dBm	ZLC12 and SN261B circuit packs
low-level-tone	Sends a fixed sequence of tones at -16dBm	SN261B circuit pack
100	Up to 5 measurements that sends a 1004 Hz tone then a quiet termination	
102	One measurement that sends a 1004 Hz tone	

The far-end server or switch containing the TTL might be any of the following:

- System 85 R2 switch, equipped with the Maintenance/Test Board (TN771B)
- System 75 R1V2 and beyond, all of which contain the circuitry required to perform the TTL function
- System 85 R2 switch, equipped with the Analog/Digital Facility Test Circuit (ADFTC, SN261)

- DIMENSION FP8, equipped with the Analog Facility Test Circuit (AFTC, ZLC-12)
- Central Office switches, equipped with various TTL equipment that provide 100, 102, or 105 test line capabilities (56A)

Other vendors' switches might be supported if compatible test lines are provided by these switches.

Four different versions of the ATMS Threshold Administration page can appear depending upon the measurements allowed by the TTL type selected. The four possibilities are:

1. 105-w-rl and 105-wo-rl - All thresholds appear.
2. high-lts and low-lts - All thresholds (except maximum C-notched noise) appear.
3. 100 - All thresholds (except maximum c-notched noise, 404Hz loss, and 2804 Hz loss) appear.
4. 102 - Only 1004 Hz loss threshold appears.

TTL Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor supplying the terminating test line (TTL).

MARGINAL / UNACCEPTABLE

Allow ATMS Busyout, Error Logging and Alarming

Valid entries	Usage
y/n	Enter **y** to allow ATMS error logging and alarming (subject to filtering depending on the service organization used to deal with alarms).

Marginal Threshold - -Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Marginal Threshold - Min -1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Appears when the **Allow ATMS Busyout**, **Error Logging and Alarming** field is **y**.

Valid entries	Usage
0, **25**, **50**, **75**, **100**	Enter the highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Unacceptable Threshold - -Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise interference in dBmC above reference noise terminating on a telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Field descriptions for Protocol Variations page

This screen appears only when the **Group Type** is **sip**.

Figure 584: Protocol Variations screen

```
add trunk-group next                                         Page 3 of x
                          PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend '+' to Calling Number? n
Send Transferring Party Information? n
Network Call Redirection? n
Telephone Event Payload Type: 127
```

Mark Users as Phone

Valid entries	Usage
y/n	When this field is set to **y**, URIs in call control signaling messages originated at the gateway are encoded with the "user=phone" parameter. No subscription messages are encoded with the "user=phone" parameter, even when the field is set to **y**. Default is **n**. **Note:** Do not change the default of **n** for this field unless you are sure that every recipient of SIP Enablement Services (SES) calls using this trunk can accept and properly interpret the optional "user=phone" parameter. Enterprise users without support for "user=phone" in their SIP Enablement Services (SES) endpoints will experience adverse effects, including rejected calls.

Network Call Redirection

Use this field to control which trunk groups the Network Call Redirection (NCR) service is signaled over. NCR only works on trunk groups connected to Service Providers that support NCR.

Valid entries	Usage
y/n	Enter **y** to specify this trunk group for Network Call Redirection. Default is **n**.

Prepend "+" to Calling Number

Appears when the **Group Type** is **sip**. When set to **y**, the calling party number in the header of the SIP message is prepended with a plus sign (+).

Valid entries	Usage
y/n	Set this field to **y** to add a plus sign (+) to the beginning of a number to accommodate international calls. Default is **n**.

Send Transferring Party Information

Valid entries	Usage
y	Enter **y** to send the transferring party information on a transferred call.
n	Default. Transferring party information is not sent.

Telephone Event Payload Type

Use this field to control the default payload type offered by Communication Manager for SIP trunks. The payload type number encoding for originating (offering) the RFC 2833 RTP "telephone-event" payload format is based on the administered number from this field. This value is used only for Communication Manager originations (outgoing offers).

Valid entries	Usage
96 to **127**, or blank	Enter the RTP payload type. Default is **127**.

Field descriptions for Group Member Assignments page

The total number of pages of the **Trunk Group** screen, and the page number of the first page of **Group Member Assignments**, vary depending on whether the **Administrable Timers** and **ATMS Thresholds** pages display. Note that the **Group Member Assignments** screen is repeated, as needed, to allow assignment of all group members. This section does not cover ISDN trunks. For descriptions of the screens and fields that are unique to ISDN trunks, see ISDN Trunk Group on page 1242.

Note:

For SIP Enablement Services (SES) trunks, the group member-assignment pages are *not* individually administrable. The system automatically populates and displays these fields based on the number of members of SES trunk groups specified on page 1. Note that these display-only group member-assignment pages of the **Trunk Group** screen are repeated, as needed, to support all the trunk group's members.

