



# **What's New in Avaya Communication Manager**

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## Contents

# About this book

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## Overview

Avaya Communication Manager is the centerpiece of Avaya applications. Running on a variety of Avaya Media Servers and DEFINITY<sup>®</sup> 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.

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## Purpose of this book

This book describes the new and changed features and enhancements available with the most recent release of Communication Manager (release 3.x) running on any of the following:

- Avaya media servers
  - DEFINITY<sup>®</sup> servers
  - S8000-series media servers
  - IBM eServer BladeCenter HS20 Blade Server Type 8832
- Avaya media servers configured as a Local Survivable Processor (LSP) or Enterprise Survivable Servers (ESS).
- Avaya media gateways

**Note:**

This document does not contain information about prior releases of Communication Manager. For information on previous releases of Communication Manager, check the Avaya customer support Web site (see [How to obtain Avaya books on the Web](#) on page 14 for more information).

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

## Contents

This document includes the following chapters:

- [What's New](#): presents short descriptions of each of the new features or changes in the most recent release of Communication Manager.
  - [Hardware](#): describes hardware that is introduced or changed with the most recent release of Communication Manager.
  - [New and changed screens](#): provides information about new administration screens, and changes to existing screens, due to the most recent release of Communication Manager.
  - [New and changed commands](#): provides information about commands that are new or have changed for the most recent release of Communication Manager.
- 

## Terms and Conventions

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 for an example of a screen and how it is shown in this book.
- Avaya uses the term "telephone" in this book. Other books might refer to telephones as 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 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 bold 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. 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:**

A note calls attention to neutral information or positive information that supplements the main text. A note also calls attention to valuable information that is independent of the main text.



**Important:**

An important note calls attention to situations that can cause serious inconvenience.



**Tip:**

A tip calls attention to information that helps you apply the techniques and the procedures that the text describes. A tip can include keyboard shortcuts, or alternative methods that might not be obvious.



**CAUTION:**

A caution statement calls attention to situations that can result in harm to software, loss of data, or an interruption of service.



**WARNING:**

A warning statement calls attention to situations that can result in harm to hardware or equipment.



**DANGER:**

A danger statement calls attention to situations that can result in physical injury to yourself or to other people.



**SECURITY ALERT:**

A security alert calls attention to situations that can increase the potential for toll fraud or other unauthorized use of your telecommunications system.



**ELECTROSTATIC ALERT:**

An electrostatic alert calls attention to situations that can result in damage to electronic components from electrostatic discharge (ESD).

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3. To find a specific book, type the document number (for example, **03-300682** for this book) in the **Search** text box.
4. In the resulting list, locate the latest version of the document, and then click the document title to view the latest version of the book.

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- Fax: 1-303-538-1741
- Contact your Avaya representative

Mention the name, number, and issue of this document: *What's New in Avaya Communication Manager*, 03-300682, Issue 1.

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If you need additional help, the following resources are available. You may need to purchase an extended service agreement to use some of these resources. See your Avaya representative for more information, or go to the Avaya Support Web site at <http://www.avaya.com/support>:



# Chapter 1: What's New

This chapter presents highlights of features and enhancements as part of the most current release of Avaya Communication Manager running on Avaya DEFINITY® servers, as well as the Avaya S8000-series media servers with associated Avaya media gateways.

The most current release of Communication Manager contains all the features of prior releases. In this document, each Communication Manager feature or enhancement is listed alphabetically by release number.

- For an overview of the features of Communication Manager, see the *Overview for Avaya Communication Manager*, 03-300468.
- For a more complete description of the features of Communication Manager, see the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205.
- For more information on how to administer these features, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

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## Release 3.1 new features and enhancements

Avaya Communication Manager, release 3.1, includes the following general telephony and system-wide features and enhancements.

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### 802.1x multi supplicants

Multi-suplicants are common in IP telephony where PC and IP endpoints are attached to the same port. For better security and to reduce interdependency between the PC and IP endpoints, the multi-suplicants mode enables each supplicant to independently authenticate itself to gain access to the network. Multi-suplicants support C360 R3, G350, and G250 platforms.

In remote sites, the multi-suplicants mode provides:

- An extra level of security by restricting access only to known users and devices
- Consistency of security features offered in the gateways' LAN interfaces in case of multi-vendor networks (Avaya gateways and Extreme switches)

---

## Administrable size for Receive Buffer TCP Window

The **IP Interfaces** screen now includes the **Receive Buffer TCP Window Size** field. You can set the value of this field to any value from 512 to 8320 (the default value). The value is the number of bytes that are allotted for the buffer that receives TCP data for a TN799 (CLAN) circuit pack.

To see the **IP Interfaces** screen, type `display ip-interface n`, where `n` is the location of the TN799 (CLAN) circuit pack.

---

## Administrable time-out for inactive SAT sessions

You can now administer the idle time that the system permits before the system shuts down a SAT session. Set the **Command Time-Out** field on the **Maintenance-Related System Parameters** screen to any value from 10 to 360 minutes.

The default value is 120 minutes.

---

## Alarm messages for unregistered LSPs

Alarms are generated on both the main server and on an LSP if Keep Alive (KA) messages are not periodically exchanged. In addition, an audit is required to verify that all LSPs have registered within twenty minutes of system initialization.

---

## ASAI support for Aux Work reason codes

You can now assign up to 99 Aux Work reason codes, rather than only 10. The description for each reason code can now be up to 16 characters, rather than only 10 characters.

**Note:**

ASAI does not currently support two-digit reason codes.

To take advantage of the additional reason codes, set the:

- **Reason Codes** field on the **Customer Options** screen to **y**
- **Expert Agent Selection (EAS) Enabled** and the **Two-Digit Aux Work Reason Codes** fields on the **Feature-Related System Parameters** screen to **y**.

---

## Avaya Video Telephony Solution

The Avaya Video Telephony Solution makes video calls as simple and easy as a regular telephone call. The Avaya Video Telephony Solution is fully integrated into your standard dial plan, enabling totally transparent and seamless voice and video conferencing, both for the desktop and for group video communications.

Communication Manager features such as hold, transfer, resume, and conference are seamless with video conferencing adjuncts from Polycom. Avaya Video Telephony Solution 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 VSX 8000 conferencing systems with Release 8.03 or later
- Polycom V500 video calling systems
- Polycom MGC video conferencing bridge platforms with Release 7.02
- Third party gatekeepers

The solution requires Communication Manager Release 3.0.1, and Avaya IP Softphone release 5.2, with Avaya Integrator for Polycom Video release 2.0.1.

The Avaya Video Telephony Solution also supports the:

- Logitech 4000 Pro web camera
- Polycom Via Video
- Creative Labs notebook webcam

---

## Banner displayed to warn of reset

When a new license file is loaded which changes the value of FEAT\_ESS from that of the previous license files, a "reset sys 4" is required in order for the change to take effect.

If the "reset sys 4" is not done, a banner is displayed on the initial SAT screen warning the user that a reset is required.

---

## Block circuit pack installation if wrong suffix

In an Enterprise Survivable Server (ESS) system:

- When a TN2305 or TN2306 Asynchronous Transfer Mode (ATM) Expansion Interface (EI) suffix "A" circuit pack is inserted in a position where a suffix "B" circuit pack is required, the circuit pack insertion is prevented and an alarm (MINOR ON\_BOARD) is raised.
- When a TN750 EI circuit pack, with a suffix other than "D," is inserted in a position where a suffix "D" circuit pack is required, the circuit pack insertion is prevented and an alarm (MINOR ON\_BOARD) is raised.

---

## Block CMS Move Agent events

This feature lets you prevent the system from sending the ASAI logout-login event messages, that are related to an agent move. When this CTI link option is activated, the changes to the agent state, such as logout followed by login and return to previous state, will not be reported to the ASAI adjunct. This operation is required by Avaya IC since the initial logout causes IC to permanently logout the agent, disrupting normal operation. IC does not need to be informed of agent skill moves via this method. This option will be available to other applications for use where needed.

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## Call Log modifications

In previous releases of Communication Manager, the extension number of some DCP and IP telephones did not appear on the Call Log for call forwarded calls and for bridged call appearance of extension calls. The word "Unavailable" appeared in the **Number** field of the Call Log in these scenarios.

Communication Manager release 3.1 now displays the extension on the Call Log for call forwarding and bridged call appearances on the following IP telephones:

- 4610
- 4620
- 4621
- 4622
- 4625

Communication Manager release 3.1 now displays the extension on the Call Log for call forwarding only on the following DCP telephones:

- 2410
- 2420

**Note:**

At this time, the current firmware version of the 2410 and 2420 telephones do not support an option for providing call logs for bridged appearances.

---

## Clear the display of collected digits

You can define when the system clears the display of collected digits (**Callr-info**) from the agent telephone. The system allows the:

- Existing default option to clear the display when the next call is received
- Option to clear the display when the call is released
- Option to keep the displayed digits while the agent is in After Call Work (ACW) mode.

---

## Compress restart escalation sequence

Problems that were not being fixed by a first cold2 restart were not necessarily being fixed by a second cold2 restart. The escalation sequence is now compressed by proceeding to the next restart level, which is a reload.

---

## Connection-preserving upgrades

When upgrading an S8700-series Media Server from 3.0 to a later release, you can preserve many connections through the course of the upgrade. For more detailed information, see the Documentation Updates in the Release Notes for S8700-series Media Servers.

---

## Detecting 655A power supply failures

If a 655A power supply should fail in a duplicated power G650 port network, it is now possible to determine what 655A power supply out of a possible 10 has failed.

## Dial backup over external ISDN modem

If there is a primary WAN failure, this feature offers a backup means for the control channel between the branch office and the main site.

The gateway attempts to reestablish the control channel through an alternate route by dialing back to the main site through an external modem that is connected to the serial port or the USB port. The external modem, connected to the PSTN, allows dial-up connection to a router or modem at the main site.

This feature is implemented on the G250 and G350 Media Gateways.

---

## Direct-region preference for IP telephones

In many customer configurations, IP telephones are placed into their own direct network regions or indirect network regions. Voice over IP (VOIP) allocation can now favor direct-connected regions over indirect connected regions.

---

## Duplicate power supply failure upgraded to alarm status

A redundant power supply can fail looking like it has no AC input. Since the power supplies are duplicated the customer experiences no service outage. This failure could occur if the power cord to the power supply is disconnected, or switching off the equipment room that powers that breaker.

This problem is first logged as a Warning. If the problem exists for 1-2 hours, the problem is escalated to a Minor alarm status. At that time, Services can contact the customer to find out if the external power is OK or if the external power is intentionally turned off.

If the external power is OK, then a technician can be dispatched to replace the 655A power supply.

---

## Enhanced feature integrations for Avaya Modular Messaging

This enhancement implements QSIG and IP integration for One-Step Recording, supporting integration with Avaya Modular Messaging.

A new button type, **audix-rec**, is added to the **Station** screen for this feature. When administered, the button requires the user's Audix hunt group extension number along with it. The new button type is not yet available on attendant consoles.

---

## Enhanced password security

When you administer a new login, that login now requires a new password the first time it is used.

---

## Enhanced TN2602AP circuit pack

Communication Manager release 3.1 includes enhancements to the TN2602AP circuit pack, described in the following paragraphs.

**Note:**

The TN2602AP IP Media Resource 320 is not supported in CMC1 and G600 Media Gateways. For more information about the TN2602AP circuit pack, see the *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

## Bearer signal duplication

The capabilities of the TN2602AP circuit pack have been expanded to provide duplicated bearer support. This enables customers to administer IP-PNC with critical bearer reliability. A port network continues to support a maximum of two TN2602AP circuit packs, but they can now be administered for duplication. This capability is in addition to the previously-offered load balanced support (see [Load balancing](#)).

Two TN2602AP circuit packs may be installed in a single port network for bearer signal duplication. In this configuration, one TN2602AP is an active IP media processor and one is a standby IP media processor. If the active media processor, or connections to it, fail, active connections failover to the standby media processor and remain active. This duplication prevents active calls in progress from being dropped in case of failure.

Duplicated TN2602AP circuit packs operate in an Active-Standby mode. State of health parameters exist between the two boards to determine when it is appropriate to interchange duplicated TN2602AP circuit packs. It is also possible to manually invoke an interchange using a software command.

For bearer duplication, both TN2602AP circuit packs must be Hardware Version 2, and must have firmware version 211 or higher.

**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.

### **Important:**

If you change from load balanced to duplicated TN2602s, existing calls retain the real IP address on the TN2602AP circuit pack. New calls are associated with the virtual IP address of the TN2602AP circuit pack. If an interchange occurs during this time, existing calls that are associated with the real IP address will drop.

## Load balancing

Up to two TN2602AP circuit packs can be installed in a single port network for load balancing or duplication. When in a load balanced mode, calls are distributed evenly among the two TN2602 circuit packs.

The TN2602AP circuit pack is also compatible with, and can share load balancing with, the TN2302 and the TN802B IP Media Processor circuit packs. Actual capacity may be affected by a variety of factors, including the capacity of the circuit pack being used, the codec used for a call, and fax support.

### **Note:**

If duplicated TN2602 circuit packs are combined with a TN2302 or TN802, Communication Manager uses the active, duplicated TN2602 to capacity before using another media processor circuit pack.

Also, when media processor circuit packs in the same port network are in different network regions, load balancing does not apply.

## Reduced channels with duplicated TN2602AP circuit packs

If a pair of TN2602AP circuit packs, previously used for load balancing, are re-administered to be used for bearer duplication, only the voice channels of the active circuit pack can be used. For example,

- If you have two TN2602 AP circuit packs in a load balancing configuration, each with 80 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you have 80, not 160, channels available.
- If you have two TN2602 AP circuit packs in a load balancing configuration, each with 320 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 320, not the maximum 640, channels available.
- When two TN2602AP circuit packs, each with 320 voice channels, are used for load balancing within a port network, the total number of voice channels available is 640, not 1280. The reason is that 640 is the maximum number of time slots available for connections within a port network.

## Support of T.38 fax relay

T.38 fax relay over IP is now supported with the TN2602AP IP Media Resource 320 circuit pack. The T.38 fax call set up may initiate as a G.711, but once fax tones are detected, traffic is encoded/decoded using the T.38 specification.

In the case of duplicated TN2602 circuit packs that are in an active-standby mode, the system sends this information to the standby circuit pack when an interchange occurs.

## V.32 modem relay

V.32 modem relay is an option that provides a low-bandwidth solution for secure voice terminals on the TN2602AP circuit pack. For customers wishing to use standard data modems, modem passthru is the appropriate solution.

Both modem passthru and V.32 modem relay already exist on the TN2302AP circuit pack, so it is now possible for these two circuit packs to interoperate.

---

## Enterprise Mobility User

Enterprise Mobility User (EMU) is a software-only feature that gives you the ability to associate the buttons and features of your primary telephone to a telephone of the same type anywhere within your company enterprise.

**Note:**

In this document, any telephone that is not the primary telephone is referred to the visited telephone and any server that is not the home server of the primary telephone is referred to as the visited server.

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.
- Communication Manager Release 3.1 and 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 an 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.
- EMU is only supported on self-designating terminals (terminals with button labels) that are downloaded from the Communication Manager server.
- Uniform Dial Plan (UDP).

## How Enterprise Mobility User works

On the dial pad of a visited telephone, a user enters the EMU activation feature access code (FAC), the extension number of their primary telephone, and a security code. 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, the home server:

- 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 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>** 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. EMU does not allow users to associate permissions from the home telephone to the remote telephone.

---

## Enterprise Survivable Server increase

The number of Enterprise Survivable Servers (ESS) that you can administer in one ESS configuration is increased to 63. For more information on Enterprise Survivable Servers, see the *Avaya Enterprise Survivable Servers (ESS) Users Guide*, 03-300428.

---

## Extended survivability to G250 Media Gateway

This feature provides basic survivable call processing functionality on the G250 Analog and IP interfaces. The survivability feature is expanded to provide basic survivability functionality for the:

- G250-DCP analog and DCP interfaces
- G250-DS1 analog interfaces
- G250-BRI analog interfaces

---

## Gateway trunk preference selection

When trunks from more than one Media Gateway (G350, G700, or G250) are in the same trunk group, Communication Manager now “prefers” trunks on the same Media Gateway as the originator.

---

## HTTP server on S8500 Media Server

This feature supports downloading of firmware for IP telephones from Communication Manager using HTTP.

---

## Increased Classes of Restriction

The Classes of Restriction (COR) feature is increased from a total of 96 possible CORs to 996 possible CORs. Classes of Restriction are numbered from 0 to 999, with four CORs - 996, 997, 998, and 999 - reserved by the system. The CORs that are available for the user to assign are from 0 to 995.

---

## Increased 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 currently available for the 2410 (release 2 or newer) and 2420 (release 4 or newer) DCP telephones. Support for the newer 46xx IP telephones may be added in the future.

---

## Increased quantity of NCA TSCs and FTSCs

In an S87x0 Media Server or an S8500 Media Server configuration, you can now have up to 999 NCA TSCs system-wide, and up to 999 NCA TSCs per signalling group. You can also have up to 250 FTSCs.

## Increased trunk members for IP signaling groups

The number of H.323 trunk members in a single signaling group that are supported on the **Trunk Groups** screen is increased from 31 to 255. Users also have the option to administer each trunk group member individually or automatically.

---

## Incremental filesyncs

The system now supports two difference sets for incremental filesyncs, one for LSPs and one for ESSs.

---

## Inter-Gateway Alternate Routing calls over Inter-Gateway Connections

The output of the command `display internal-data s-tab` now includes information about IGAR (Inter-Gateway Alternate Routing) calls using shared connection (“dumbbell”) topology over Inter-Gateway Connections (IGCs). An IGAR call can use up to 120 IGCs.

A new field **igccount** displays the total quantity of dumbbell IGCs used by an IGAR call. Dumbbell information for those IGCs appears on additional pages of the output.