Figure 585: Group Member Assignments screen

```
add trunk-group next                                              Page y of x
                                 TRUNK GROUP
                                        Administered Members(min/max): xxx/yyy
                                           Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port      Code  Sfx  Name          Night           Mode      Type    Ans Delay
    1: _______              ___________   _____           ________  _______  ____
    2: _______              ___________   _____           ________  _______  ____
    3: _______              ___________   _____           ________  _______  ____
    4: _______              ___________   _____           ________  _______  ____
    5: _______              ___________   _____           ________  _______  ____
    6: _______              ___________   _____           ________  _______  ____
    7: _______              ___________   _____           ________  _______  ____
    8: _______              ___________   _____           ________  _______  ____
    9: _______              ___________   _____           ________  _______  ____
   10: _______              ___________   _____           ________  _______  ____
   11: _______              ___________   _____           ________  _______  ____
   12: _______              ___________   _____           ________  _______  ____
   13: _______              ___________   _____           ________  _______  ____
   14: _______              ___________   _____           ________  _______  ____
   15: _______              ___________   _____           ________  _______  ____
```

Administered Members (min/max)

This display-only field shows the minimum and maximum member numbers that have been administered for this trunk group.

Ans Delay

CAUTION:
Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the length of time (in ms) your server running Communication Manager will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.
blank	Same as setting the field to zero.

This delay serves two purposes:

- It ensures that the answer supervision signal is valid and not a secondary delay-dial or wink-start signal.
- It ignores momentary off-hook signals resulting from connections made off-network through certain No. 5 Crossbar CCSA switches as the connection is being established. Therefore, calls aren't dropped inappropriately.

Code

This display-only field shows the type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address you enter, the field is blank.

Mode

This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip-switch settings on the circuit pack.

CAUTION:

Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
e&m	Enter **e&m** for 6-wire connections that pair 2 signaling wires with 4 voice wires. You'll use **e&m** in the vast majority of systems in the U.S.
simplex	Enter **simplex** for 4-wire connections that do not use an additional signaling pair. This configuration is very rare in the U.S.
protected	

Name

Your vendor, as well as Avaya technical support staff, sometimes need to identify specific trunks to work with your system. Therefore, the name you give to a trunk should identify the trunk unambiguously.

Valid entries	Usage
Up to 10 characters	Examples: ● The telephone number assigned to incoming trunks ● The Trunk Circuit Identification number assigned by your service provider

Night

Use this field only if you want to assign a night service destination to individual trunks that is different from the group destination entered in the **Night Service** field on page 1. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid entries	Usage
a valid extension	Enter the extension of the night destination for the trunk.
attd	Enter **attd** if you want calls to go to the attendant when night service is active.
blank	

Port

If this trunk is registered as an endpoint in an IP application, this field will display T00000.

Valid entries	Usage
1 to **64** (S87XX/S8300 Servers)	First and second characters are the cabinet number.
A to **E**	Third character is the carrier.
0 to **20**	Fourth and fifth characters are the slot number.
01 to **04** (Analog TIE trunks) **01** to **31** (CSI, S87XX Servers)	Six and seventh characters are the circuit number.
1 to **250** (S87XX/S8300 Servers)	Gateway
	1 of 2

Valid entries	Usage
V1 to **V9** (DEFINITY CSI, S87XX Servers)	Module
01 to **31** (DEFINITY CSI, S87XX Servers)	Circuit
	2 of 2

Note:

In DCS networks, trunks must be assigned the same member number at both nodes.

Members assigned to IP trunk groups will display a value of **T00001**.

When administering analog trunks connected to a TIM518, physical ports 17-24 are administered as ports 9 to 16 in Communication Manager.

Sfx

This display-only field shows the model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address you enter, the field is blank.

Total Administered Members

This display-only field shows the total number of members administered in the trunk group.

Type

The **Type** column appears when the **Trunk Type** field is blank or **cont**. The **Type** column does not display if the **Trunk Type** field is **dis**.

This field specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the **Trunk Type** field is **cont**), TN437, TN439, TN464 with any suffix, or TN458 circuit packs.

CAUTION:
Customers should not attempt to administer this field. Please contact your Avaya technical support representative for assistance.

Valid entries	Usage
t1-stan	t1-stan (DEFINITY, S87XX Series IP-PNC)
t1-comp	t1-comp (DEFINITY, S87XX Series IP-PNC)
t5-rev	(S87XX Series IP-PNC) The value of **t5 rev** is allowed only for the TN760D vintage 10 or later. When Type is **t5 rev**, Mode must be **e&m**.
type-5	type-5 (S87XX Series IP-PNC)

Related topics

See Tips for working with trunk groups on page 477 for instructions on adding and managing trunk groups.

See "Trunks and Trunk Groups" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information on all types of trunk groups except ISDN.

See "ISDN Service" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Uniform Dial Plan Report

Figure 586: Uniform Dial Plan Report screen

```
list uniform-dialplan                                              Page 1 of x
                             UNIFORM DIAL PLAN REPORT

 Matching Pattern    Len    Del     Insert Digits     Net      Conv     Node Num
    2                 4      0          817           aar       n
    4                 5      1          334           aar       n
    43659             5      1          928           aar       y
    6                 6      1                        ext       n
    73012             5      1                        enp       n          31
    74100             5      0          81            ars       y
```

Field descriptions for page 1

Matching Pattern

The number you want Communication Manager to match to dialed numbers.