The information listed for each PSTN IGC includes:

- Fabric Type (PSTN)
- Master Trunk Port ID
- Slave Trunk Port ID

The information listed for each IP IGC includes:

- Fabric Type (IP)
- Master Ephemeral IP Address (on a port network or gateway)
- Master Ephemeral Port
- Slave Ephemeral IP Address (on a port network or gateway)
- Slave Ephemeral Port

---

## Listen-only FAC for service observing

The system provides a no-talk, listen-only service observing feature access code (FAC). This FAC does not reserve a second timeslot for potential toggle to talk and listen mode. This feature is for call recording applications that use Service Observing of stations/ACD agents to provide increased call recording capacity by reducing the timeslot usage.

---

## Local ringback administration

A new field is added to the **Trunk Group** screen, allowing the administrator to set if local ringback tone should be sent to a caller.

If the **Apply Local Ringback?** field is set to **y**, and the system does not receive a **PI\_IBI** in **ALERT**, then the system sends a local ringback tone to the caller. The local ringback tone is removed when the system receives a connect, and the channel will cut through.

---

## More BRI Trunk circuit packs

An S8700, S8710, S8500, or S8300 Media Server running Communication Manager can now have up to 250 TN2185 (BRI Trunk) circuit packs.

---

## More Leave Word Calling messages

Communication Manager can now handle up to 12,000 Leave Word Calling (LWC) messages.

---

## More than nine static routes allowed

A radio button is added to the **Set Static Routes** section of the configure server web pages that allows more than nine static routes.

---

## Music on hold played from nearest source

An IP telephone in one network region (the “calling party”) calls another IP telephone in another network region (the “called party”). If the calling party places the called party on hold, the called party hears music from the nearest music source. Assuming that the gateways in both network regions had music installed, music would come from the called party’s gateway.

## Notification about 802.1q changes

If you change the settings for 802.1q on the **ip-network-region** screen, a message now notifies you that the changes will not take effect until the server restarts.

---

## Parameterized data for NSF

The **isdn network-facilities** screen now provides a new column, administrable for user-entered Network Specific Facility (NSF) names. The value in this column indicates whether the NSF handles parameterized data. The default value is **n**, but you can change it to **y**.

When this value is set to **y** for **outwats-bnd** or any user-administered NSF name, you can see the **Parm** column on the route-pattern screen. The value in this column provides information for handling the parameterized data. For instance, if the NSF is SCOCS, it defines the class of service requested for the parameterized data. It is blank by default, but you can give it any numeric value up to 5 digits.

---

## Prepend '+' to calling number

The **SIP Trunk Group** screen now provides the field **Prepend + To Calling Number**. The default setting is **n**. If you set the field to **y**, the character **+** is added at the beginning of the calling number for that trunk group.

---

## Processor Ethernet

The Processor Ethernet (PE) interface is one way of connecting VoIP and IP-based devices to Communication Manager. The PE interface is the appearance of the computer's native ethernet interface inside of the Communication Manager application.

An Enterprise Survivable Server (ESS) and a Local Survivable Processor (LSP) registers with the main server when it is first configured, and every time it receives a file sync from the main server. An ESS server and an LSP uses the PE interface to register with the main server. You do not have to administer the PE interface for registration purposes. The system software enables the use of the PE interface on servers configured as an LSP or and ESS server.

 **CAUTION:**

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.

On simplex main servers, you can administer the server where IP endpoints may register through:

- The PE interface only
- CLAN interfaces only (requires the configuration to have CLANs)
- Either the PE interface or specified CLAN interfaces (requires the configuration to have CLANs) That is, both interfaces must be able to be enabled at the same time with some endpoints registering through the PE interface and some through CLANs.

## Adjuncts

If the main server is a simplex server, adjuncts that connect to the simplex main server using a CLAN can also connect to the simplex main server using the server's PE interface.

Three adjuncts are supported for connectivity to the PE interface of a LSP or an ESS server:

- CMS
- CDR
- Application Enablement Services (AES)

An adjunct link is established between the LSP or the ESS server. Administration that allows dedicated and shared connections between the adjuncts and the servers must allow for the link to remain active at all times. When the LSP or ESS server is not active, the adjunct does not receive data from that server.

For information on how to administer the adjunct link, see the adjunct documentation that is specific to your adjunct.

## H.248 and H.323 registration

The system uses the PE interface on an LSP to register H.248 gateways and H.323 endpoints. Starting with Communication Manager Release 3.1, the use of the PE interface to register H.248 gateways and H.323 endpoints has been expanded to include the simplex main server.

## S8500 Media Servers

With the increased functionality of the PE interface, the role of the S8500 Media Server has been expanded. This expansion includes an S8500 configured as an LSP, and an S8500 configured as a main server in an IP configuration with no port networks.

The PE interface is not supported for H.248, H.323, and adjunct connectivity on a duplex (S8700-series) server.

## QSIG path optimization simplified

Improvements were made to the dial plan to simplify path replacement and diversion with rerouting.

---

## QSIG redirection display is administrable

When a Do-Not-Call (DNC) server authorizes a call through QSIG, the server returns the routing number to the CM using QSIG Redirection. The word “forward” and the secret routing number were displayed on the caller’s telephone display.

QSIG redirection is now administrable so a customer can turn off the information from the caller’s telephone display if they choose.

---

## R2-MFC support on G250 Media Gateway

Add support to the trunks of the G250 Media Gateway for MF Signaling (R2MFC, Russian AMI/MFR) detection, and MF signaling generation, on analog trunks and on E1/T1 trunks (CAS signaling).

---

## Remote upgrades for branch gateways

This allows remote upgrade of software/firmware of the branch office server and media gateway without the need for presence of a technician on site. An existing Communication Manager Server CD-ROM serves as the vehicle for delivering major releases, minor releases, and updates. Communication Manager delivers desired software and firmware to the branch location, assuming there is a USB DVD-ROM/CDRW drive connected to the S8300B in a G250, G350, or G700 Media Gateway.

There are three possible remote upgrade scenarios:

- Upgrades performed over the customer network
- Upgrades performed using Secure Access and Control (SAC)
- Upgrades performed outside of the customer network

**Note:**

A USB modem, connected to the S8300B, is required when the remote upgrades are performed outside of the customer network over an analog PSTN connection.

---

## Rerouting and path replacement by trunk group

You can now administer individual QSIG trunk groups not to use rerouting and path replacement, while leaving these capabilities active for other trunk groups. The **Trunk Group** screen now provides a new page containing two new fields for this purpose:

- **Diversion By Reroute**
- **Path Replacement**

These two new fields are visible only if the **Basic Supplementary Services** and the **Supplementary Services With Reroute** fields on the **Optional Features** screen are set to **y**, and the **Supplementary Service Protocol** field on the **Trunk Group** screen is set to **b**. The default value for both fields is **y**.

- If you set the **Diversion By Reroute** field to **n**, the Call Diversion feature uses forward switching rather than rerouting.
- If you set the **Path Replacement** field to **n**, the **Path Replacement With Retention** and the **Path Replacement Method** fields are no longer visible, and the trunk group does not use path replacement.

---

## Reset IP stations by subnet enhancement

Customers can reset telephones in a multi-floor building by subnet, and perform controlled station resets using the `reset ip-stations` command to reset by subnet.

---

## Secure Shell and Secure FTP

The Secure Shell (SSH) and Secure FTP (SFTP) capabilities are highly-secure methods for remote access. Administration for this capability also allows disabling Telnet when it is not needed, creating a more secure system.

SSH/SFTP functionality does not require a separate Avaya license, nor are there any entries in the existing Communication Manager license needed.

---

## Applicable platforms or hardware

You can log in remotely to the following platforms or hardware using SSH as a secure protocol:

- G350 Media Gateway
- C350 Multilayer Modular switch
- S8300, S8500, S8700, or S8710 Media Server command line

## What's New

- IBM e-server BladeCenter Type 8677 command line
- Communication Manager System Administration Terminal (SAT) interface on a media server using port 5022.

**Note:**

The client device for remote login must also be enabled and configured for SSH. Refer to your client PC documentation for instructions on the proper commands for SSH.

Secure Shell (SSH) and/or Secure FTP (SFTP) remote access protocols are provided on these circuit packs:

- TN799DP (CLAN)
- TN2501AP (VAL)
- TN2312AP/BP (IPSI)
- TN2602AP (Crossfire)

SAT commands enable S/FTP sessions through login/password authentication on the CLAN and VAL circuit packs and SSH/Telnet on the Crossfire circuit pack. The Maintenance Web Interface and a Communication Manager command line enable the IPSI session. Unencrypted Telnet and FTP capabilities are enabled on these circuit packs.

## Using Secure Shell to retrieve backup information

From the Maintenance Web page of Communication Manager, customers can perform a backup such that the file can be either sent or retrieved using SSH/SCP/SFTP.

---

## Security of IP telephone registration/H.323 signaling channel

The Security of IP telephone registration/H.323 signaling channel feature provides a secure mechanism for an H.323 IP endpoint and a Communication Manager gatekeeper to mutually authenticate each other. The IP endpoint and the Communication Manager gatekeeper authenticate each other by implicitly showing that each knows the assigned PIN.

The system uses the procedures of H.235.5, formerly published as H.235 Annex H, Security Profile 1 to accomplish this authentication. The system also used H.235.5 to negotiate a strong shared secret using the Encrypted Key Exchange (EKE) method.

An authentication key, derived from the master key using a one-way function, is used to authenticate the contents of the messages. The H.323 endpoint and a Communication Manager gatekeeper exchange this authentication key during IP registration, admission, and status (RAS) and during call signaling.

An encryption key, derived from the master key using a one-way function, is used to encrypt private information that is carried within these messages. Two examples are media encryption keys and proprietary signaling elements. The proprietary signaling elements carry display information and dialed digits.

If one or the other parties does not possess the correct PIN, the computed shared secrets are different. As a result, RAS message authentication fails and the parties refuse to communicate with each other.

With the Security of IP Telephone Registration/H.323 Signaling Channel feature, the IP endpoint and the Communication Manager gatekeeper:

- Authenticate each other
- Negotiate a strong shared secret
- Authenticate each message that is sent or received
- Digitally sign all RAS and call signaling messages
- Encrypt selected elements of RAS and call signaling messages, such as:
  - media session keys
  - proprietary elements

You can also use H.235.5 procedures and security mechanisms for IP trunking by administering the appropriate **Signaling Group** screen.

With this feature, the quality of the communication includes:

- Privacy for selected elements of call signaling, including media session encryption keys and dialed digits.
- Security of past or future communications, even if one session is penetrated by an attacker with knowledge of that session's keys. This is known as "perfect forward secrecy."
- Efficient reuse of the negotiated strong secret, identified by a unique session ID, to derive strong keys for new signaling links between Communication Manager and endpoints, or between other Communication Manager servers, such as IP trunks.

## Capacities

**S8300 Media Servers** - There are no capacity constraints on S8300 Media Servers when this feature is enabled.

**S8500 Media Servers** - There are no capacity constraints on S8500 Media Servers when this feature is enabled.

**S8700 Media Servers** - Fewer IP endpoints are supported on S8700 Media Servers when this feature is enabled. Full system recovery from a network outage on an S8700 Media Server, with 1,500 IP endpoints and this feature enabled, will take roughly 15 minutes. Therefore, use of this feature on an S8700 Media Server with more than 1,500 IP endpoints is not supported.

**S8710 Media Servers** - Fewer IP endpoints are supported on S8710 Media Servers when this feature is enabled. Full system recovery from a network outage on an S8710 Media Server, with 3,000 IP endpoints and this feature enabled, will take roughly 15 minutes. Therefore, use of this feature on an S8710 Media Server with more than 3,000 IP endpoints is not supported.

## Feature Interactions

**Survivable Remote Processor** - Survivable Remote Processor (SRP) is not supported when this feature is enabled.

**IP Softphone** - If an IP Softphone is in a network region that has the **Security Procedures** field on the **ip-network-region** screen set either to **pin-eke** (H.235.5) or **strong** (which currently is equivalent to **pin-eke**), the IP Softphone will not register. For the IP Softphone to register, the **Security Procedures** field must be set either to **any-auth** or **challenge**.

**Note:**

Currently, IP Softphone does not support **pin-eke** (H.235.5). IP Softphone may support **pin-eke** in a later release.

Use of Avaya IP Softphone in “Control Avaya Telephone (via the telephone)” configuration is possible only if the **Security Procedures** field on the **ip-network-region** screen that applies to the IP Softphone is set to **challenge**.

## Compatibility

The Security of IP Telephone Registration/H.323 Signaling Channel feature requires IP telephone software R2.3 or later.

IP telephone software R2.3 or later requires TN799C hardware, vintage 3 or later, circuit packs. TN799C hardware, vintage 1 and 2 circuit packs, do not work with IP telephone software R2.3.

All versions of TN799DP circuit packs are compatible with IP telephone software R2.3. Therefore, use of the Security of IP Telephone Registration/H.323 Signaling Channel feature requires TN799DP or TN799C hardware vintage 3 or later circuit packs.

---

## Shadowing data on servers

Prior to this update, if a server interchange occurs, a customer might like to review the occupancy of the system prior to the interchange to see if the load might possibly have been a catalyst. The customer must interchange back to the other server in order to review the information.

For example, Server A is active from 13:00 to 15:30 when something causes an interchange. Now Server B becomes active. At 15:40, the customer or service representative tries to review system occupancy using the `list measurement occupancy last` or `list measurement occupancy summary` command.

The only occupancy information that is provided on Server B is data prior to 13:00 when Server B was the active server, and from 15:30 to 15:40.

Now, **list measurement** information is shadowed (shared) between servers so that the customer can see the data.

---

## SIP Enablement Services

Session Initiation Protocol (SIP) is an endpoint-oriented signaling standard. SIP is a text-based protocol based on elements of Hypertext Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP). SIP Enablement Services (SES) supports several types of communication sessions that include voice, video, or instant text messaging.

For a complete description of SES updates and enhancements for release 3.1, including updates to screens and commands, see:

- *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8400, S8500, S8500B, S8700, and S8710 Media Server, 555-245-206*
- *Installing and Administering SIP Enablement Services R3.1, 03-600768*

---

## Site data warning when adding station to TTI port

If you add a station to a TTI port which already has site data, you now receive a warning message informing you that this action affects the site data.

---

## Support caller ID on call waiting for MM711 and MM714

Calling Identity Delivery on Call Waiting (CIDCW) provides visual information on an incoming call while the called party is on an existing call. This feature combines the functionality of Call Waiting (CW) and Calling Identity Delivery (CID).

---

## Support for Enterprise Linux

This feature ports Communication Manager from Red Hat Linux 8.0 to Red Hat Enterprise Linux 4.0.

## Translations file timestamps

The resolution on the translations file timestamps is now adequate to differentiate consecutive incremental filesyncs.

---

## Web firewall settings simplified

The firewall script was changed to generate the list of services that can be administered using the web page. The list of services is maintained in a new firewall configuration (config) file, and the list was reduced to only protocols that are implemented by some command that is installed on the server. This list is in a separate config file to allow on the fly changes (e.g., in the field).

 **CAUTION:**

There is no backup and restore capability, which means changes to this new config file will have to be manually propagated to new loads.

The Firewall web page was changed to handle the new line format from the firewall script, and use that output as the list of services that can be administered instead of /etc/services. The web page was also changed to use the punch/patch option to the firewall script for changing administrations rather than creating a temporary config file.

---

## Web interface for synchronization plan

A customer can now use a web interface to check if their synchronization plan is working correctly.

---

## Web upgrade tool checks file corruption/presence

When attempting a switch/LSP/MG upgrade, if an upgrade file is not present or corrupted, the upgrade will fail. An upgrade to the Web upgrade tool now lists the cause of the failure and what, if any, remaining upgrade tasks failed to begin (i.e., MG FW upgrade).

---

## Web upgrade tool common media module option

The upgrade tool requires entry of file names for G700 and G350 Media Gateways. Some media modules are common to both platforms. The upgrade tool now has the option to use the same file name for common media modules for both gateways.

---

## Release 3.0 new features and enhancements

Avaya Communication Manager, release 3.0, includes the following general telephony and system-wide features and enhancements.

---

### Administrable Periodic Registration Timer

The **IP-Options System Parameters** screen now provides the **Periodic Registration Timer (PRT)** field. This field governs the interval, in minutes, between instances of re-registering endpoints that are unregistered due to moving to another telephone, for example, when IP Softphone has controlled the extension. You can administer the number of minutes to any value from 1 to 60 minutes. The default value is **20**.

---

### Alarm log entries for MG-ICC

When a KA failure happens for an LSP, entries in the alarm log now show MG-ICC instead of ICC.

---

### Analog bearer frequency for IP encoding

The **Signaling Group** screen now provides a new field, **DCP/Analog Bearer Capability**, when the **Group Type** is **H.323**. The default value of this field is **3.1kHz**, but you can also set it to **speech**. If you set the field to **speech**, the bearer channel is available for IP speech encoding to networks that do not support 3.1kHz encoding.

---

### Application Enablement Services

Application Enablement Services (AE Services) provides a connection between applications and Communication Manager. This connection allows development of new applications and new features without having to modify Communication Manager or to expose its proprietary interfaces.

AE Services provides a single common platform architecture for call control, device control, media control, and management. AE Services enables internal Avaya developers and external partners to create powerful applications that harness the extensive Communication Manager feature set.

## What's New

Avaya provides two different AE Services deployment options:

- [Software-only option](#)
- [Bundled server option](#)

The same client applications and software development kits (SDK) can run against both options.

### Software-only option

Avaya provides the AE Services software, which is the AE Services connector server software and the AE Services SDK. The customer obtains the prerequisite hardware, platform software, and third-party software. The customer then installs and maintains all software and hardware.

### Bundled server option

Avaya provides the hardware server and all of the necessary software:

- AE Services connector server software
- AE Services SDK
- Platform
- Third-party software

Avaya service technicians install the hardware and software. If the customer buys a maintenance and/or a service contract, Avaya also provides maintenance and/or service for the system. Otherwise, the customer maintains the system.

### Adjunct Switch Application Interface

Adjunct Switch Application Interface (ASAI) links Communication Manager and adjunct applications. The interface allows adjunct applications to access Communication Manager features and supply routing information to the system.

ASAI is the Avaya recommendation for Computer Telephony Integration (CTI). ASAI is based on the Q.932 protocol.

### CVLAN

CallVisor LAN (CVLAN) is an application programming interface (API) that enables applications to communicate with Communication Manager. CVLAN sends and receives ASAI messages over shared ASAI links on TCP/IP. An application can use ASAI messages to monitor and control Communication Manager resources.

CVLAN software consists of a client component and a server component. The CVLAN client can be installed on a server or on a client workstation. The CVLAN client provides clients with access to the switch using the CVLAN server.

## DEFINITY LAN Gateway

DEFINITY LAN Gateway (DLG) is a software service that tunnels the ASAI call control protocol messages onto IP packets for transport between a customer Computer Telephony Integration (CTI) server or application and Communication Manager.

## Device and media control API

Device and media control API provides a connector to Communication Manager that allows clients to develop applications that provide first party call control. Applications can register as IP extensions on Communication Manager and then monitor and control those extensions.

Device and media control API consists of connector server software and a connector client API library. The connector server software runs on a hardware server that is independent from Communication Manager. That is, device and media control API does not run co-resident with Communication Manager.

Ask your Avaya representative for a complete list of device and media control API documentation.

## System Management Service

System Management Service (SMS) is a web service that exposes management features of Communication Manager to clients. SMS enables its clients to display, list, add, change, and remove specific managed objects on Communication Manager that are available through the OSSI protocol and SAT screens.

SMS is one of the web services that resides on the Application Enablement Services platform (AE Services).

## Telephony Service

Telephony Service (TS) is a web service that exposes basic outbound call control features of Communication Manager. Telephony Service enables its clients to originate an outbound call, drop a call, transfer a call, or conference a party into a call.

Telephony Service is one of the web services that resides on the Application Enablement Services platform (AE Services).

## User Service

User Service provides a common way of administering, retrieving, and programmatically operating on user data. User Service provides a common user store and a programmable interface for products and applications with which to integrate. User Service has a common industry-standard data store (LDAP) as the repository for common user profile data.

User Service has web services as the infrastructure. This infrastructure allows products to integrate with User Service at your schedule. User Service exposes a programmatic SOAP interface that allows clients to write third party applications to utilize its functionality.

This integration occurs through the use of software adapters to User Service. The adapter and web services technology allows User Service to publish user events to the product spaces, and the product spaces to publish events to the common user area.

So if an administrator adds a user to the common store, a user event is sent to all participating products with the appropriate information. Likewise, if a product level administrator modifies a user record in its own user system, an event is sent to User Service for the modified data to be stored in the common store. User Service then relays this user event to the other participating product areas.

---

## Auto fallback to primary for H.248 media gateways

This feature automatically returns a fragmented network, where a number of H.248 media gateways are being serviced by one or more Local Survivable Processors (LSP), to the primary media server. This feature is targeted to H.248 media gateways only.

This feature allows the administrator to define any of the following rules for migration:

- Whether or not the media gateways, serviced by LSPs, should automatically migrate to the primary media gateway.
- Whether or not the media gateway should migrate immediately when possible, regardless of active call count.
- Whether or not the media gateway should only migrate if the active call count is 0.
- Whether or not the media gateway should only be allowed to migrate within a window of opportunity, by providing day of the week and time intervals per day. This option does not take call count into consideration.
- Whether or not the media gateway should be migrated within a window of opportunity by providing day of the week and time of day, or immediately if the call count reaches 0. Both rules are active at the same time.

Internally, the primary call controller gives priority to registration requests from those media gateways that are currently not being serviced by an LSP. This priority is not administrable.

For more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

---

## Button pushes in list trace station command

The `list trace station n` command, where `n` is the station that you want to trace, now provides information on button pushes at the traced station.

---

## Connection preserving failover/failback for H.248 media gateways

The Connection Preserving Migration (CPM) feature preserves existing bearer (voice) connections while an H.248 media gateway migrates from one Communication Manager server to another. Migration might be caused by a network or server failure.

Only stable calls are preserved. Call that are not preserved are:

- Unstable calls. An unstable is any call where the call talk path between parties has not been established, or is not currently established. Some examples are:
  - Calls with dial tone
  - Calls in dialing stage
  - Calls in ringing stage
  - Calls listening to announcements
  - Calls listening to music
  - Calls on hold (soft, hard)
  - Calls in ACD queues
  - Calls in vector processing
- IP trunks, both SIP and H.323
- ISDN-BRI telephones
- ISDN-BRI trunks

Users on connection-preserved calls cannot use such features as Hold, Conference, or Transfer.

For more information, see *Maintenance Procedures for Avaya Communication Manager 3.0, Media Gateways and Servers*, 03-300432.

## Connection preserving upgrades for duplex servers

The connection preserving upgrades for duplex servers feature provides connection preservation on upgrades of duplex servers for:

- connections involving IP telephones
- connections involving TDM connections on port networks
- connections on H.248 gateways
- IP connections between port networks and media gateways

Stable calls are preserved. Unstable calls are dropped.

**Note:**

This feature applies to upgrades from Communication Manager release 3.0 to a future release of Communication Manager. This feature does not apply to upgrades from a previous release of Communication Manager to Communication Manager release 3.0.

---

## Disable active logins

In the **Days To Disable After Inactivity** field on the **Login** screen, you can specify the number of days of inactivity, from 1 to 180, before the login is automatically disabled. The default value is blank, which means that the disabling feature is turned off for this login.

---

## Display for bridged no-ring calls

The **Feature-Related System Parameters** screen now provides a new field, **Display Information With Bridged No Ring Call**. The default value for this field is **n**. If you set this field to **y**, the caller's name and number appear on the display of a telephone receiving a bridged no-ring call.

---

## E-mail backups no longer supported

The email option for performing server backups on an S8X00 Media Server is no longer supported.

---

## Emergency calls from unnamed IP endpoints

With the Emergency calls from unnamed IP endpoints feature, an IP telephone can register without an extension number. The Emergency calls from unnamed IP endpoints feature places the IP telephone into Terminal Translation Initialization (TTI) service. Users can dial a feature access code (FAC) to either associate an extension number with a telephone, or to dissociate an extension number from a telephone.

If Communication Manager is appropriately administered, a user can use an IP telephone that is in TTI service to make emergency or other calls.

The Emergency Calls from Unnamed IP Endpoints feature requires IP telephone software R2.3 or later, and TN799C hardware circuit packs vintage 3 or later.

### Note:

Turning on the Emergency Calls from Unnamed IP Endpoints feature turns off the re-registering of endpoints that are unregistered. An example is when an extension is moved to another telephone when using IP Softphone (IP Softphone has control of the extension).

---

## Enhanced quality for Music On Hold

Music On Hold delivered using the G.711 codec by a source that is not on the same media gateway or port network as the caller has better quality than if delivered using the G.729 codec. To ensure using the G.711 codec, set the **Always Use G.711 (30ms, No SS) For Intra-Switch Music-On-Hold** field on the **IP-Options System Parameters** screen to **y**.

---

## Enterprise Survivable Servers

The Enterprise Survivable Server (ESS) provide survivability by allowing backup servers to be placed in various locations in the customer network. The backup servers supply service to port networks in the case where the S8500-series media server, or the S8700-series media server pair fails, or connectivity to the main server or server pair is lost.

In an ESS environment, there can only be one main server, either one S8500-series media server, or one pair of S8700-series media servers. If the main server is an S8500-series media server, all ESSs in the configuration must also be S8500-series media servers. During normal operation, the main server communicates with and controls all the port networks. The main server contains a license file that identifies the server as the main server and activates the ESS functionality.

For more complete information on ESS, see the *Avaya Enterprise Survivable Server (ESS) Users Guide*, 03-300428.

## Enterprise Wide Licensing

Enterprise Wide Licensing (EWL) is a technology within Communication Manager release 3.0 and Remote Feature Activation (RFA). EWL is used to partially support a developing offer known as Enhanced Software License Program (ESLP). ESLP gives customers the ability to bulk purchase and then share license capacities across multiple locations.

**Note:**

Launch of the ESLP offer is to be announced at a later date.

---

## Expanded Meet-me Conferencing

Use the Expanded Meet-me Conferencing application to set up multi-party conferences consisting of more than six parties. The Expanded Meet-me Conferencing application supports up to 300 parties.

The actual number of ports that are available for Expanded Meet-me Conferencing is determined by the Communication Manager license file. The license file allocates ports in groups of 50. For example, the license file can allocate 50, 100, 150, 200, 250, or 300 ports. The maximum number of parties included in any single conference is administered with Communication Manager.

The Expanded Meet-me Conferencing application requires an external Avaya Expanded Meet-me Conferencing S8500B Server. For more information, see the Expanded Meet-me Conferencing section of the *Feature Description and Implementation for Avaya Communication Manager*, 555-245-205. See also [Avaya Expanded Meet-me Conferencing Server](#) on page 60.

---

## Extension to Cellular

In addition to features in previous releases of Communication Manager and Extension to Cellular, Extension to Cellular enhancements for release 3.0 include the following:

- The Self Administration Feature (SAFE) Access Code allows users to self-administer their cell phone number. Users can add or change their cell phone number through this feature access code. An administrator can still enter or change cell phone numbers.
- A barge-in tone increases security for Extension to Cellular calls. The tone alerts parties on an active Extension to Cellular call when another person joins the call on the Extension to Cellular associated office phone. The barge-in tone works only when the exclusion feature is not activated.

- Calls are redirected by Redirect on OPTIM Failure (ROOF) procedures if a call terminates to an Extension to Cellular user cell phone (of any application type) that does not have an associated office phone (or other method of terminating to a physical phone) or the office phone is out of order and Communication Manager receives a disconnect prior to the call being answered. The procedure redirects the call so that the caller does not hear ringback forever. In general, the ROOF procedures apply to busy treatment.
- Three new Feature Name Extensions are added:
  - Automatic Call Back allows a user to choose whether they want an extension to automatically call them back. Users are called back when they place a call to a busy or unanswered telephone and after the called telephone becomes available to receive a call.
  - Call Pickup Extended Group allows a user to answer calls that were directed to another call pickup group.
  - Whisper Page Activation allows a user to make whisper pages. A whisper page is a low volume message. Users can send a whisper page when they want only one person on a conference call to hear a message.

For more information, see the following documents:

- *Avaya Extension to Cellular and Off-Station PBX (OPS) Installation and Administration Guide, 210-100-500*
- *Avaya Extension to Cellular User Guide, 210-100-700*

---

## Improved button downloads for IP telephones

This feature greatly reduces the time that the system needs to send button labels to an IP telephone upon a new registration. This feature also reduces system traffic that is sent to IP telephones upon a re-registration when button labels have not changed.

This improvement applies to IP telephones that are running 3.0 firmware. This feature does not apply to DCP telephones.

---

## Improved voice mail coverage at WAN failure

When a Wide Area Network (WAN) failure occurs and a Local Spare Processor (LSP) takes over, Communication Manager now includes called-party information in voice mail.

## Increased packet size supported

On the **IP Codec Set** screen, Communication Manager supports packets of up to 120 milliseconds for voice and modem-relay calls using codec G.729, G.723-5.3K, or G.723-6.3K.

---

## Integrating IP-connected port networks with direct/multi-connect configurations

Starting with Communication Manager release 3.0, the system allows configurations that mix IP-connected port networks with existing center stage switches (CSS), asynchronous transmission mode (ATM) networks, and direct-connected port networks.

Communication Manager allows the S8700-series media servers to support configurations that combine IP-connected port networks (PNs) with direct-connect PNs, CSS-connected PNs, or ATM-connected PNs.

Additionally, Communication Manager allows the media servers to support configurations that contain both single control networks and duplicated control networks, and both single bearer networks and duplicated bearer networks.

A Communication Manager configuration can contain a mix of the following port network connectivity (PNC) methods:

- IP-connect and direct-connect
- IP-connect and Center Stage Switch (CSS)
- IP-connect and Asynchronous Transmission Mode (ATM)

**Note:**

You cannot mix CSS and ATM port network connections in the same configuration. You also cannot mix direct-connect PN connections with ATM or CSS port network configurations.

---

## Inter-Gateway Alternate Routing

Inter-Gateway Alternate Routing (IGAR) is designed for single-server systems that use the IP-WAN to connect bearer traffic between port networks or media gateways. IGAR provides a means of alternately using the public switched telephone network (PSTN) when the IP-WAN is incapable of carrying the bearer connection.

IGAR may request that bearer connections be provided by the PSTN under the following conditions:

- The number of calls allocated or bandwidth allocated via Call Admission Control-Bandwidth Limits (CAC-BL) has been reached
- VoIP RTP resource exhaustion in a MG/PN is encountered
- A codec set is not specified between a network region pair
- Forced redirection between a pair of network regions is configured

IGAR takes advantage of existing public and private network facilities provisioned in a network region. Most trunks in use today can be used for IGAR. Examples of the better trunk facilities for use by IGAR are:

- Public or Private ISDN PRI/BRI
- R2MFC

IGAR provides enhanced Quality of Service (QoS) to large distributed single-server configurations.

For more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

---

## List IP addresses for IP interface circuit packs

The command `list ip-interface` now provide the IP address that is associated with each IP interface circuit pack.

---

## Locally sourced announcements and music

Use the Locally Sourced Announcements and Music feature to access announcement and music audio sources on a local port network or media gateway.

Locally sourced audio can:

- Improve the quality of audio
- Reduce resource usage, such as VoIP resources
- Provide a backup mechanism for announcement and music sources

## MLPP privileges at any endpoint

You can now activate the MLPP privileges specified on the **Multiple Level Precedence & Preemption Parameters** screen at any endpoint by entering a special code. To administer this code, set the **Limited Line Load Control Origination** field on the **Multiple Level Precedence & Preemption Parameters** screen to **y**. Then set the **Authorization Access Code** field on the MLPP Features page of the **Feature Access Codes (FAC)** screen to the desired code.

---

## Modem over IP

The modem over IP (MoIP) feature allows for transport of data over a 64kbps unrestricted clear channel. Starting with Communication Manager release 3.0, when a clear channel data call is originated, the system communicates to the media processor or VoIP engine to allow a 64kbps clear channel to be opened for transport.

For more information, see *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

---

## More options for changing display messages

You can now change user-defined static display messages not only when you type the `change display-messages` command, but also by editing the files `/etc/opt/defty/i18n/avaya_user-defined.txt` and `/etc/opt/defty/i18n/custom_user-defined.txt`.

---

## More system-wide message retrieval extensions

The number of extensions on the **Feature-Related System Parameters** screen, that can have the System-Wide Message Retrieval capability on an S8710, S8700, S8500, or S8300 Media Server, has increased from 10 endpoints to 30 endpoints.

---

## Multiple SNMP trap destinations

The bash command `almsnmpconf` now supports multiple SNMP trap destinations.

---

---

## Native support of NI-BRI data

Communication Manager supports U.S. national ISDN (NI-1 and NI-2) NI-BRI endpoints, as well as NI-BRI data. This feature meets the criteria that is set by the U.S. Government Department of Defense to be certified as a small end office (SMEO) solution.

This enhancement also allows the S8300 and S8700 media servers, with G350 and G700 H.248 media gateways, to support BRI endpoints.

---

## Prevent MLPP preemption of emergency calls

You can now prevent emergency calls from being preempted by the Multiple Level Precedence & Preemption (MLPP) feature. To do so, set the **Pre-Empt Emergency Call** field on the **Multiple Level Precedence & Preemption Parameters** screen to **n**.

---

## QSIG support for Unicode

The QSIG support for Unicode feature extends the Unicode support on a single server to multi-node Communication Manager networks. This feature allows Unicode support across large campus configurations.

Many configurations contain multiple Communication Manager servers due to scalability requirements. This feature also allows Unicode support across large corporate networks, frequently multinational corporations, where multiple Communication Manager servers are almost always provisioned.

---

## RAM disk for S8300 Media Server

RAM disk capabilities are now available on an S8300 Media Server.

---

## Remove assigned DID

In the Property Management System (PMS), you can now remove the direct inward dial (DID) number that is assigned to a room using a VIP message without going through the checkout process.

## Ringback during coverage interval

The **Hunt Group** screen now contains the **Provide Ringback** field. This field is available if the value of the **Message Center** field is **qsig-mwi**.

If you set the **Provide Ringback** field to **y**, a call covering to the message center provides ringback to the caller during the coverage interval.

---

## Secure Shell and Secure FTP for circuit packs

This development addresses customers security concerns by adding Secure Shell (SSH) and and secure FTP (SFTP) remote access protocols to these circuit packs:

- TN799DP (CLAN)
- TN2501 (VAL)
- TN2312AP/BP (IPSI)

New SAT commands enable SFTP sessions through login/password authentication on the CLAN and VAL circuit packs. The Maintenance Web Interface and command line interface (CLI) continue to enable the IPSI session. The Telnet and FTP capabilities remain enabled on these circuit packs.

---

## Security of IP telephone registration/H.323 signaling channel

**Note:**

Please check with your Avaya Sales Representative or your Avaya Authorized Business Partner for availability of this feature.

This feature provides a secure mechanism for an H.323 endpoint and a Communication Manager gatekeeper to mutually authenticate themselves and the contents of the messages that they exchange during IP registration, admission, and status (RAS). This authentication is based on the simple 3-to-8 digit PIN administered for each extension, the H.235 Annex H SP1 Execution of Encrypted Key Exchange (EKE) procedures during RAS results in the negotiation of a strong secret that is shared between the endpoint and the gatekeeper. This strong secret is used to derive a set of secrets which are used to digitally sign all RAS and call signaling messages, and to encrypt selected elements of call signaling messages, media session keys and CCMS messages. If one or the other parties does not possess the correct PIN, the computed shared secrets will, in fact, be different. Message authentication fails, and the parties refuse to communicate.