Len

The number of user-dialed digits the system collects to match to the dialed string.

Del

The number of digits deleted before routing the call.

Insert Digits

The digits that will be inserted at the beginning of the dialed number.

Net

The server or switch network used to analyze the converted number.

Conv

Indicates whether additional digit conversion is allowed.

Node Num

The Extension Number Portability (ENP) node number.

Uniform Dial Plan Table

The **Uniform Dialing Plan** field must be **y** on the **System Parameters Customer-Options (Optional Features)** screen before you can administer this table.

The UDP provides a common dial plan length — or a combination of extension lengths — that can be shared among a group of Avaya S8XXX Servers. Additionally, UDP can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

See "Uniform Dial Plan" in *Feature Description and Implementation for Avaya Communication Manager,* 555-245-205, for more information.

Field descriptions for page 1

Figure 587: Uniform Dial Plan Table screen

```
change uniform-dialplan 0                                    page 1 of x
                        UNIFORM DIAL PLAN TABLE
                                                       Percent Full:   2
Matching Pattern Len Del Insert Digits Net Conv Node Num
1234567890123..  12  1   1234567890    123 n    123
```

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion

Del

Valid entries	Usage
0 to **9**	Enter the number of digits to delete before routing the call. This number must be less than or equal to the number entered in the **Len** field.

Insert Digits

Valid entries	Usage
0 to **9** (1 to 4 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
L*x* (1 to 5)	The variable ***x*** represents a number of digits taken from the locations prefix on the **Locations** screen. These digits are prepended to the dialed string. The value for ***x*** must less than the number of digits in the location prefix. Type a number between 1 and 5 that represents the number of leading x digits that should be prepended to (added to the front of) the dialed string.

Len

Valid entries	Usage
1 to **18**	Enter the number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the number entered in the **Matching Pattern** field. The value **2** can be used only when **Insert Digits** contains an **L*x*** value, where ***x*** is the number of leading digits to prepend for the location of an originating call.

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 7 digits)	Enter the number you want Communication Manager to match to dialed numbers.

Net

Enter the server or switch network used to analyze the converted number.

Valid entries	Usage
aar **ars** **enp** **ext**	The converted digit-string will be routed either as an extension number or via its converted AAR address, its converted ARS address, or its ENP node number. If you enter **enp**, you must enter the ENP node number in the **Node Num** field. The **Insert Digits** field must be blank, and **Conv** must be **n**.

Node Num

This is the ENP (Extension Number Portability) Node Number.

Valid entries	Usage
1 to **999**	Enter the ENP node number.

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used.

Acceptable Service Level (sec)

Only appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **BCMS/VuStats Service Level** field is **y** and the **Measured** field is **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this VDN should be answered. This will allow BCMS to print out a percentage of calls that were answered within the specified time.

User Profile

This screen is described in *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431.

For more information on administering user profiles and logins, see "AAA Services" in *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.

Variables for Vectors

Use this screen to create variables and define the necessary parameters for each variable type. You can specify the variable type, the name to use for the variable, the size of the variable, how the variable gets set/assigned and whether the variable is local or global. Up to 702 variables can be supported using A to Z and AA to ZZ rows.

Field descriptions for page 1

Figure 588: Variables for Vectors screen - page 1

```
change variables                                                Page  1 of x
                            VARIABLES FOR VECTORS

Var   Description       Type  Scope Length Start   Assignment        VAC
A:    _______________   _____ __    ____   ___
B:    _______________   _____ __    ____   ___
C:    _______________   _____ __    ____   ___
D:    _______________   _____ __    ____   ___
E:    _______________   _____ __    ____   ___
F:    _______________   _____ __    ____   ___
G:    _______________   _____ __    ____   ___
H:    _______________   _____ __    ____   ___
I:    _______________   _____ __    ____   ___
J:    _______________   _____ __    ____   ___
K:    _______________   _____ __    ____   ___
L:    _______________   _____ __    ____   ___
M:    _______________   _____ __    ____   ___
```

Figure 589: Variables for Vectors screen - page 2

```
change variables                                                Page  2 of x
                            VARIABLES FOR VECTORS

Var   Description       Type   Scope Length Start   Assignment        VAC
N:    _______________   _____ __     ____   ___
O:    _______________   _____ __     ____   ___
P:    _______________   _____ __     ____   ___
Q:    _______________   _____ __     ____   ___
R:    _______________   _____ __     ____   ___
S:    _______________   _____ __     ____   ___
T:    _______________   _____ __     ____   ___
U:    _______________   _____ __     ____   ___
V:    _______________   _____ __     ____   ___
W:    _______________   _____ __     ____   ___
X:    _______________   _____ __     ____   ___
Y:    _______________   _____ __     ____   ___
Z:    _______________   _____ __     ____   ___
```

Assignment

This field only allows entry when the **Type** is **value** or **collect G**. Entry of an **Assignment** for **value** or **collect G** is optional. That is, it can be left blank. The current value/assignment for each global variable is always displayed in the **Assignment** column when you access the **Variables for Vectors** screen. This includes the **doy**, **dow**, and **tod** types which show the current values from the switch clock as a display-only entry in the **Assignment** column.