In summary, these procedures permit:

- The endpoint and the gatekeeper to authenticate each other;
- The endpoint and the gatekeeper to sign/authenticate each message sent;

- Privacy for selected elements of call signaling, including media session encryption keys and dialed digits.
- Security of the endpoint/gatekeeper communication even if an observer obtains the user's PIN.
- Security of past or future communications even if one session is penetrated by an attacker with knowledge of the PIN. (This is known as "perfect forward secrecy".)
- Reuse of the negotiated strong secret (identified by a unique session ID) to secure new signaling links between parties for re-registration or trunking.

---

## Serial number for license validation

The **License Serial Numbers** screen, that you can access with the `list configuration license` command, displays the serial number for validating the server license.

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## Shorter time-out for list trace ras command

The `list trace ras` command from OSSI times out if it does not complete in 2 minutes, rather than 15 minutes.

---

## Station licensing

Prior to release 2.0, Communication Manager was sold by licensed ports that included stations and trunks. The system displayed the total of licensed ports in the **Maximum Ports** field on the **Optional Features** screen.

As of release 2.0 of Communication Manager, Avaya sells licenses for stations, but not trunks. Currently, the **Maximum Ports** field on the **Optional Features** screen is used for licensing ports, which include both trunks and stations.

With Communication Manager release 3.0, a separate field, **Maximum Stations**, is created on the [Optional Features](#) screen to track station licenses only. Customers can easily identify the number of station licenses on the system.

---

## User-defined phone message files

The Communication Manager Phone Message File web page (formerly the Unicode Message File web page) now shows user-defined files currently installed or available for installation.

---

## Web interface from multiple IP addresses

You can now choose to enable access to the Web interface to Communication Manager from multiple IP addresses.

By default, as a security measure, only the IP address that first accesses the interface has permission to do so again. However, you can now disable this restriction with the SAT command `webchkip disable`. To re-enable the restriction, use the SAT command `webchkip enable`. To see the current status of the restriction, use the command `webchkip` by itself.

# Chapter 2: Hardware

This chapter presents highlights of any hardware as part of the most current releases of Avaya Communication Manager running on Avaya DEFINITY® servers, as well as the Avaya S8000 series Media Servers (with associated Avaya Media Gateways).

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## Release 3.1 hardware additions

Avaya Communication Manager, release 3.1, includes the following general hardware additions.

---

### G250 DCP Media Gateway

The G250 DCP Media Gateway supports:

- Four analog trunks, Loop Start only (no support for Ground Start or CAMA)
- Two analog stations and/or DID trunks.
- Twelve DCP ports
- Two Ethernet LAN ports
- One 10/10 Ethernet WAN port
- One expansion slot for an ACM server module
- One expansion slot for a data WAN media module
- One console RS232 interface
- One USB host interface
- One contact closure relay control
- ETR

### USB support for G250 Media Gateway

Through the USB port on the front panel of the G250 Media Gateway, you can connect external modems for WAN dial-up backup.

---

## G250 DS1 Media Gateway

The G250 DS1 Media Gateway supports:

- One T1/E1/PRI trunk with fractional trunks allowed
- One analog trunk with loop start only (no support for ground start or CAMA)
- Two analog lines and/or DID trunks (one with ETR)
- ETR
- Eight Ethernet LAN PoE ports
- 10/100 Ethernet WAN
- One expansion slot for an ACM server module
- One expansion slot for a data WAN media module
- One console RS232 interface
- One USB host interface
- One contact closure relay control

---

## MM316 HDMM for G350 Media Gateway

The MM316 HDMM Media Module for the G350 branch gateway supports 40 10M/100M PoE ports with 1G/100 Copper port. The MM316 HDMM allows supporting up to 40 IP endpoints connected to the G350 Media Gateway with no external switch power.

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## MM716 analog media module

The MM716 Analog Media Module is a 24 port analog circuit pack. All ports are dedicated for station (line). The station (line) ports support an LED Message Waiting Indicator, and are used for DID trunk (wink or immediate start) port connections.

The MM716 is supported in the G350 and G700 Media Gateways. Communication Manager 3.0 or higher, and equivalent gateway software, is necessary to support the MM716.

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## S8400 Media Server

The S8400 Media Server is a Linux-based server that occupies a single slot in a standard TN carrier. The S8400 Media Server provides Communication Manager processing functionality in stand alone, single port network (PN), telephony systems requiring up to 900 stations.

The S8400 Media Server is composed of the:

- TN8400AP Media Server circuit pack
- TN8412AP S8400 IP Interface (SIPI) circuit pack

The S8400 Media Server can replace the:

- CSI
- DEFINITY One/S8100
- IP600/S8100

The S8400 Media Server consists of a TN8400 hardware/software platform, Communication Manager, and optionally, IA770 messaging software.

For new installations, the PN uses the G650 Media Gateway. In current installations, the S8400 Media Server is used as an upgrade path for a current PN based on G650 and G600 Media Gateways and CMC carriers.

## TN8412AP circuit pack

The S8400 Media Server uses the TN8412AP S8400 IP Interface (SIPI) circuit pack to provide:

- Circuit pack control within its port network
- Cabinet maintenance
- Tone-clocks
- Emergency transfer switch functionality
- Customer/external alarms.

**Note:**

An S8400 system is shipped with a TN8412AP SIPI circuit pack. However, the TN2312BP IPSI circuit pack is also compatible with S8400 systems.

The TN799DP Control-LAN (C-LAN) circuit pack provides firmware download functionality while the TN2501 Voice Announcement over LAN (VAL) circuit pack provides announcement functionality.

The S8400 Media Server provides a Voice over Internet Protocol (VoIP) based integrated messaging capability for up to 450 light duty users. This option requires 8 ports of VoIP resources be provisioned with the S8400 Media Server. The hard disk drive (HDD) stores the messages and a TN2302AP IP Media Processor circuit pack usually provides the VoIP resources.

### S8720 Media Server

The S8720 (HP DL 385) is the next generation S87XX-series media server, succeeding the S8710 media server. Most aspects of the S8720 are the same as the S8710, including:

- Hardware architecture
- Interfaces
- Call flow
- maintenance
- Serviceability
- Software initialization, recovery, redundancy
- Availability, reliability, and survivability strategies

Differences between the S8720 and S8710 include:

- The S8720 has greater call processing performance than the S8710.
- Software duplication is available on the S8720. The S8720 is ordered without the DAL1 card used for hardware duplication. Hardware duplication, and the DAL1 card, can be purchased as an option.
- If software duplication is used with the S8720, the functions of the Eth0 and Eth2 interfaces are reversed with respect to the hardware duplication functions.
- The S8720 has an additional (third) USB port located on the front panel.
- There are minor differences in the mounting rails and how the rails attach to the S8720 and the rack.

### Software Duplication

The software duplication feature eliminates the need for the DAL1 memory duplication cards in duplicated S8720 servers. This feature is supported only on the S8720 media server.

The S8720 is shipped without the DAL1 hardware duplication card. Hardware duplication is an option for the S8720. If purchased, the DAL1 hardware duplication cards, and the dual fiber cable linking the DAL1 cards, are installed in the S8720 media servers at the customer site. The duplication type (hardware or software) is administered as a Configure Server step on a new Server Duplication web page in the Maintenance Web Interface.

For software duplication, memory duplication messages are sent over the server duplication TCP/IP link. This link uses a dedicated or non-dedicated TLS connection and TCP port 5098. Memory duplication messages can optionally use AES encryption.

**⚠ CAUTION:**

Software duplication degrades the performance of the communications system as compared with hardware duplication. Encrypting duplication messages will further degrade performance. For software duplication, Avaya recommends a dedicated duplication link. If the duplication link is routed or switched, the link should have a bandwidth of 1 Gigabit per second.

---

## Release 3.0 hardware additions

Avaya Communication Manager, release 3.0, includes the following general hardware additions.

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### 4621SW IP telephone

The 4621SW IP telephone is based on the 4620SW IP telephone. The 4621SW IP telephone has a large screen with an adjustable backlit graphic display. The 4621SW IP telephone does not support IR interface.

The 4621SW IP telephone has the following features:

- The same interface as the 4620SW IP telephone
- A backlight that can be adjusted by the user. The backlight can be administered to turn off when the telephone is idle, or it can stay lit continuously.
- Native support that gives the customer the ability to administer and maintain the telephone without using an alias
- The 4621SW IP telephone supports the EU24BL adjunct. The EU24BL is the same as the EU24 adjunct, except that the EU24BL adjunct has a backlit display.

---

### 4622SW IP telephone

The 4622SW IP telephone is based on the 4620SW IP telephone. The 4622SW telephone provides the same advanced feature functionality, with an intuitive and innovative user interface as the 4620SW IP telephone. The 4622SW telephone is designed for the call center environment. The 4622SW IP telephone does not support IR interface.

The 4622SW has the following features:

- Two headset jacks
- A large screen backlit graphic display

## Hardware

- A backlight that can be adjusted by the user. The backlight can be administered to turn off when the telephone is idle, or it can stay lit continuously.
- The telephone stand has one extra height setting. This setting is the same as the highest setting for the 4610SW telephone.
- Native support that gives the customer the ability to administer and maintain the telephone without using an alias
- The 4622SW IP telephone supports the EU24BL adjunct. The EU24BL is the same as the EU24 adjunct, except that the EU24BL adjunct has a backlit display.

---

## 4625SW IP telephone

The 4625SW IP telephone is similar to the 4620SW IP telephone. The 4625SW provides the same advanced feature functionality with an intuitive and innovative user interface as the 4620SW IP telephone. The 4625SW telephone provides telephony, speed dial, call log, and Web browsing functionality. The 4625SW IP telephone has all of the applications and options of the 4620SW IP telephone.

The 4625SW has the following features:

- A color 1/4-VGA backlit display
- Native support that gives the customer the ability to administer and maintain the telephone without using an alias

The 4625SW IP telephone does not support:

- Multi-byte characters
- Multi-byte user interface languages
- An IR interface

---

## Avaya Expanded Meet-me Conferencing Server

The Avaya Expanded Meet-me Conferencing Server is an S8500B server that connects to a Communication Manager server over the customer LAN to provide [Expanded Meet-me Conferencing](#). Expanded Meet-me Conferencing supports a conference bridge of up to 300 ports. This increase in the number of parties on a conference is much greater than the limit of a 6-port conference bridge on a Communication Manager system without Expanded Meet-me Conferencing.

All SIP-enabled media servers can use the Avaya Expanded Meet-me Conferencing Server and the Expanded Meet-me Conferencing feature of Communication Manager. All Communication Manager telephones can use Expanded Meet-me Conferencing. However, for SIP telephones and SIP softphone to be able to use Expanded Meet-me Conferencing, the Communication Manager configuration also requires the Converged Communication Server (CCS).

---

## Converged Network Analyzer

The Converged Network Analyzer (CNA) is a distributed system for real-time monitoring of IP networks. CNA uses active measurements. CNA supports various network tests including connectivity tests with pings, topology tests with traceroute, and QoS tests with synthetic RTP streams.

Within a CNA system, test plugs are the entities that execute the tests according to instructions from CNA schedulers, and return the results. For more information about administrating the CNA system, see *IM R3.0 Converged Network Analyzer (CNA) Configuration*.

---

## DNS Resolver for gateways

The G350/G250 supports the DNS Resolver application, as part of VPN enhancements. A DNS Resolver resolves host names to IP addresses. It does this by querying DNS servers according to an ordered list. The list of DNS servers is compiled using either DNS servers entered manually by the user, or DNS servers gathered automatically by means of DHCP or PPP protocols, or both.

The user can also optionally aid the DNS Resolver by specifying a list of domain names that the DNS Resolver adds as a suffix to non-FQDN names, to help resolve them to an IP address. The DNS Resolver feature is intended to provide a backup mechanism for VPN hubs using DNS. For more information see the *Technical Summary for G350/G250 R3.0 VPN*.

---

## G250 Media Gateway

The G250 Media Gateway is designed for very small branch offices (2-8 users) of large enterprises that seek a centrally-managed branch solution, a seamless element of the enterprise's communication network.

The G250 is available in two different models:

- G250-Analog
- G2500-BRI

Each model has a different front-panel configuration based on the interface(s) necessary the the endpoints that connect to the two media modules that it houses. Each of these G250 models provides:

- Analog and IP telephony
- Local connectivity to the PSTN
- LAN/PoE infrastructure
- Secured WAN connectivity to the enterprise headquarters

## Hardware

### Note:

The G250 Media Gateway supports legacy analog telephones, but not DCP telephones.

G250 is a gateway subtending to a Communication Manager server located at a larger branch or at the enterprise headquarters. This means that the gateway gets its call control from the Communication Manager server in its intermediary position between the server and the endpoints that it serves. The WAN connection between the small office (gateway and endpoints) and the server can be over a public IP network and/or private network (frame relay, leased line, etc.).

The G250 Media Gateway has two survivability options:

- Basic local survivability functionality, or Standard Local Survivability that is resident in the media gateway firmware
- Full survivability with a S8300B server as its Local Survivable Processor (LSP), offering complete Communication Manager functionality.

Gateways that are connected to a non-private network have these security features:

- IPSec VPN capabilities for up to 3 tunnels for signal and/or bearer information to the enterprise headquarters
- Pre-configured firewall for general access to public IP networks

For more information on the G250 Media Gateway, see the *Overview for the Avaya G250 and the Avaya G350 Media Gateways*, 03-300435.

## Standard Local Survivability

Standard Local Survivability (SLS) provides a local G250 Media Gateway with a limited subset of Communication Manager functionality when there is no IP-routed WAN link available to the main server.

### Note:

SLS is not supported on the G250-BRI.

SLS provides:

- Outbound dialing through the local PSTN (local trunk gateway) from analog and IP phones
- Inbound calls from each trunk to pre-configured local analog or IP phones that have registered
- Local call progress tones (dial tone, busy, etc.)
- Emergency Transfer Relay (ETR) in survivable mode on the media gateway hardware in cases of power loss

---

## SP-1020A SIP business telephone

The SP-1020A SIP business telephone, also known as the Toshiba® SIP handset, communicates with Communication Manager through a SIP trunk group. The SP-1020A SIP business telephone requires Communication Manager release 3.0 or later.

The SP-1020A SIP business telephone is designed by Toshiba initially for the Japanese market, but has long-range implications for other markets as well.

The SP-1020A SIP business telephone has:

- 24-feature buttons
- duplex speakerphone
- 4-line display supporting English and Japanese characters
- Menu key and navigation keys for operations, including a 20-entry call log
- Full SIP-compliant signaling with these expanded features:
  - automatic callback
  - whisper page
  - speed dial list
  - send all calls
  - message waiting indication and full integration with Avaya voice mail systems
  - VIP priority calling
  - Call pickup - directed, group, and extended
  - Distinctive alerting - priority, internal, external, and intercom
  - Intercom calling
  - Hold recall
  - Transfer recall
- Other features include:
  - Automatically downloadable “personality” including volume levels, button placement, and ringer settings to provide true mobility from handset to handset across the network
  - IEEE standard power over Ethernet
  - Terminal upgrades

---

## TN2602AP IP Media Resource 320 circuit pack

The TN2602AP IP Media Resource 320 circuit pack provides high-capacity voice over Internet (VoIP) protocol audio access to the switch for local stations and outside trunks. The TN2602AP IP Media Resource 320 circuit pack provides audio processing for the following types of calls:

- TDM-to-IP and IP-to-TDM
- IP-to-IP

The TN2602AP IP Media Resource 320 circuit pack has two capacity options, both of which are determined by the license file installed on Communication Manager:

- 320 voice channels, considered the standard IP Media Resource 320
- 80 voice channels, considered the low-density IP Media Resource 320

Up to two TN2602AP circuit packs may be installed in a single port network for load balancing. The TN2602AP circuit pack is also compatible with and can share load balancing with the TN2302 IP Media Processor circuit pack. Actual capacity might be affected by a variety of factors, including the codec that is used for a call and fax support.

**Note:**

The TN2602AP IP Media Resource 320 is not supported in CMC1 and G600 Media Gateways.

The TN2602AP IP Media Resource 320 circuit pack supports hairpin connections and shuffling of calls between TDM connections and IP-to-IP direct connections. The TN2602AP can also perform the following functions:

- Echo cancellation
- Silence suppression
- Adaptive jitter buffer (120 ms)
- Dual-tone multifrequency (DTMF) detection
- AEA Version 2 and AES media encryption
- Conferencing
- 802.1p and 802.Q layer 2 QoS mechanisms
- RSVP protocol

The TN2602AP IP Media Resource 320 circuit pack supports the following codecs for voice, conversion between codecs, and fax detection:

- G.711, A-law or Mu-law, 64 kbps
- G.726A-32 kbps
- G.729 A/AB, 8 kbps audio

The TN2602AP also supports transport of the following devices:

- Fax, Teletypewriter device (TTY), and modem calls over a corporate IP intranet using pass-through mode
- 64-kbps clear channel transport in support of firmware downloads, BRI secure telephones, and data appliances
- Fax and TTY calls using proprietary relay mode

**Note:**

The path between endpoints for fax and modem transmissions must use Avaya telecommunications and networking equipment. T.38 fax is not supported.



**SECURITY ALERT:**

Faxes sent to non-Avaya endpoints cannot be encrypted.



# Chapter 3: New and changed screens

This chapter displays the new and changed administration screens for Avaya Communication Manager.

---

## Release 3.1 new screens

Avaya Communication Manager, release 3.1, includes the following new screens. For a more complete explanation of the screens and their function, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

---

### Enable Session

Use this screen to enable a SSH instead of a Telnet session. This screen was prompted by the [Secure Shell and Secure FTP](#) feature.

To view the **Enable Session** screen:

1. Type `enable session`. Press **Enter**.

The system displays the **Enable Session** screen ([Figure 1: Enable Session screen](#) on page 67).

---

**Figure 1: Enable Session screen**

```
enable session a03                                     Page 1
                                                    ENABLE SESSION
      Login:_____
      Password:_____
      Password:_____
      Secure?
      Time to Login:
```

## New and changed screens

### Field descriptions

**Login** - Enter your login.

Valid entries	Usage
3 to 6 alphanumeric characters	Enter your login.

**Password** - Enter your password.

Valid entries	Usage
7 to 11 characters, with at least one number	Enter your password.

**Password** - Repeat your password for verification. Entry must be identical in both **Password** fields.

Valid entries	Usage
7 to 11 characters, with at least one number	Repeat your password for verification. Entry must be identical in both <b>Password</b> fields.

**Secure** - Enter **y** to indicate that SSH is used instead of Telnet.

Valid entries	Usage
<b>y/n</b>	Enter <b>y</b> to indicate that SSH is used instead of Telnet. Default is <b>y</b> .

**Time to Login** - This field appears only if the board in question is a TN2302.

Valid entries	Usage
<b>0 to 255</b>	Enter the number of minutes to login before the session times out. Default is blank.

---

## Survivable Processor

Use the **Survivable Processor** screen to add information specific to a Local Survivable Processor (LSP) or an Enterprise Survivable Server (ESS). Before administering this screen, you must first assign node names for each LSP and ESS media server on the **IP Node Names** screen. This screen was prompted by the [Processor Ethernet](#) feature.

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).

**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.

To view the **Survivable Processor** screen:

1. Type `add survivable-processor node-name`. Press **Enter**.

The system displays the **Survivable Processor** screen ([Figure 2: Survivable Processor - Processor Ethernet screen](#) on page 69).

---

**Figure 2: 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
```

## Page 1

### Field descriptions

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.

**Network Region** - The only administrable field on this page is the **Network Region** field.

Valid entries	Usage
1 to 250	Enter the network region in which the PE interface of the LSP or ESS server resides.

2. Click **Next** to view the next page.

The system displays page 2 of the **Survivable Processor** screen ([Figure 3: Survivable Processor - Processor Channels screen](#) on page 70).

**Figure 3: Survivable Processor - Processor Channels screen**

SURVIVABLE PROCESSOR - PROCESSOR CHANNELS									
Proc Chan	Enable	Appl.	Mode	Interface Link/Chan	Destination Node	Port	Session Local/Remote		
1	i	mis	s	9 5001	CMS_hogan	0	1 1		

## Page 2

### Field descriptions

Use this page to administer processor channels.

**Appl** - This display-only field identifies the server application type or adjunct connection used on this channel. Currently **mis** is the only application displayed.

**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 media 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 <b>0</b> 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
<b>i(nherit)</b>	Enter <b>i(nherit)</b> to indicate that this link is to be inherited by the LSP or ESS server. When you enter <b>i(nherit)</b> , 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 <b>i(nherit)</b> option in the following situations: <ul style="list-style-type: none"> <li>• The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity</li> <li>• 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</li> </ul>
<b>n(o)</b>	This processor channel is disabled on the LSP or the ESS server. The survivable processor does not use this channel. This is the default.
<b>o(verwrite)</b>	The survivable processor overwrites the processor channel information sent from the main server with the information from this screen. The <b>o(verwrite)</b> option causes the <b>Interface Link</b> field to change to display-only with a value of <b>p</b> . 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**. The value **5001** is recommended for CMS, and **5003** is recommended for DCS.

## New and changed screens

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
1 to 64	For link type x.25.
0, 5000 to 64500	For ethernet or ppp. The channel number 0 means any port can be used.

**Interface Link** - This display-only field identifies the physical link carrying this processor (virtual) channel. 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
<b>c(lient)</b>	Indicate whether the IP session is passive <b>c(lient)</b> or active <b>s(erver)</b> .
<b>s(erver)</b>	

**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 media server must equal the Remote Session number on the remote media 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.

3. Click **Next** to view the next page.

The system displays page 3 of the **Survivable Processor** screen ([Figure 4: Survivable Processor - IP-Services screen](#) on page 73).

Figure 4: Survivable Processor - IP-Services screen

page 3 of x					
SURVIVABLE PROCESSOR - IP-SERVICES					
Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port
CDR1	i	gert_clan6	0	cdr_1	9003
CDR2	i	gert_clan1	0	cdr_rsp	9000

## Page 3

### Field descriptions

Use this page of the **Survivable Processor** screen to administer IP services.

**Enabled** - Use this field to specify how data for each specified service type is transferred to the survivable processor.

Valid entries	Usage
<b>i(nherit)</b>	Enter <b>i(nherit)</b> to indicate that this link is to be inherited by the LSP or ESS server. When you enter <b>i(nherit)</b> , 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 <b>i(nherit)</b> option in the following situations: <ul style="list-style-type: none"> <li>• The main server connects to the adjuncts using a CLAN and you want the ESS server to use the same connectivity</li> <li>• 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</li> </ul>
<b>n(o)</b>	This IP services link is disabled on the LSP or the ESS server. This is the default.
<b>o(verwrite)</b>	The survivable processor overwrites the service type information sent from the main server with the information from this screen. Entering <b>o(verwrite)</b> causes the <b>Local Node</b> field to change to <b>procr</b> . 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.

## New and changed screens

**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**.

4. Click **Next** to view the next page.

The system displays page 4 of the **Survivable Processor** screen ([Figure 5: Survivable Processor - IP-Services - Session Layer Timers screen](#) on page 74).

**Note:**

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 5: Survivable Processor - IP-Services - Session Layer Timers screen**

SURVIVABLE PROCESSOR - IP-SERVICES - Session Layer Timers						
Service Type	Reliable Protocol	Packet Resp Timer	Session Connect Message Cntr	SPDU Cntr	Connectivity Time	
CDR1	n	30	3	3	60	
CDR2	y	30	3	3	60	

page 4 of x

## Page 4

### Field descriptions

**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 <b>y</b> to indicate that you want to Use reliable protocol if the adjunct on the far end of the link supports it. Default is <b>y</b> .

**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.

## New and changed screens

**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.

---

## Release 3.0 new screens

Avaya Communication Manager, release 3.0, includes the following new screens. For a more complete explanation of the screens and their function, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

---

### Announcement Group Board Usage

A new screen, the **Announcement Group Board Usage** screen, displays all the audio groups in a system. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **Announcement Group Board Usage** screen:

1. Type `list usage integ-annc-board n`, where `n` is the 5-character circuit pack location number. Press **Enter**.

The system displays the **Announcement Group Board Usage** screen ([Figure 6: Announcement Group Board Usage screen](#)).

---

**Figure 6: Announcement Group Board Usage screen**

list usage integ-annc-board 01B18					Page 1
ANNOUNCEMENT GROUP BOARD USAGE					
USED BY					
Announcement	Announcement	2087	Extension	3005	
Audio Group	Group	4	Member	28	
Audio Group	Group	23	Member	101	

---

## Audio Group

The **Audio Group** screen allows an administrator to add, change, or delete audio source locations in an audio group. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **Audio Group** screen:

1. Type `add audio-group n`, where *n* is the group number. Press **Enter**.

The system displays the **Audio Group** screen ([Figure 7: Audio Group screen](#) on page 77).

---

**Figure 7: Audio Group screen**

add audio-group 1				Page 1 of 4
AUDIO GROUP 1				
Group Name: Relaxation Music__				
AUDIO SOURCE LOCATION				
1: __01A0905	16: _____	31: _____	46: _____	61: _____
2: __ 019v9	17: _____	32: _____	47: _____	62: _____
3: _____	18: _____	33: _____	48: _____	63: _____
4: _____	19: _____	34: _____	49: _____	64: _____
5: _____	20: _____	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: _____

---

### Field descriptions

**Group Name** - The **Group Name** field is an identifier name for the audio group.

**Audio Source Location** - The **Audio Source Location** field contains the list of VAL boards or V VAL location designators for each audio source in the audio group.

---

## Audio Groups

The **Audio Groups** screen is a display-only screen that lists the identifying group number, name, and the number of sources of all audio groups in a system. This change is prompted by the [Locally sourced announcements and music](#) feature.

## New and changed screens

To view the **Audio Groups** screen:

1. Type `list audio-group`. Press **Enter**.

The system displays the **Audio Groups** screen ([Figure 8: Audio Groups screen](#) on page 78).

---

**Figure 8: Audio Groups screen**

```
list audio-group
```

AUDIO GROUPS		
Group	Name	Number of Sources
1	Relaxation Music	2
12	Forgive the wait	20
23	Best deals for today	30
At End of Member List		

---

### Field descriptions

**Group** - The **Group** field displays the assigned number for the group.

**Name** - The **Name** field displays the assigned name for the group.

**Number of Sources** - The **Number of Sources** field displays the number of assigned audio or music sources for the group.

---

## IP Interfaces

The **IP Interfaces** screen, when you use the `list` command, is a display-only screen. The screen lists all of the TN2302AP Media Processor and TN2602AP Media Resource 320 circuit packs in a system. The screen also indicates whether a circuit pack is duplicated. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

**Note:**

For release 3.0, duplication is not yet available.

To view the **IP Interfaces** screen:

1. Type `list ip-interface medpro`.

The system displays the **IP Interfaces** screen ([Figure 9: IP Interfaces screen](#) on page 79).

Figure 9: IP Interfaces screen

```
list ip-interface medpro
```

IP INTERFACES									
ON Slot	Code	Sfx	IP Address	Subnet Mask	Gateway Address	Net		Rgn	Dup
----	----	----	-----	-----	-----	----	----	----	----
y 01A04	TN2302		172.22.22.179	255.255.0.0	172.22.22.254	1	0		n
y 01B10	TN2302		172.22.22.182	255.255.0.0	172.22.22.254	1	0		n
y 01A13	TN2302		172.22.22.185	255.255.0.0	172.22.22.254	1	0		n
y 64B09	TN2302		172.23.23.32	255.255.0.0	172.23.23.254	3	0		n
y 64B10	TN2302		172.23.23.33	255.255.0.0	172.23.23.254	3	0		n
y 01B09	TN2302		172.22.22.233	255.255.0.0	172.22.22.254	1	0		n
y 01A11	TN2602		172.22.22.234	255.255.0.0	172.22.22.254	1	n		n
y 26A05	TN2302		172.22.22.249	255.255.0.0	172.22.22.254	2	0		n
y 01B11	TN2302		172.22.22.191	255.255.0.0	172.22.22.254	1	0		n
y 26B02	TN2302		172.22.22.192	255.255.0.0	172.22.22.254	2	0		n
y 26B03	TN2302		172.22.22.193	255.255.0.0	172.22.22.254	2	0		n
y 26A02	TN2302		172.22.22.194	255.255.0.0	172.22.22.254	2	0		n
y 26A06	TN2302		172.22.22.195	255.255.0.0	172.22.22.254	2	0		n
y 64A05	TN2302		172.23.23.30	255.255.0.0	172.23.23.254	3	0		n
y 64A10	TN2302		172.23.23.31	255.255.0.0	172.23.23.254	3	0		n

## MOH Group

Use the new **MOH Groups** screen to administer music on hold groups. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **MOH Groups** screen:

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

The system displays the **MOH Groups** screen ([Figure 10: MOH Group screen](#) on page 80).

**Figure 10: MOH Group screen**

Page 1 of 4

add moh-analog-group 1

MOH GROUP 1

Group Name: Relaxation Music\_\_

MOH SOURCE LOCATION

1: __01A0905	16: _____	31: _____	46: _____	61: _____
2: __019v401	17: _____	32: _____	47: _____	62: _____
3: _____	18: _____	33: _____	48: _____	63: _____
4: _____	19: _____	34: _____	49: _____	64: _____
5: _____	20: _____	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: _____

**Field descriptions**

**Group Name** - The **Group Name** field is an identifier name for the music-on-hold group.

Valid entries	Usage
1 to 27 alpha-numeric characters	Type a free form identifier of the music-on-hold group.

**MOH Source Location** - The **MOH Source Location** field contains the list of VAL boards or V VAL location designators for each audio source in the music-on-hold group.

Valid entries	Usage
1 to 5 alpha-numeric characters	Type a valid VAL board or V VAL location designator of an audio source.

---

## Music-on-Hold Groups

A new screen, the **Music-On-Hold Groups** screen, displays all the music-on-hold groups in a system. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **Music-On-Hold Groups** screen:

1. Type `list moh-analog-group`. Press **Enter**.

The system displays the **Music-On-Hold Groups** screen ([Figure 11: Music-on-Hold Groups screen](#) on page 81).

---

**Figure 11: Music-on-Hold Groups screen**

list moh-analog-group			Page 1
MUSIC-ON-HOLD GROUPS			
Group	Name	Number of Sources	
8	Elevator Music	5	
10	Latin Renditions	20	
11	Polka music	30	
At End of Member List			

---

## System Parameters Media Gateway Automatic Recovery Rule

A new screen, the **System Parameters Media Gateway Automatic Recovery Rule** screen, defines the rules for automatically 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 media server. This change is prompted by the [Auto fallback to primary for H.248 media gateways](#) feature.

To view the **System Parameters Media Gateway Automatic Recovery Rule** screen:

1. Type `change system-parameters mg-recovery-rule n.`, where *n* is the recovery rule number. Press **Enter**.

The system displays the **System Parameters Media Gateway Automatic Recovery Rule** screen.

**Note:**

The system displays a different time screen and warning message depending on the option that you select for the **Migrate H.248 MG to primary** field. The following four figures show the screen that appears for each option in the **Migrate H.248 MG to primary** field:

- **immediately** ([Figure 12: System Parameters Media Gateway Automatic Recovery Rule screen](#) on page 82).
- **0-active-calls** ([Figure 13: System Parameters Media Gateway Automatic Recovery Rule screen](#) on page 83).
- **time-day-window** ([Figure 14: System Parameters Media Gateway Automatic Recovery Rule screen](#) on page 83).
- **time-window-OR-0-active-calls** ([Figure 15: System Parameters Media Gateway Automatic Recovery Rule screen](#) on page 84).

---

**Figure 12: System Parameters Media Gateway Automatic Recovery Rule screen**

```
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 13: System Parameters Media Gateway Automatic Recovery Rule screen**

```
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 14: System Parameters Media Gateway Automatic Recovery Rule screen**

```
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 15: System Parameters Media Gateway Automatic Recovery Rule screen**

```

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 the **Migrate H.248 MG to primary** field to indicate the auto-fallback preferences. For each option, the system displays a unique warning message and screen.

Valid entries	Usage
<b>immediately</b>	This is the default value. The first media gateway registration that comes from the media gateway is honored, regardless of call count or time of day.
<b>0-active calls</b>	The first media gateway registration reporting “0 active calls” is honored.

Valid entries	Usage
<b>time-day-window</b>	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.
<b>time-window-OR-0-active-calls</b>	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. <b>Note:</b> You must specify an <b>x</b> or <b>X</b> 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.

### 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
<b>3 to 15</b>	Enter the number of minutes before auto-fallback is attempted. The default value is <b>3</b> .

### Recovery Rule Number

Valid entries	Usage
<b>1 to the server maximum</b>	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.

---

## TTI Service IP Stations

A new screen, **TTI Service IP Stations**, lists IP telephones that are in TTI service. The **Registered IP Stations** screen does not list telephones that are in TTI service. An IP telephone that is in TTI service also does not appear in the **Unregistered IP Telephones** report. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

To view the **TTI Service IP Stations** screen:

1. Type `list tti-ip-stations`. Press **Enter**.

The system displays the **TTI Service IP Stations** screen ([Figure 16: TTI Service IP Stations screen](#) on page 86).

---

**Figure 16: TTI Service IP Stations screen**

```
list tti-ip-stations Page 1
```

TTI SERVICE IP STATIONS					
Station	Product	Prod	Net	Gatekeeper	
IP Address	ID	Rel	Rgn	IP Address	Port
135.9.159.130	IP_Phone	2.200	1	135.9.159.128	S00074

---

All fields are display only. The system sorts the information on the screen by the **Station IP Address** field.

---

## Virtual MAC Addresses

The **Virtual MAC Addresses** screen is a read-only display of virtual Media Access Control (MAC) addresses that are shared by duplicated TN2602AP Media Resource 320 circuit packs. The duplicated TN2602AP Media Resource 320 circuit packs must be in an S8500 or S8700-series Media Server system. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

To view the **Virtual MAC Addresses** screen:

1. Type `display virtual-mac-address n`, where `n` is the MAC address table number.

The system displays the **Virtual MAC Addresses** screen ([Figure 17: Virtual MAC Addresses screen](#) on page 87).

Figure 17: Virtual MAC Addresses screen

```

display virtual-mac-address 1
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
Obtained from list	Virtual MAC address shared by duplicated TN2602AP circuit packs.

### Used

The system populates the **Used** field, and indicates whether a virtual MAC address has been assigned in the system.

Valid entries	Usage
y/n	If <b>y</b> , the associated virtual MAC address has been assigned in the system.

## Release 3.1 changed screens

Avaya Communication Manager, release 3.1, includes the following changed screens. For a more complete explanation of the screens and their function, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

## Class of Restriction

The **Class of Restriction** screens changed. You can now assign up to 995 CORs. This change is prompted by the [Increased Classes of Restriction](#) feature.

Beginning with page #3 of the **Class of Restriction** screens, you can assign Classes of Restriction 0-99. Additional pages are added to assign CORs 100-199, 200-299, and so on.

Beginning with page #13, you can assign Service Observing permissions on the CORs. The following screens show the changes.

To view the **Class of Restriction** screen:

1. Type `change cor n`, where *n* is the Class of Restriction that you want to change. The value *n* can be a number between **0** and **995**. Press **Enter**.

The system displays the **Class of Restriction** screen.

2. Click **Next** until you see the **Calling Permission** section ([Figure 18: Class of Restriction screen](#) on page 88).

**Figure 18: Class of Restriction screen**

change cor 5						Page 3 of 22
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? y	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	78? 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					

3. To assign Service Observing permissions to a COR, click **Next** until you see the **Service Observing Permission** section ([Figure 19: Class of Restriction screen](#) on page 89).