Valid entries	Usage
digits	Enter a number to pre-assign to the variable. This field displays the current value for global values

Description

Valid entries	Usage
up to 27 alphanumeric characters, or blank	Optionally enter an identifying name or description of the vector variable. Default is blank.

Length

This field specifies the maximum number of digits from the data to assign to the variable. Length does not apply to the **tod**, **doy**, **dow** or **vdn** variables. When **Type** is **value**, the length is pre-populated with **1**. A length entry is required for all types to which it applies.

Valid entries	Usage
1 to **16**	Enter the maximum length of digits to use in the variable.

Scope

This field only allows an entry for variables that can be *either* local or global. For those variables that can only be one or the other, the **L** or **G** value is pre-populated automatically after you enter the **Type**.

Valid entries	Usage
G/L	Indicate whether the variable is to be used locally (**L**) or globally (**G**).

Start

This field specifies the beginning character position of the data digits string to be used for assigning to the variable. The combination of the **Start** position and maximum length of the digits string defines what is to be assigned to the variable. If the number of digits to be used is less than the maximum length specified, only that portion is assigned to the variable. **Start** only allows entry when **Type** is **collect** or **asaiuui**.

Valid entries	Usage
1 to **96**	Enter the character position of the first digit to be stored in the variable.

Type

Valid entries	Usage
ani **asaiuui** **collect** **dow** **doy** **stepcnt** **tod** **value** **vdn** **vdntime**	Enter the vector variable type.

Var

Valid entries	Usage
A to **Z**, **AA-ZZ**	Display only. The letter identifying the row of a specific vector variable.

VAC

The **VAC** (Variable Access Code) column only allows entry (**1** to **9** or blank) when the **Type** is **value**. Entry is not required for this type. If VAC is left as a blank, assignment is done using the **Assignment** column. The **VVx** entry is one of the Vector Variable feature items on the **Feature Access Code (FAC) screen** that can be assigned a feature access code (FAC).

Valid entries	Usage
1 to **9** or blank	Displays the Vector Variable Feature Access Code (FAC) to use for changing the value.

Vector

See **Call Vector**.

Vector Directory Number

This screen defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector.

VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed via direct dial CO trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN can be Night Destination for LDN.

See *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780, for more information.

Field descriptions for page 1

Figure 590: Vector Directory Number screen

```
change vdn nnnn                                                      Page  1 of x
                       VECTOR DIRECTORY NUMBER

                             Extension: nnnn
                                  Name*:
                         Vector Number: xxxx
                   Attendant Vectoring: n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                    COR: 59
                                    TN*: 1
                               Measured: none
         Acceptable Service Level (sec):
                Service Objective (sec):
         VDN of Origin Annc. Extension*: 301
                             1st Skill*:
                             2nd Skill*:
                             3rd Skill*:

* Follows VDN Override Rules
```

1st/2nd/3rd Skill

Only appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **Expert Agent Selection (EAS)** field is **y** and the **Meet-me Conferencing** field is **n**.

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **999** or blank	Enter the desired Skill numbers in each field (or leave blank). The default is blank.

Acceptable Service Level (sec)

Only appears when, on the **System Parameters Customer-Options (Optional Features)** screen, the **BCMS/VuStats Service Level** field is **y** and the **Measured** field is set to **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds or blank	Enter the number of seconds within which calls to this VDN should be answered. This allows BCMS to report the percentage of calls that were answered within the specified time. The default is blank.

Allow VDN Override

This field appears if the **Meet-me Conferencing** field is **n**. The **Allow VDN Override** field allows the system to change the "active" VDN for a call. The “active” VDN is the VDN to be used for parameters associated with the call such as VDN name, skills, tenant number, BSR application, VDN variables, etc.

Parameters (VDN fields) for the call that are defined by the “active” VDN include the fields in the following list. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to each field name, indicating that the field follows VDN override rules when the system changes the "active" VDN for a call.

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement Extension
- VDN skills (1st, 2nd, 3rd)
- Return Destination
- VDN Timed ACW Interval
- BSR Application
- BSR Available Strategy
- BSR Tie Strategy
- Display VDN for Route-to DAC

- ISDN Trunk ASAI Messages (depending on field setting)
- BSR Local Treatment
- VDN Variables
- VDN Time Zone Offset

Note:

The "active" VDN can be specified in some vector commands as a keyword. When a vector step with the keyword "active" is executed, the extension for the call's "active" VDN as defined by VDN override rules is substituted for the keyword when processing the vector command. The keyword "active" can be used as the VDN extension for the **goto** command "counted-calls" conditional, the **goto** command "rolling-asa for VDN" conditional, the messaging command mailbox extension, or can be defined as the "vdn" vector variable type assignment. The keyword "latest," (the last VDN routed to), can also be assigned in these same vector commands or variables, but the "latest" VDN is not changed by VDN Override settings.

Valid entries	Usage
y	Entering **y** in this field allows a routed-to VDN (by a route-to number or route-to digits vector command) to become the “active” VDN. The first VDN reached by the call becomes the “active” VDN.
n	The routed-to VDN does not become the active VDN. The parameters of the original VDN are used. This is the default.

Attendant Vectoring

This field appears when, on the **System Parameters Customer-Options (Optional Features)** screen, **Attendant Vectoring** is **y**. This field indicates if the vector you are defining is an attendant vectoring VDN.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. This entry will dynamically change the rest of the screen to eliminate field options available with other types of vectors.
n	Default.

COR

Specifies the class of restriction (COR) of the VDN.

Valid entries	Usage
0 to **995**	Enter a 1 or 2-digit number. This field cannot be blank.

Extension

This is a display-only field showing the extension number of the VDN. The extension is a number that starts with a valid first digit and length as defined by the system's dial plan.

Measured

This field appears if the **Meet-me Conferencing** field is **n**. Used to collect measurement data for this VDN. Data can be collected for reporting by BCMS or CMS.

Note:

On the **System Parameters Customer-Options (Optional Features)** screen, the **BCMS** field must be **y** for the **Measured** field to be set to **internal** or **both**. In addition, the appropriate CMS release must be administered on the **Feature-Related System Parameters** screen if this field is being changed to **external** or **both**.

Valid entries	Usage
internal	Data will be measured internally by BCMS.
external	Data will be measured internally by CMS.
both	Data will be measured internally by both BCMS and CMS.
none	Data will not be measured. This is the default.

Meet-me Conference

This field appears only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Enhanced Conferencing** field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

Note:

If the VDN extension is part of your DID block, external users will be able to access the conference VDN. If the VDN extension is not part of your DID block, only internal callers on the your network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this VDN. If Meet-me Conference is **y**, only **Extension**, **Name**, **Vector Number**, **Meet-me Conference**, **COR**, and **TN** fields display and the fields for page 2 change. Both **Attendant Vectoring** and **Meet-me Conference** cannot be enabled at the same time. If **Enhanced Conferencing** is **y**, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Note:

If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller might not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Name

This is an optional field that need not contain any data. It is the name associated with the VDN. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
Enter up to a 27-character alphanumeric name that identifies the VDN.	The name might be truncated on agents' displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent. **Note:** The **Name** field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, Sage, Spark, and 9600-series Spice telephones. Unicode is also an option for the 2420J telephone when **Display Character Set** on the **System Parameters Country-Options** screen is **katakana**. For more information on the 2420J, see *2420 Digital Telephone User's Guide*, 555-250-701.For more information on Unicode language display, see Administering Unicode display on page 203. Avaya BRI stations support ASCII characters only. They do not support non-ASCII characters, such as Eurofont or Kanafont. Therefore, if you use non-ASCII characters in any Communication Manager **Name** field, such characters will not display correctly on a BRI station.

Service Objective

This field appears in either of the following scenarios:

- When, on the **System Parameters Customer-Options (Optional Features)** screen, the **BCMS/VuStats Service Level** field is set to **y**, and the **Measured** field is set to **internal** or **both**. Enter the number of seconds within which calls to this VDN should be answered. This will allow BCMS to report the percentage of calls that were answered within the specified time. Valid entries are **0** to **9999**, or blank. Default is blank.
- When, on the **System Parameters Customer-Options (Optional Features)** screen, the **Dynamic Advocate** field is set to **y**. This field enables the Dynamic Queue Position feature, which allows you to queue calls from multiple VDNs to a single skill, while maintaining different service objectives for those VDNs. Enter the service level, in seconds, that you want to achieve for the VDN. Valid entries are **1** to **9999**. The default is **20**.

TN

Specifies the Tenant Partition number for this VDN. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **100**	For S87XX Series IP-PNC.

VDN of Origin Annc. Extension

Use this field to specify the extension number of the VDN of Origin announcement. A VDN of Origin announcement is a short recording that identifies something about the call originating from the VDN. The agent hears the recording just prior to the delivery of the call. Data for this field appears only when, on the **System Parameters Customer-Options (Optional Features)** screen, the **VDN of Origin Announcement** field is **y** and, on this screen, the **Meet-me Conferencing** field is **n**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
VDN extension	Enter the extension number of the VDN of Origin announcement.

Vector Number

Valid entries	Usage
1 to **system max**	Enter an identifying number that specifies a particular call vector that is accessed through the VDN. This field cannot be blank.

Field descriptions for page 2 (Meet-me Conference is n)

The second page of the **Vector Directory Number** screen contains the name of the corresponding Audix server (if present), the BSR available agent strategy, whether the VDN is displayed in route to direct agent call situations, and settings for other optional features.