Figure 19: Class of Restriction screen

```

change cor 5
                                                    Page 13 of 22
                                CLASS OF RESTRICTION

SERVICE OBSERVING PERMISSIONS
(Enter "y" to grant permission to service observe the 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? y    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    78? 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

```

## Class of Service

The Class of Service screen did not change with Communication Manager release 3.1. However, the [Enterprise Mobility User](#) feature prompted a change to the **Personal Station Access (PSA)** field.

**Personal Station Access (PSA)** - the **Personal Station Access (PSA)** field allows users to associate a telephone to their extension with their programmed services using a feature access code (FAC). 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.

---

## Console Parameters

A new field, **Busy Indicator for Call Parked on Analog Station Without Hardware?**, is added to the **Console Parameters** screen.

To view the **Console Parameters** screen:

1. Type `change console-parameters`. Press **Enter**.

The system displays the **Console Parameters** screen.

2. Click **Next** until you see the **Busy Indicator for Call Parked on Analog Station Without Hardware?** field ([Figure 20: Console Parameters screen](#) on page 90).

---

**Figure 20: Console Parameters screen**

```
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?
```

---

### Field descriptions

#### Busy Indicator for Call Parked on Analog Station Without Hardware? -

Valid entries	Usage
y/n	Enter <b>y</b> to indicate that the Busy Indicator lamp will light for incoming calls parked on AWOH stations. Default is <b>n</b> .

---

## CTI Link

### Change 1

The field description for the **CTI Link** field is changed on the **CTI Link** screen. A new field, **Two-Digit Aux Work Reason Codes?**, is also added to the **CTI Link** screen. These changes are prompted by the [ASAI support for Aux Work reason codes](#) feature.

#### Note:

The **CTI Link** screen is available only if, on the **Optional Features** screen, either the **ASAI Link Core Capabilities** and/or the **Computer Telephony Adjunct Links** field is **y**.

To view the **CTI Link** screen:

1. Type `add cti-link next`. Press **Enter**.

The system displays the **CTI Link** screen ([Figure 22: CTI Link screen](#) on page 92).

---

**Figure 21: CTI Link screen**

```

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

```

---

### Field descriptions

**CTI Link** - This field is display-only, and indicates the CTI link number.

Valid entries	Usage
1 to system max	Avaya Communication Manager on a DEFINITY Server CSI, DEFINITY G3i, S8300 Media Server, S8700 Series Multi-Connect

## New and changed screens

2. Click **Next** until you see the **Two-Digit Aux Work Reason Codes?** field ([Figure 22: CTI Link screen](#) on page 92).

**Figure 22: CTI Link screen**

add cti-link next	CTI LINK	Page 2 of x
FEATURE OPTIONS		
Event Minimization?	Special Character for Restricted Number?	
	Send Disconnect Event for Bridged Appearance?	
	Two-Digit Aux Work Reason Codes?	
	Block CMS Move Agent Events?	

### Field descriptions

#### Two-Digit Aux Work Reason Codes? -

Valid entries	Usage
y/n	Enter <b>y</b> 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 <b>y</b> when <b>Two-Digit Aux Work Reason Codes?</b> on the <b>System-Parameters Features</b> screen is set to <b>y</b> . Default is <b>n</b> .

#### Change 2

A new field, **Block CMS Move Agent Events?**, is added to the **CTI Link** screen. This change is prompted by the [Block CMS Move Agent events](#) feature.

### Field descriptions

#### Block CMS Move Agent Events? -

Valid entries	Usage
y/n	When this option is set to <b>y</b> , 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 <b>n</b> .

## Enterprise Survivable Server Information

Beginning with Communication Manager release 3.1, node names are used in place of IP addresses for ESS servers on pages 1 through 5 of the **Enterprise Survivable Server Information** screen. Other field values are increased.

To view the **Enterprise Survivable Server Information** screen:

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

The system displays the **Enterprise Survivable Server Information** screen ([Figure 23: Enterprise Survivable Server Information screen](#) on page 93).

**Figure 23: Enterprise Survivable Server Information 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
-----
-----
-----
-----
-----
-----
-----
-----
-----
-----

```

### Field descriptions

**Server A ID** - The valid entries are increased to **256**.

Valid entries	Usage
1 to 256 or blank	Enter the Server ID (SVID) of the S8400 or S8500 Media Server, or the SVID of the A-side of the S8700 series media server. If this field is blank, all other entries on the line display default values.

## New and changed screens

**Server A Node Name** - This field is changed from **IP Address** to **Node Name**.

Valid entries	Usage
Valid node name administered on the <b>IP Node Names</b> screen.	Enter the node name for the S8400 or S8500 Media Server, or the A-side of the S8700-series Media Server.

**Server B ID** - The valid entries are increased to **256**.

Valid entries	Usage
<b>1 to 256</b> or blank	S8700-series Media Servers only. Enter the Server ID (SVID) of the B-side of the S8700-series Media Server.

**Server B Node Name** - This field is changed from **IP Address** to **Node Name**.

Valid entries	Usage
Valid node name administered on the <b>IP Node Names</b> screen.	S8700-series Media Servers only. Enter the node name for the B-side S8700-series Media Server.

---

## Feature Access Code (FAC)

**Note:**

There are numerous changes to the **Feature Access Code (FAC)** screen for Communication Manager release 3.1. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **Feature Access Code (FAC)** screen:

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

The system displays the **Feature Access Code (FAC)** screen.

2. Click **Next** until you see the **Enterprise Mobility User Activation** field ([Figure 24: Feature Access Code \(FAC\) screen](#) on page 95).

Figure 24: Feature Access Code (FAC) screen

```

change feature-access-codes                                     Page 2 of x
                                FEATURE ACCESS CODE (FAC)
                                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
                                Leave Word Calling Send A Message: *60
                                Leave Word Calling Cancel A Message: #60
                                Malicious Call Trace Activation:           Deactivation:
                                Meet-me Conference Access Code Change:

```

## Page 2

### Change 1

Two new fields, **Enterprise Mobility User Activation** and **Enterprise Mobility User Deactivation**, are added to the **Feature Access Code (FAC)** screen. These changes are prompted by the [Enterprise Mobility User](#) feature.

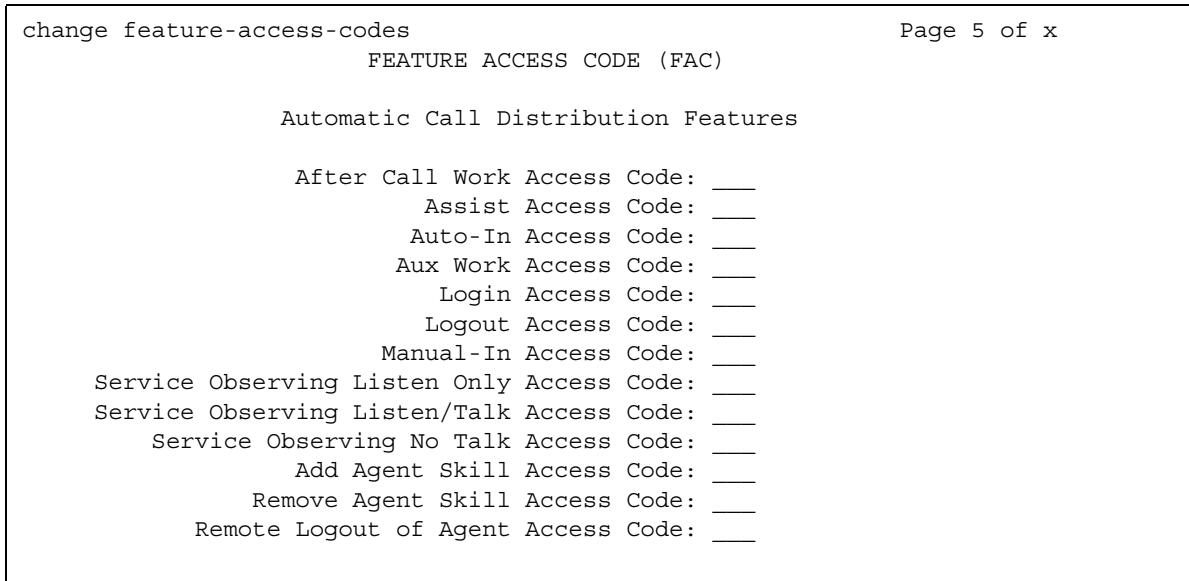
### Field descriptions

**Enterprise Mobility User Activation** - Type a feature access code number to allow users to activate the Enterprise Mobility User feature, associating the features and permissions of their primary telephone to a telephone of the same type anywhere within the customer's enterprise.

**Enterprise Mobility User Deactivation** - Type a feature access code number to allow users to deactivate the Enterprise Mobility User feature.

3. Click **Next** until you see the **Service Observing No-Talk Access Code** field ([Figure 24: Feature Access Code \(FAC\) screen](#) on page 95).

**Figure 25: Feature Access Code (FAC) screen**



## Page 5

### Change 2

A new field, **Service Observing No-Talk Access Code**, is added to the **Feature Access Code (FAC)** screen. This change is prompted by the [Listen-only FAC for service observing](#) feature.

#### Field descriptions

**Service Observing No-Talk Access Code** - This field appears only if **Expert Agent Selection (EAS) Enabled** is set to **y** 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.

---

## Feature-Related System Parameters

### Note:

There are numerous changes to the **Feature-Related System Parameters** screen for Communication Manager release 3.1. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **Feature-Related System Parameters** screen:

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 Processing Overload Mitigation** section ([Figure 26: Feature-Related System Parameters screen](#) on page 97).

**Figure 26: Feature-Related System Parameters screen**

```

change system-parameters features                                     Page 3 of 17

                                FEATURE-RELATED SYSTEM PARAMETERS

TTI/PSA PARAMETERS

WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

Terminal Translation Initialization (TTI) Enabled?  ___
  TTI State: _____      TTI Security Code: _____
Record CTA/PSA/TTI Transactions in History Log?    ___
Enhanced PSA Location/Display Information Enabled?  ___
  Default COR for Dissociated Sets: _____
  CPN, ANI for Dissociated Sets: _____
Customer Telephone Activation (CTA) Enabled?       ___

CALL PROCESSING OVERLOAD MITIGATION
Restrict Calls: _____

```

## Page 3

### Change 1

The **Call Processing Overload Mitigation** section, **Restrict Calls** field, is moved from the **Maintenance-Related System Parameters** screen to the **Feature-Related System Parameters** screen.

Since changes to this field control how the system reacts to overload, and since the system currently denies all intercom and outband calls on overload, changes to this field only improves the way in which the system reacts to overload conditions.

When the **Restrict Calls** field was part of the **Maintenance-Related System Parameters** screen, a user could not change the field unless the user had maintenance system permissions (MSP). Without MSP, the user had to call services to have them change the field. By moving the **Restrict Calls** field to the **Feature-Related System Parameters** screen, users with super-user permissions can now change this field without having to call services for help.

## New and changed screens

### Field descriptions

**Restrict Calls** - This field indicates 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.

3. Click **Next** until you see the **AUDIX One Step Recording** section ([Figure 27: Feature-Related System Parameters screen](#) on page 98).

**Figure 27: 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:

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):
```

## Page 7

### Change 2

Two new fields, **Apply Ready Indication Tone to Which Parties in the Call?** and **Interval for Applying Periodic Alerting Tone (seconds)**, are added to the **Feature-Related System Parameters** screen. These changes are prompted by enhancements to Avaya Modular Messaging.

#### Field descriptions

**Apply Ready Indication Tone to Which Parties in the Call?** - This field is for administering who hears the AUDIX recording ready tone.

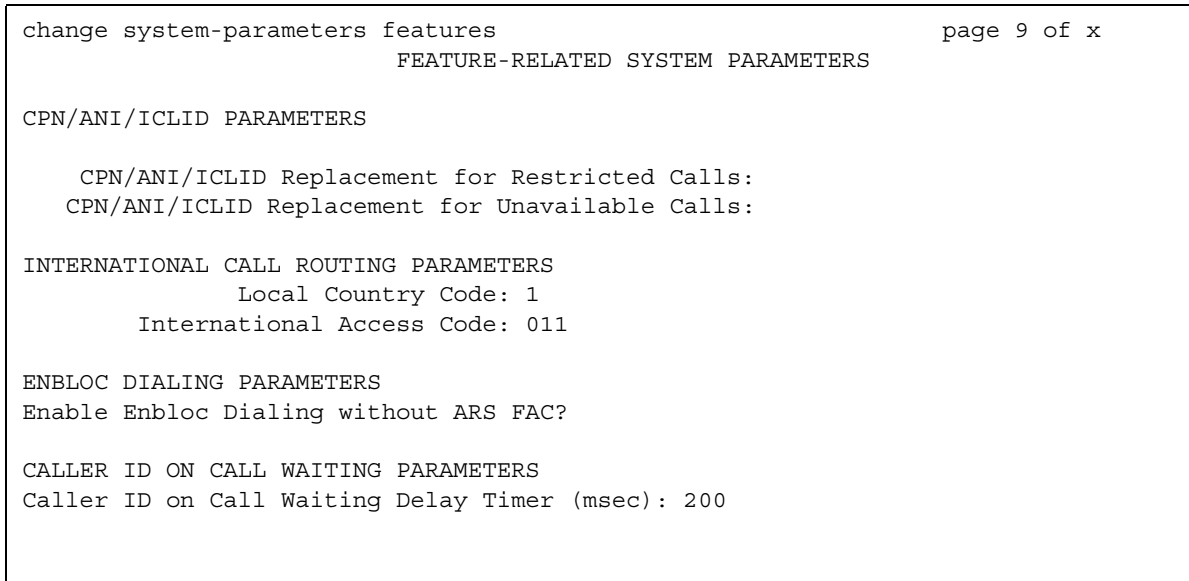
Valid entries	Usage
<b>all, initiator, or none</b>	Enter a value for which party or parties on the call should hear the ready-to-record indication tone. The default is <b>all</b> . This field cannot be left blank.

**Interval for Applying Periodic Alerting Tone (seconds)** - This field appears only if the **Apply Ready Indication Tone To Which Parties In The Call?** field is set to **all**.

Valid entries	Usage
<b>0 to 60</b>	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.

- Click **Next** until you see the **Caller ID on Call Waiting Parameters** section ([Figure 28: Feature-Related System Parameters screen](#) on page 100).

**Figure 28: Feature-Related System Parameters screen**



## Page 9

### Change 3

A new section heading, **Caller ID on Call Waiting Parameters**, and a new field, **Caller ID on Call Waiting Delay Timer (msec)**, are added to the **Feature-Related System Parameters** screen. These changes are prompted by [Support caller ID on call waiting for MM711 and MM714](#).

### Field descriptions

#### 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.

5. Click **Next** until you see the **Vectoring** section ([Figure 29: Feature-Related System Parameters screen](#) on page 101).

Figure 29: 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

VECTORIZING
    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? _
    Selection on BSR Ties?

SERVICE OBSERVING
    Service Observing: Warning Tone? n      or Conference Tone?
    Service Observing Allowed with Exclusion? n

```

## Page 11

### Change 4

A new field, **Selection on BSR Ties?**, is added to the **Feature-Related System Parameters** screen.

### Field descriptions

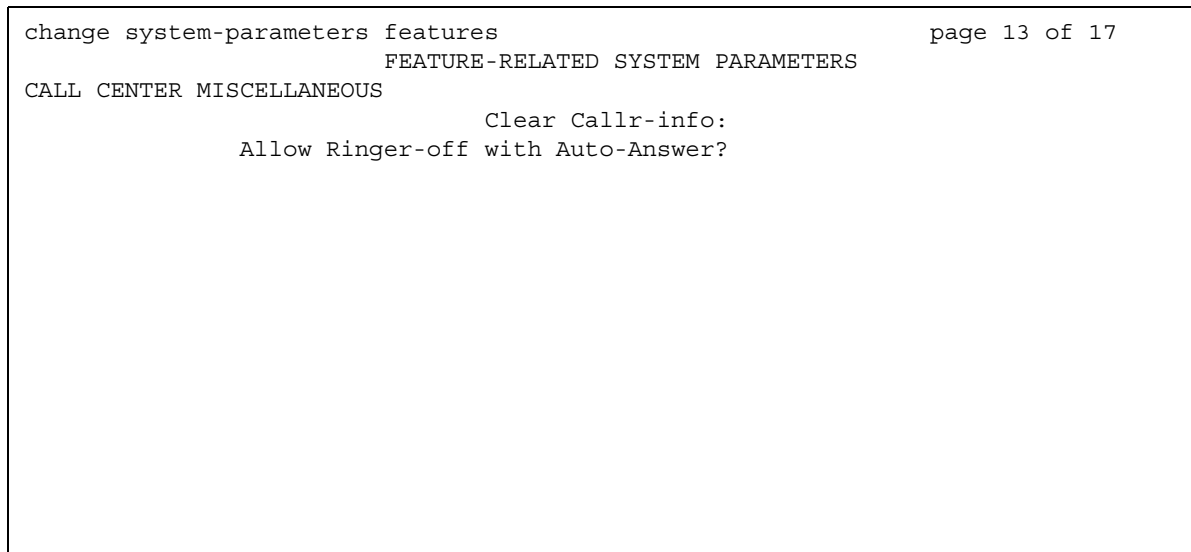
**Selection on BSR Ties?** - This field appears only when the **Vectoring (Best Service Routing)** field on the **Optional Features** screen is **y**.

Valid entries	Usage
<b>1st-found</b>	BSR uses the first selection for routing. This is the default.
<b>alternate</b>	Allows alternating 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 active VDN, the selection from the consider step with the tie is used instead of the first selected split/skill or location to send the call. This helps balance the routing over the considered local splits/skills and remote locations when cost of routing remotely is not a concern.

## New and changed screens

- Click **Next** until you see the **Call Center Miscellaneous** section ([Figure 30: Feature-Related System Parameters screen](#) on page 102).

**Figure 30: Feature-Related System Parameters screen**



## Page 13

### Change 5

A new page and two new fields are added to the **Feature-Related System Parameters** screen.

### Field descriptions

**Clear Callr-info** - Use this field to specify when collected digits from a previous caller is to be removed from the agent/station display.

Valid entries	Usage
<b>leave-ACW</b>	Leaves the display up while the agent is in ACW (After-call work) mode.
<b>next-call</b>	Clears the display when the next call is received. This is the default.
<b>on-call-release</b>	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 presses the release button.

**Allow Ringer-off with Auto-Answer** - Use this field to prevent ringing on EAS auto-answer calls.

Valid entries	Usage
y/n	Enter <b>y</b> to allow a user to use the <b>ringer-off</b> feature button to prevent ringing on EAS auto-answer calls.

7. Click **Next** until you see the **IP Parameters** section ([Figure 31: Feature-Related System Parameters screen](#) on page 103).

**Figure 31: 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):

```

## Page 16

### Change 6

The **IP Audio Hairpinning?** field on the **Feature-Related System Parameters** screen now defaults to **n**.

**Note:**

In addition to the **Feature-Related System Parameters** screen, the **IP Audio Hairpinning?** field now defaults to **n** on the following additional screens:

- **Attendant Console**
- **IP Network Region**
- **Signaling Group**
- **Station**

---

## Group Paging Using Speakerphone

A new field, **TN**, is added to the **Group Paging Using Speakerphone** screen. This new field allows the system administrator to assign group paging at the tenant partition level, instead of to an entire campus.

To view the **Group Paging Using Speakerphone** screen:

1. Type `change group-paging n`, where **n** is the number of the paging group. Press **Enter**.

The system displays the **Group Paging Using Speakerphone** screen ([Figure 32: Group Paging Using Speakerphone screen](#) on page 105).

**Figure 32: Group Paging Using Speakerphone screen**

```

change group-paging 1                                     Page 1

                                GROUP PAGING USING SPEAKERPHONE

    Group Number: 1                                     Group Extension:
    Group Name:                                         COR: 1
GROUP MEMBER ASSIGNMENTS                               TN: 1
  Ext      Name (first 26 characters)      Ext      Name (first 26 characters)
  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:

```

**Field descriptions**

**TN** - The **TN** field identifies the number of a tenant partition.

**Hunt Group**

A new field, **Provide Ringback?**, is added to the **Hunt Group** screen.

To view the **Hunt Group** screen:

1. Type `change hunt-group n`, where *n* is the number of the hunt group. Press **Enter**.  
The system displays the **Hunt Group** screen.
2. Click Next until you see the **Provide Ringback?** field ([Figure 33: Hunt Group screen](#) on page 106).

**Figure 33: Hunt Group screen**

```

change hunt-group 3                                     Page 2 of X
                                     HUNT GROUP

                                     Message Center: _____
                                     Voice Mail Number: _____
Routing Digits (e.g. AAR/ARS Access Code): _____
                                     Send Reroute Request? y           Provide Ringback? n
                                     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? _
    
```

**Field descriptions**

**Provide Ringback?** - This field 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 <b>y</b> , 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 <b>n</b> .

**IP Interfaces**

**Change 1**

Six new fields are added to the **IP Interfaces** screen, prompted by the [Processor Ethernet](#) feature. The six new fields are:

- [Allow H.323 Endpoints?](#)
- [Allow H.248 Endpoints?](#)
- [Gatekeeper Priority](#)
- [Target socket load](#)

- [Target socket load and Warning level](#)
- [Link](#)

To view the **IP Interfaces** screen:

1. Type `change ip-interface procr`.

The system displays the **IP Interfaces** screen for Processor Ethernet ([Figure 34: IP Interfaces screen](#) on page 107).

**Figure 34: IP Interfaces screen**

```

change 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:
    
```

**Field descriptions**

**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 <b>y</b> to allow H.323 endpoint connectivity to the Processor Ethernet (PE) interface. Enter <b>n</b> if you do not want H.323 endpoint connectivity to the PE interface. <b>Note:</b> For an Enterprise Survivable Server (ESS), this field is display -only and is set to <b>n</b> . 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 <b>y</b> .

## New and changed screens

**Allow H.248 Endpoints?** - 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 <b>y</b> to allow H.248 endpoint connectivity to the PE interface. Enter <b>n</b> if you do not want H.248 endpoint connectivity to the PE interface. <b>Note:</b> For an Enterprise Survivable Server (ESS), this field is display-only and is set to <b>n</b> . 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 <b>y</b> .

**Gatekeeper Priority** - This field appears only if the **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. The default is <b>5</b> .

**Target socket load** - This field appears when the **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.

Valid entries	Usage
1 to platform maximum as follows: <ul style="list-style-type: none"><li>● S8700 series: <b>3500</b></li><li>● S8500: <b>3500</b></li><li>● S8400: <b>2500</b></li><li>● CHAWK/BOXTER: <b>2000</b></li><li>● VM/BLADE: <b>1700</b></li></ul>	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.

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 <b>400</b> .

**Link** - This display-only field shows the administered link number for an Ethernet link.

Valid entries	Usage
y/n	This display-only field shows the unique number for the Ethernet link assigned on the <b>Data Module</b> screen.

## Change 2

The **IP Interfaces** screen, when you use the `list` command, is a display-only screen. The screen lists all of the TN2302AP Media Processor and TN2602AP Media Resource 320 circuit packs in a system.

The **IP Interfaces** screen also indicates whether a TN2602 circuit pack is duplicated. Duplicated TN2602 circuit packs are listed together, that is, one after the other.

For more information on the **IP Interfaces** screen using the `list` command, see [IP Interfaces](#) on page 146.

---

## IP Network Region

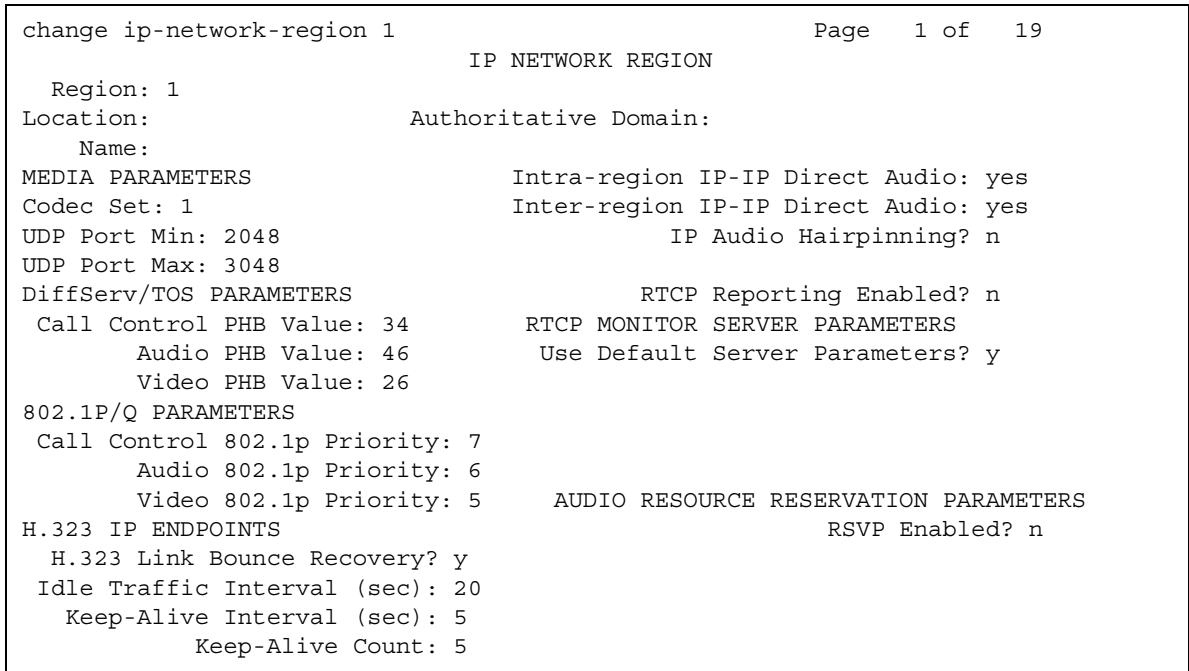
The **IP Audio Hairpinning?** field on the **IP Network Region** screen now defaults to **n**.

To view the **IP Network Region** screen:

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

The system displays the **IP Network Region** screen ([Figure 35: IP Network Region screen](#) on page 110).

**Figure 35: IP Network Region screen**

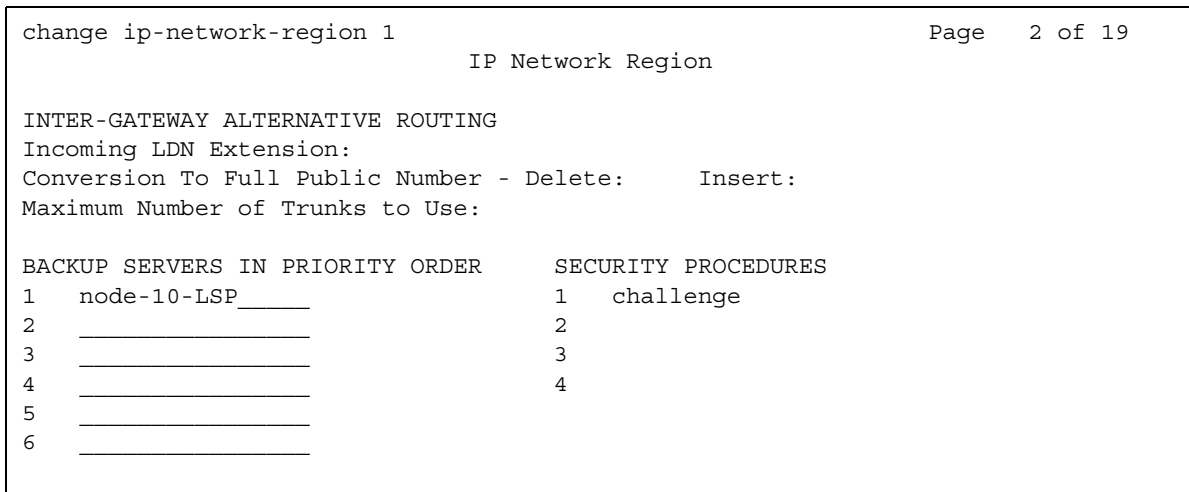


On page 2 of the **IP Network Region** screen, the **LSP Servers in Priority Order** section was renamed to **Backup Servers in Priority Order**.

2. Click **Next** to see page 2.

The system displays the **IP Network Region** screen ([Figure 36: IP Network Region screen](#) on page 110).

**Figure 36: IP Network Region screen**



## IP Server Interface (IPSI) Administration

Use the **IP Server Interface (IPSI) Administration** screen to add a TN2312 IPSI (IP Server Interface) circuit pack. The S8700 Series Media 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.

This change was prompted by the [S8400 Media Server](#).

To view the **IP Server Interface (IPSI) Administration** screen:

1. Type `change ipserver-interface n`, where *n* is the number of the port network that you want to change. Press **Enter**.

The system displays the **IP Server Interface (IPSI) Administration** screen ([Figure 37: IP Server Interface \(IPSI\) Administration screen](#) on page 111).

**Figure 37: 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                VAL on IPSI? y                Call Control 802.1p: 4
      Host: ipsi-A01a
      DHCP ID: ipsi-A01a                Call Control DiffServ: 42

Secondary IPSI
-----
Location: 1B01                VAL on IPSI? y
      Host: ipsi-A01b
      DHCP ID: ipsi-A01b

```

### Field descriptions

**VAL on IPSI?** - This field appears only for TN8400/S8400 systems.

Valid entries	Usage
y/n	Indicates whether the primary TN2312BP/TN8412AP circuit pack has VAL functionality running on it.

---

## Language Translations

A new field, **Audix Record**, was added to the **Button Labels** section of the **Language Translations** screen. This change is prompted by enhancements to Avaya Modular Messaging.

To view the **Language Translations** screen:

1. Type `change display-messages button-labels`. Press **Enter**.

The system displays the **Station** screen.

2. Click **Next** until you see the **Audix Record** field ([Figure 38: Language Translations screen](#) on page 112).

---

**Figure 38: Language Translations screen**

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.	*****

---

---

## Media-Processor Status

The **Media-Processor Status** screen now shows duplication of a TN2602 circuit pack. This change is prompted by [Bearer signal duplication](#) on page 23.

To view the **Media-Processor Status** screen:

1. Type `status media-processor all`. Press **Enter**.

The system displays the **Media-Processor Status** screen ([Figure 39: Media-Processor Status screen](#) on page 113).

**Note:**

For information on the `status media-processor all` command, see [status media-processor](#) on page 180.

---

**Figure 39: Media-Processor Status screen**

```

status media-processor all                                     Page 1 of x
                                     MEDIA-PROCESSOR STATUS

      Alarms  Links
Slot  Code   Mj Mn Wn Pr Cl El Dup St   Slot  Code   Mj Mn Wn Pr Cl El Dup St
02A04 TN2602  0 14  0 |dn|up|up|n/a|act
03A04 TN2302  0  0  0 |na|na|up|n/a|n/a
04A05 TN2602  0 12  0 |dn|up|up|n/a|act

Pr=Peer Link, Cl=Control Link, El-Ethernet Link, Dup-Duplicate Slot, St=State

```

---

## Optional Features

Two new fields, **Enterprise Survivable Server** and **ESS Administration**, are added to the **Optional Features** screen. These changes are prompted by the [Enterprise Survivable Server increase](#) feature.

To view the **Optional Features** screen:

1. Type `display system-parameters customer-options`. Press **Enter**.

The system displays the **Optional Features** screen.

2. Click **Next** until you see the **Enterprise Survivable Server** field ([Figure 40: Optional Features screen](#) on page 114).

---

**Figure 40: 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                                             Internet Protocol (IP) PNC? y
  Enhanced Conferencing? y                                           ISDN Feature Plus? y
    Enhanced EC500? y                                               ISDN 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
  Hospitality (G3V3 Enhancements)? y
    IP Trunks? y

IP Attendant Consoles? y

(NOTE: You must logoff & login to effect the permission changes.)
```

---

### Field descriptions

**Enterprise Survivable Server** - This license file-activated field indicates that this server is an Enterprise Survivable Server (ESS).

**ESS Administration** - This license file-activated field allows administration of an Enterprise Survivable Server (ESS) on the **Enterprise Survivable Server Information** screen.

## Route Pattern

A new field, **Secure SIP?**, is added to the **Route Pattern** screen. This change is prompted by CSS Unique Certificates for SIP.

To view the **Route Pattern** screen:

1. Type `change route-pattern n`, where *n* is the pattern number. Press **Enter**.

The system displays the **Route Pattern** screen ([Figure 41: Route Pattern screen](#) on page 115).

**Figure 41: Route Pattern screen**

```

change route-pattern 1                                     Page 1 of X
                                     Pattern Number: 1_   Pattern Name:
                                     SCCAN? n             Secure SIP? n

      Grp. FRL NPA Pfx Hop Toll Del   Inserted          DCS/
      No.   Mrk Lmt List Dgts  Digits          QSIG   IXC
      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 3 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
  
```

### Field descriptions

#### 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 trunk preference. If SIP trunks are not specified on the <b>Route Pattern</b> screen, the call will be routed over whatever trunk is specified. Therefore, to ensure a SIP TLS connection when such a route pattern is invoked, only SIP trunks should be specified. The only instance for entering <b>y</b> in this field is when the source provider <i>requires</i> a secure SIP protocol. Default is <b>n</b> .

## Security-Related System Parameters

A new section, **Remote Managed Services**, and three new fields, **RMS Feature Enabled?**, **Port Board Security Notification**, and **Port Board Security Notification Interval**, are added to the **Security-Related System Parameters** screen.

To view the **Security-Related System Parameters** screen:

1. Type `change system-parameters security`. Press **Enter**.  
The system displays the **Security-Related System Parameters** screen.
2. Click **Next** until you see the **Remote Managed Services** section ([Figure 42: Security-Related System Parameters screen](#) on page 116).

**Figure 42: Security-Related System Parameters screen**

```

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
    
```

### Field descriptions

**RMS Feature Enabled** - This field enables the Remote Managed Services (RMS) feature.

Valid entries	Usage
y/n	Enter <b>y</b> to enable the Remote Managed Services feature. Default is <b>n</b> .

**Port Board Security Notification** - This field appears only if the **RMS Feature Enabled** field is set to **y**.

Valid entries	Usage
y/n	Enter <b>y</b> to enable port board denial of service notification. Default is <b>n</b> .

**Port Board Security Notification Interval** - This field appears only if the **Port Board Security Notification** field is set to **y**.

Valid entries	Usage
<b>60 to 3600</b> , in increments of 10	Enter the desired interval, in seconds, before port board denial of service notification is sent. Default is <b>60</b> .

---

## Station

### Note:

There are numerous changes to the **Station** screen for Communication Manager release 3.1. The changes are all listed here, by page. The information also mentions what feature prompted the change.

## Page 2

### Change 1

Two fields are renamed, and new values assigned.

- The **Call Waiting Indication?** field was changed to **Call Waiting Indication:**. A value of **c** was added to the accepted values for this field.
- The **Att. Call Waiting Indication?** field was changed to **Att. Call Waiting Indication:**. A value of **c** was added to the accepted values for this field.

These changes are prompted by [Support caller ID on call waiting for MM711 and MM714](#).

To view the **Station** screen:

1. Type **change station n**, where **n** is the telephone extension. Press **Enter**.  
The system displays the **Station** screen.
2. Click **Next** until you see the **EMU Login Allowed?** field ([Figure 43: Station screen](#) on page 118).