Figure 591: Vector Directory Number screen

```
change vdn nnnn                                                Page  2 of  x
                       VECTOR DIRECTORY NUMBER

                                AUDIX Name:

                       Return Destination*:
                   VDN Timed ACW Interval*:
                          BSR Application*:
             BSR Available Agent Strategy*: 1st_found
                         BSR Tie Strategy*: 1st_found
                   Observe on Agent Answer? n

             Display VDN for Route-To DAC?* n
VDN Override for ISDN Trunk ASAI Messages?* n

                      BSR Local Treatment?* n

* Follows VDN Override Rules
```

AUDIX Name

Only appears for S87XX Series IP-PNC configurations. If this VDN is associated with the AUDIX vector, enter the name of the AUDIX machine as it appears in the **IP Node Names** screen.

BSR Application

To use multi-site Best Service Routing with this VDN, enter a one to three-digit number to specify an application plan for the VDN. This field appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Lookahead Interflow (LAI)** and **Vectoring (Best Service Routing)** fields are **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **255** or blank	Enter a 1 to 3-digit number. For DEFINITY CSI.
1 to **511** or blank	Enter a 1 to 3-digit number. For S8300/S87XX Servers.

BSR Available Agent Strategy

The available agent strategy determines how Best Service Routing identifies the best split or skill to service a call in an agent surplus situation. To use Best Service Routing with this VDN, enter an agent selection strategy in this field.

This field only appears if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Vectoring (Best Service Routing)** field is **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1st-found	BSR uses the first selection for routing; that is, the current best selected from the previous consider commands.
UCD-LOA	The call is routed to the least occupied agent, without regard to skill level. Can be set only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
UCD-MIA	The call is routed to the most idle agent, without regard to skill level. This type of call distribution ensures a high degree of equity in agent workloads even when call-handling times vary.
EAD-LOA	The call is routed to the highest skill level agent with the lowest occupancy. Can be set only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Least Occupied Agent (LOA)** or **Business Advocate** field is **y**.
EAD-MIA	The call is routed to the highest skill level, most idle agent. Can be set only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Expert Agent Selection (EAS)** field is **y**.

BSR Local Treatment

In a multi-site BSR configuration, a call that arrives at a local communication server can be rerouted to a remote server located in a different part of the world. This feature maintains control at the local server and allow you to provide local audio feedback for IP and ISDN calls, or to take back the call while the call waits in queue on a remote server. When **Meet-me**

Conferencing is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
y/n	A **y** entry in this field allows you to provide local audio feedback for IP and ISDN calls while a call waits in queue on a remote server. **Note:** The **BSR Local Treatment** field must be set to **y** on both the local and remote vdns, or else call interflow attempts might result in dropped calls.

BSR Tie Strategy

This field appears only when **Vectoring (Best Service Routing)** on the **System Parameters Customer-Options (Optional Features)** screen is **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
system	The setting of the **BSR Tie Strategy** field on the **Feature-Related System Parameters screen** applies.
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Display VDN for Route-To DAC

This field can be set to **y** only if, on the **System Parameters Customer-Options (Optional Features)** screen, the **Expert Agent Selection (EAS)** field is **y**. This field applies when either:

- A route-to number with coverage = y or route-to digits with coverage = y vector command routes a call to an agent as an EAS direct agent call
- Adjunct routing routes a direct agent call to the agent

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

For more information, see *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, 07-600780.

Valid entries	Usage
y/n	Enter **y** to allow display of the VDN.

Observe an Agent Answer

Valid entries	Usage
y/n	This field allows for a service observer to start observing a call to the VDN when the call is delivered to the agent/station.

Return Destination

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
VDN extension or blank	Enter the VDN extension number to which an incoming trunk call will be routed if it returns to vector processing after the agent drops the call.

VDN Override for ISDN Trunk ASAI Messages

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.This field appears only if the following conditions are set on the Communication Manager license file:

- On the **System Parameters Customer-Options (Optional Features)** screen, the **ASAI Link Core Capabilities** field is **y**.
- On the **System Parameters Customer-Options (Optional Features)** screen, the **G3 Version** field is set to **V10** or later

Additionally, you can set this field to **y** only when the **Allow VDN Override** field on this screen is also set to **y**. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
y	When an incoming call routes, the "Called Number" information sent in the "Call Offered," "Altering," "Queued," and "Connect" ASAI events and the "Adjunct Route Request" ASAI message, is the "active VDN" extension associated with the routed call.
n	The "Called Number" information sent for the ASAI event notification and adjunct-request messages does not change for a ISDN-PRI trunk. It is always the number in the Called Number IE sent in the incoming ISDN call's SETUP message.

VDN Timed ACW Interval

When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field takes precedence over the **Timed ACW Interval** field on the **Hunt Group** screen. When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

Valid entries	Usage
1 to **9999** or blank	Enter the number of seconds the agent should remain in ACW following the call.

Field descriptions for page 2 (Meet-me Conference is y)

The fields on this screen are displayed when the **Meet-me Conference** field on page 1 of the **Vector Directory Number** screen is **y**.

Figure 592: Vector Directory Number screen (if the Meet-me Conference field is y)

```
change vdn nnnn                                                 Page 2 of x
                         VECTOR DIRECTORY NUMBER

                      MEET-ME CONFERENCE PARAMETERS
    Conference Access Code:123456
     Conference Controller:
           Conference Type: expanded
           Route-to Number:
```

Note:

If the vector for Meet-Me conferencing allows a new party to join a conference immediately, and that party is joining as an H.323 ip trunk user, the caller might not have talkpath with the others in the conference. To prevent this, include in the vector a short delay before a new party joins the Meet-Me conference, such as a step to collect digits, a 1-second delay, or play an announcement. Since Meet-Me vectors are always configured with announcements and digit collections, this should rarely be an issue.

Conference Access Code

To ensure conference security, you should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
6-digit number or blank	Enter a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the *init* superuser login.

Conference Controller

This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in via remote access trunks.

Valid entries	Usage
extension number or blank	If an extension number is entered, only a user at that extension can change the access code for that VDN using a feature access code. If this field is blank, any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Conference Type

Use this field to select the conference type that is appropriate for your call. For six or fewer participants, enter **6-party**. For a conference with more than six participants, select **expanded**.

Valid entries	Usage
6-party **expanded**	Enter **expanded** to enable the Expanded Meet-me Conference feature. Default is **6-party**.

Route-to Number

This field appears only if the **Conference Type** field is **expanded**. This field allows administration of the routing digits (the ARS/AAR Feature Access Code with the routing digits and the Conference ID digits for the VDN).

Valid entries	Usage
up to 16 digits	Enter the ARS or AAR Feature Access Code (FAC) followed by the routing digits. Alternately, you can enter the unique UDP extension. **Note:** The Route-to Number must be unique across all Expanded Meet-me Conference VDNs.

Field descriptions for page 3

Figure 593: Vector Directory Number screen

```
VECTOR DIRECTORY NUMBER                                      Page 3 of x

                                 VDN VARIABLES*

                    Var  Description        Assignment
                    V1   _______________    ________________
                    V2   _______________    ________________
                    V3   _______________    ________________
                    V4   _______________    ________________
                    V5   _______________    ________________
                    V6   _______________    ________________
                    V7   _______________    ________________
                    V8   _______________    ________________
                    V9   _______________    ________________

VDN Time Zone Offset*: + HH:MM
                   Daylight Savings Rule*: system
* Follows VDN Override Rules
```

Assignment

The assignment field assigns an up to 16-digit unvalidated decimal number to each of the VDN variables V1 through V5. Valid entries for each assignment can be a string of up to 16 digits using **0** to **9**, or blank.

Daylight Savings Rule

Use this field to define the daylight saving time rule. This field is used with the **VDN Time Zone Offset** field. The daylight saving time rule and the time zone offset are applied to `goto` time-of-day commands in the vector that is assigned to the VDN. The time-of-day calculations are based on the local time of the receiving call's VDN. The assigned rule number applies start and stop rules that are administered on the system **Daylight Savings Rule** field for that rule number.

Tip:

Use the **`list usage vdn-time-zone-offset`** command to find VDNs containing an administered daylight saving time rule.

Valid entries	Usage
system	The system uses the same daylight saving time rule as the system clock shown in the **display/set time** field.
0	No daylight saving rule is applied. If the system time has a daylight saving rule specified, this rule is removed before evaluating the **`goto if time-of-day`** conditional.
1 to **15**	Indicates the rule as defined on the Daylight Savings Rule field. When you use a number other than 0, the rule associated with the main server clock display time and the main server offset are not used. The offset and rule assigned to the active VDN for the call are applied to the operating system standard time so that local time for the VDN is used to test the time-of-day step.

Description

This field is displayed only if VDN Variables is active. The description field allows users to describe the VDN variable using up to 15 characters.

Var

The number assigned to the VDN variable. You can assign up to 9 VDN variables.

VDN Time Zone Offset

This field is applied against the switch clock when a time of day vector command is executed. Daylight savings time changes are handled by the switch clock using the existing operation. Based on a syntax of +HH:MM, the valid entries are:

- **+**, **-**
- **00-23** - hour
- **00-59** - minute

The default is **+00:00**. When the default is set, the system switch time is used without modification.When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the field name, indicating that this field follows VDN override rules when the system changes the "active" VDN for a call.

For more information about this feature, see *Avaya Call Center Release 4.0 Automatic Call Distribution (ACD) Guide*, 07-600779.

VDN Variables

When **Meet-me Conferencing** is **n**, an asterisk (*) appears next to the heading, indicating that variables **V1** through **V9** follow VDN override rules when the system changes the "active" VDN for a call.

Video Bridge

Use the **Video Bridge** screen to configure available ad-hoc conferencing resources. You can administer up to 40 video bridges.

For more information on Ad-hoc Conferencing, see Administering Ad-hoc Video Conferencing. For more detailed information on Avaya Video Telephony, see *Avaya Video Telephony Solution Networking Guide*, 16-601423.

Field descriptions for page 1

Figure 594: Video Bridge screen

```
add video-bridge next                                          Page 1 of x
                                VIDEO BRIDGE
Bridge ID: 3
     Name:

Max Ports:

Trunk Groups: (Must have at least one incoming and one outgoing, or a two-way)
1: ____
2: ____
3: ____

  Far End Resource Info:

               ID Range: _________ to _________
Priority Factory Number:_________
Standard Factory Number:_________
```

Bridge ID

Valid entries	Usage
1 to 40	Display only. Shows the ID number for this video bridge.

Far End Resource Info

Valid entries	Usage
y	The far end tracks port usage and provides updates on resource availability.
n	No resource information is provided from the far end.

ID Range Start/End

These fields appear when the **Group Type** field on the **Trunk Group** screen is **h.323**.

Valid entries	Usage
Enter 1-9 digits (0,9)	Enter a range of conference IDs that this video bridge can use. There must be enough IDs so that all of the ports can be used – one ID for every six ports. Default is blank.

Max Ports

Valid entries	Usage
3 to system max	Enter the maximum number of video conferencing ports for this video bridge. Default is none.

Name

Valid entries	Usage
up to 30 alphanumeric characters	Enter a name for identifying this video bridge. Default is blank.

Priority Factory Number

This field appears when the **Group Type** field on the **Trunk Group** screen is **h.323** or **sip**, and the **Far End Resource Info** field on the **Video Bridge** screen is **y**. When creating an ad-hoc conference call, Communication Manager first contacts the conference factory, which allocates the ad-hoc conference ID, and establishes an audio channel between the video bridge and Communication Manager audio resources. Priority vs. Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates it, the priority factory is used, which may have better bandwidth or a dedicated video bridge. Note that the Priority/Standard distinction only applies when the **Far End Resource Info** field on the **Video Bridge** screen is **y**.

Valid entries	Usage
1-9 digits (0,9), or blank	At least one of **Priority Factory Number** or **Standard Factory Number** must be filled in. If **Priority Factory Number** is blank, priority calls can use the bridge, but will prefer a bridge with a priority factory. Standard and Priority factory numbers can be the same. Default is blank.

Standard Factory Number

This field appears when the **Group Type** field on the **Trunk Group** screen is **h.323** or **sip**. For h.323, the **Far End Resource Info** field on the **Video Bridge** screen must be **y**. When creating an ad-hoc conference call, Communication Manager first contacts the conference factory, which allocates the ad-hoc conference id, and establishes an audio channel between the video bridge and Communication Manager audio resources. Priority vs. Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates it, the priority factory is used, which may have better bandwidth or a dedicated video bridge. Once established as a priority conference, the call remains priority even if the priority user drops off. Note that the Priority/Standard distinction only applies when the **Far End Resource Info** field on the **Video Bridge** screen is **y**.

Valid entries	Usage
1-9 digits (0,9), or blank	At least one of Priority Factory Number or Standard Factory Number must be filled in. If **Standard Factory Number** is blank, non-priority conferences are unable to use this video bridge. Standard and Priority factory numbers can be the same. Default is blank.

Trunk Groups

Use this field to assign trunk groups to this video bridge. You must have at least one incoming and one outgoing trunk, or a two-way trunk. Note that all trunks on a given video bridge must be the same type; you cannot mix H.323 and SIP.

Valid entries	Usage
1 to **2000**	Enter administered SIP or ISDN H.323 trunk groups. Default is blank.

Virtual MAC Addresses

The **Virtual MAC Addresses** screen lists the virtual Media Access Control (MAC) addresses on your system.

Field descriptions for page 1

Figure 595: Virtual MAC Addresses

```
display virtual-mac-address 1                                    Page 1 of x
                   VIRTUAL MAC ADDRESSES - TABLE: 1
      MAC Address     Used                       MAC Address     Used
    00:04:0d:4a:53:c0   y                      00:04:0d:4a:53:cf   n
    00:04:0d:4a:53:c1   n                      00:04:0d:4a:53:d0   n
    00:04:0d:4a:53:c2   n                      00:04:0d:4a:53:d1   n
    00:04:0d:4a:53:c3   n                      00:04:0d:4a:53:d2   n
    00:04:0d:4a:53:c4   n                      00:04:0d:4a:53:d3   n
    00:04:0d:4a:53:c5   n                      00:04:0d:4a:53:d4   n
    00:04:0d:4a:53:c6   n                      00:04:0d:4a:53:d5   n
```

MAC Address

Valid entries	Usage
15 alpha-numeric characters	Virtual MAC address shared by duplicated TN2602AP circuit packs. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Used

Valid entries	Usage
y/n	This field is autopopulated. If **y**, the associated virtual MAC address has been assigned in the system

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