**Figure 43: Station screen**

```

change station n                                     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:
    Recall Rotary Digit? n -                        Select Last Used Appearance? n
                                                    Coverage After Forwarding? _
                                                    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
    
```

**Field descriptions**

**Call Waiting Indication -**

Valid entries	Usage
<b>y</b>	Enter <b>y</b> to activate Call Waiting (without Caller ID information) for the telephone. This is the Default.
<b>n</b>	Call Waiting is not enabled for the station.
<b>c</b>	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 <b>Type</b> field is <b>CallrID</b> .

**Att. Call Waiting Indication -**

Valid entries	Usage
<b>y</b>	Enter <b>y</b> to activate Call Waiting (without Caller ID information) for the telephone. This is the Default.
<b>n</b>	Call Waiting is not enabled for the station.
<b>c</b>	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 <b>Type</b> field is <b>CallrID</b> .

**Change 2**

A new field, **EMU Login Allowed?**, is added to the **Station** screen. This field allows the station to be used as a visited station by an Enterprise Mobility User (EMU) visitor user. This change is prompted by the [Enterprise Mobility User](#) feature.

**EMU Login Allowed -**

Valid entries	Usage
<b>y/n</b>	Enter <b>y</b> to allow the station to be used as a visited station by an Enterprise Mobility User (EMU) visitor user. Default is <b>n</b> .

---

## Stations with Off-PBX Telephone Integration

A new screen, **Stations with Off-PBX Telephone Integration**, is added as part of the Extension to Cellular feature. New commands are also added (see [Release 3.1 new commands](#) on page 171).

Use the **Stations with Off-PBX Telephone Integration** screen to map an office telephone to a cell phone through the Extension to Cellular feature. The office telephone 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.

To view the **Stations with Off-PBX Telephone Integration** screen:

1. Type `add off-pbx-telephone station-mapping`. Press **Enter**.

The system displays the **Stations with Off-PBX Telephone Integration** screen ([Figure 44: Stations with Off-PBX Telephone Integration screen](#) on page 120).

**Figure 44: Stations with Off-PBX Telephone Integration screen**

add off-pbx-telephone station-mapping <span style="float: right;">Page 1 of 2</span>					
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set
43001	EC500	_____	- 9736831204	ars	1
43001	OPS	_____	- 12345	ars	5
43009	OPS	_____	- 67890	aar	2
43011	CSP	_____	- 6095343211	ars	3
43013	SCCAN	_____	- 9738765432	ars	4/1
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___
_____	_____	_____	- _____	_____	___

**Field descriptions**

**Station Extension** - The **Station Extension** field is an administered extension in your dial plan. This number is the extension of the office telephone.

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** - The **Application** field indicates the type of off-PBX application that is associated with the office telephone. You can assign more than one application to an office telephone.

Valid entries	Usage
blank	Default is blank.
<b>EC500</b>	cell phone with Extension to Cellular
<b>OPS</b>	SIP-enabled telephone
<b>CSP</b>	cell phone with Extension to Cellular provided by the cellular service provider
<b>SCCAN</b>	wireless SIP telephone and cell phone

**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 telephone number is long-distance. Enter "011" if the phone number is international. Default is blank.

**Phone Number** - Enter the telephone number of the off-PBX telephone.

Valid entries	Usage
0-9	Type up to fifteen digits. Enter the complete 10-digit number. Default is blank.

**Trunk Selection** - This field defines what trunk group you will use for outgoing calls.

Valid entries	Usage
ars aar trunk group number	

**Configuration Set** - Use the **Configuration Set** field to administer the Configuration Set number. This number contains the desired call treatment options for the Replace variable w/ short feature name 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(s). Default is blank.

- Click **Next** to view page 2 of the **Stations with Off-PBX Telephone Integration** screen ([Figure 45: Stations with Off-PBX Telephone Integration screen, page 2](#) on page 122). Finish the administration steps to map an office telephone to an off-PBX telephone. The information that you entered on page 1 appears as read-only information on page 2.

**Figure 45: Stations with Off-PBX Telephone Integration screen, page 2**

add off-pbx-telephone station-mapping					Page 2 of 2
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 Replace variable w/ short feature name calls that can be active simultaneously. Default is 2.

**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 Replace variable w/ short feature name 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
<b>both</b>	Default = <b>both</b> when the <b>Phone Number</b> field was previously administered for another extension with a <b>Mapping Mode</b> of <b>termination</b> or <b>none</b> . Default = <b>termination</b> when the <b>Phone Number</b> field was previously administered with a <b>Mapping Mode</b> of <b>origination</b> or <b>both</b> . In the both mode, users can originate and receive calls from the office phone with the cell phone.

Valid entries	Usage
<b>termination</b>	In termination mode, users can only use their Replace variable w/ short feature name 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.
<b>origination</b>	In origination mode, users can only originate Replace variable w/ short feature name cell phone calls from the associated office phone. Users cannot use the cell phone to receive calls from the associated office phone.
<b>none</b>	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 Replace variable w/ short feature name station. The **Calls Allowed** values filter the type of calls to the office phone that a user can receive on an Replace variable w/ short feature name cell phone.

Valid entries	Usage
<b>all</b>	Default is <b>all</b> . The cell phone receives both internal and external calls.
<b>internal</b>	The cell phone receives only internal calls.
<b>external</b>	The cell phone receives only external calls.
<b>none</b>	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 Replace variable w/ short feature name cell phone. The valid entry definitions are the same as the **Mapping Mode** field entries.

Valid entries	Usage
<b>both termination origination none</b>	Default is <b>both</b> .

---

## Trunk Group

**Note:**

There are numerous changes to the **Trunk Group** screen for Communication Manager release 3.1. The changes are all listed here, by page. The information also mentions what feature prompted the change. Some changes are specific to the value that is in the **Group Type** field on page 1 of the **Trunk Group** screen.

**Change 1**

A new field, **Apply Local Ringback?**, is added to the **Trunk Group** screen. This new field allows the system administrator to set the system to provide ringback tone to a caller. This change is prompted by the [Local ringback administration](#) feature.

**Note:**

For this field to appear, the **Carrier Medium** field on page 1 of the **Trunk Group** screen must be set to **PRI\_BRI**.

To view the **Trunk Group** screen:

1. Type `change trunk-group n`, where *n* is the number of the trunk group. Press **Enter**.  
The system displays the **Trunk Group** screen
2. Click **Next** until you see the **Apply Local Ringback?** field ([Figure 46: Trunk Group screen](#) on page 124).

---

**Figure 46: Trunk Group screen**

```
change trunk-group 1                                     Page 3 of x
                                     TRUNK FEATURES
      ACA Assignment? n                               Measured: none____
                                                    Maintenance Tests? y
      Data Restriction? n
      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? y
```

**Field descriptions**

**Apply Local Ringback?** - This field appears for ISDN and H.323 trunk groups when the **Carrier Medium** field on page 1 is **PRI\_BRI**.

Valid entries	Usage
y/n	Enter <b>y</b> to provide a local ringback tone to the caller. The local ringback is removed when the call is connected. Default is <b>n</b> .

**Change 2**

A new field, **Send EMU Visitor CPN?**, is added to the **Trunk Group** screen. This change is prompted by the [Enterprise Mobility User](#) feature.

3. Click **Next** until you see the **Send EMU Visitor CPN?** field ([Figure 47: Trunk Group screen](#) on page 125).

**Figure 47: Trunk Group screen**

```

change trunk-group 1                                     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? n

Suppress # Outpulsing? _      Numbering Format: _____
Outgoing Channel ID Encoding: _____      UII IE Treatment: _____
                                     Maximum Size of UII IE Contents: _____
                                     Replace Restricted Numbers? _
                                     Replace Unavailable Numbers? _
                                     Send Connected Number: _
                                     Hold/Unhold Notifications? _

      Send UII IE? _
      Send UCID? _      BRS Reply-best DISC Cause Value: __
                                     Dsl 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 _____
    
```

## New and changed screens

**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**.

Valid entries	Usage
<b>y</b>	Sends calling party identification information on the extension of the EMU user's telephone.
<b>n</b>	Sends calling party identification information on the primary telephone.

### Change 3 (for ISDN trunks)

A value on the **Carrier Medium** field is changed, and three new fields, **Member Assignment Method**, **Signaling Group**, and **Number of Members**, are added to the **Trunk Group** screen. These changes are prompted by the [Increased trunk members for IP signaling groups](#) feature.

To view the **Trunk Group** screen for ISDN trunks:

1. Type `change trunk-group n`, where *n* is the trunk group number. Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 48: Trunk Group screen](#) on page 126). Make sure that the **Group Type** field is set to **isdn**.

**Figure 48: Trunk Group screen**

```
change trunk-group 1                                     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: H.323
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:
```

## Field descriptions

**Carrier Medium.** The value **IP** was replaced with **H.323**.

**Member Assignment Method** - This field appears when the **Carrier Medium** field is set to **H.323**.

Valid entries	Usage
<b>manual</b>	Default. Users manually assign trunk members to a signaling group.
<b>auto</b>	The system automatically generates members to a specific signaling group. Entering <b>Auto</b> causes the <b>Signaling Group</b> and <b>Number of Members</b> fields to appear.

**Signaling Group** - This field appears when **Carrier Medium** is **H.323** and **Member Assignment Method** is **auto**.

Valid entries	Usage
<b>1 to 650</b> , or blank	Enter assigned H.323 or SIP signaling group number between 1 and 650, or blank.

**Number of Members** - This field appears when **Carrier Medium** 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.

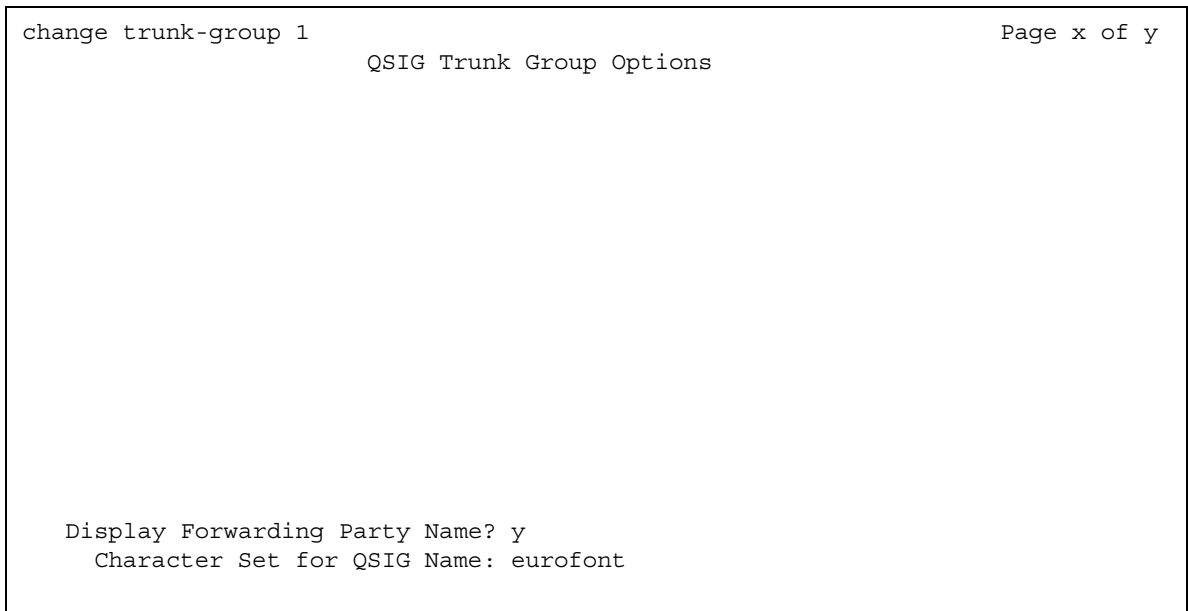
Valid entries	Usage
<b>0 to 255</b>	Enter the number of trunks assigned to this signaling group. Default is <b>0</b> .

### Change 4 (for ISDN trunks)

A new page, **QSIG Trunk Group Options**, and two new fields, **Display Forwarding Party Name** and **Character Set for QSIG Name**, are added to the **Trunk Group** screen. The fields on this screen appear only when the **Group Type** field is **isdn**, and the **Supplementary Service Protocol** field is **b**.

2. Click **Next** until you see the **QSIG Trunk Group Options** screen ([Figure 49: QSIG Trunk Group Options screen](#) on page 128).

**Figure 49: QSIG Trunk Group Options screen**



**Field descriptions**

**Display Forwarding Party Name** - This field appears only when the **Group Type** field on page 1 is **isdn**, and the **Supplementary Service Protocol** field on page 2 is **b**.

Valid entries	Usage
y/n	Enter <b>y</b> if you want the system to display the name of the party who is forwarding the call. Default is <b>y</b> .

**Character Set for QSIG Name** - This field appears only when the **Group Type** field on page 1 is **isdn**, the **Supplementary Service Protocol** field on page 2 is **b**, and the **Display Character Set** field on the **System Parameters Country-Options** screen is **Roman**.

Valid entries	Usage
eurofont	This option sets the Roman Eurofont character set. This option is the default.
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.  <b>Note:</b> 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.

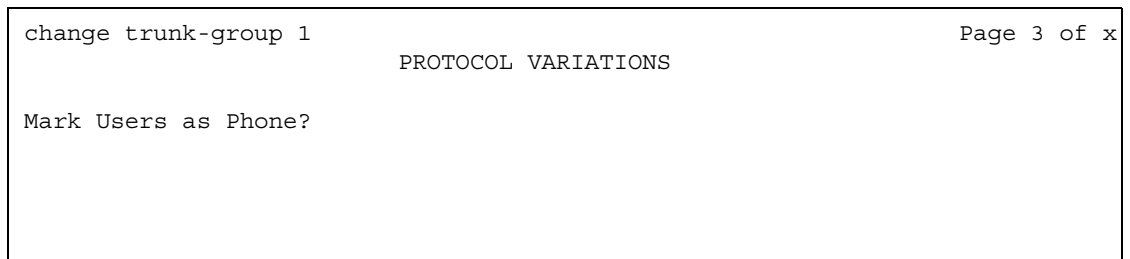
**Change 5 (for SIP trunks)**

A new page, **Protocol Variations**, and one new field, **Mark Users as Phone?**, is added to the **Trunk Group** screen. This screen appears only when the **Group Type** field is **sip**.

To view the **Trunk Group** screen for SIP trunks:

1. Type `change trunk-group n`, where **n** is the trunk group number. Press **Enter**.  
The system displays the **Trunk Group** screen. Make sure that the **Group Type** field on page 1 is set to **sip**.
2. Click **Next** until you see the **Protocol Variations** page ([Figure 50: Trunk Group screen](#) on page 129).

**Figure 50: Trunk Group screen**

**Field descriptions****Mark Users as Phone? -**

Valid entries	Usage
y/n	<p>When the field is set to <b>y</b>, 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 <b>y</b>. Default is <b>n</b>.</p> <p><b>Note:</b> Do not change the default of 'no' (<b>n</b>) for this field unless you are sure that every recipient of SIP calls via this trunk can accept and properly interpret the optional “user=phone” parameter. Enterprise users without support for “user=phone” in their SIP endpoints will experience adverse effects, including rejected calls.</p>

**Change 6 (for SIP trunks)**

A new field, **Preferred Minimum Session Refresh Interval (sec)**, is added to the **Trunk Group** screen. This screen appears only when the **Group Type** field is **sip**.

## New and changed screens

To view the **Trunk Group** screen for SIP trunks:

1. Type **change trunk-group n**, where **n** is the trunk group number. Press **Enter**.

The system displays the **Trunk Group** screen. Make sure that the **Group Type** field on page 1 is set to **sip**.

2. Click **Next** until you see the **Trunk Parameters** section ([Figure 51: Trunk Group screen](#) on page 130).

**Figure 51: Trunk Group screen**

```
change trunk-group 1                                     Page 2 of x
  Group Type: sip

Trunk Parameters:

  Unicode Name? y

                                                    Redirect on OPTIM Failure: 5000

  SCCAN? n                                             Digital Loss Group: 18
  Preferred Minimum Session Refresh Interval (sec): 120
```

### Field descriptions

**Preferred Minimum Session Refresh Interval (sec)** - This field appears when the **Group Type** field on page 1 is **sip**, and the **SCCAN** field on page 2 is **n**. This field sets the session refresh timer value of a SIP session for non-SCCAN applications. The timer starts once a SIP 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 120.

---

## Variables for Vectors

A new screen, **Variables for Vectors**, is added for Call Center-related enhancements for Communication Manager release 3.1.

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.

To view the **Variables for Vectors** screen:

1. Type `change variables`. Press **Enter**.

The system displays the **Variables for Vectors** screen ([Figure 52: Variables for Vectors screen](#) on page 131).

---

**Figure 52: Variables for Vectors screen**

change variables		VARIABLES FOR VECTORS						Page 1 of x
Var	Description	Type	Scope	Length	Start	Assignment	VAC	
A:	_____	_____	___	_____	_____			
B:	_____	_____	___	_____	_____			
C:	_____	_____	___	_____	_____			
D:	_____	_____	___	_____	_____			
E:	_____	_____	___	_____	_____			
F:	_____	_____	___	_____	_____			
G:	_____	_____	___	_____	_____			
H:	_____	_____	___	_____	_____			
I:	_____	_____	___	_____	_____			
J:	_____	_____	___	_____	_____			
K:	_____	_____	___	_____	_____			
L:	_____	_____	___	_____	_____			
M:	_____	_____	___	_____	_____			

2. Click **Next** to view page 2 of the **Variables for Vectors** screen ([Figure 53: Variables for Vectors screen](#) on page 132).

**Figure 53: Variables for Vectors screen**

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:	_____	_____	___	_____	___		

**Var -**

Valid entries	Usage
A to Z	Display only. The letter identifying the row of a specific vector variable.

**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.

**Type -**

Valid entries	Usage
<b>ani</b> <b>asaiuui</b> <b>collect</b> <b>dow</b> <b>doy</b> <b>stepcnt</b> <b>tod</b> <b>value</b> <b>vdn</b> <b>vdntime</b>	Enter the vector variable type.

**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
<b>G/L</b>	Indicate whether the variable is to be used locally ( <b>L</b> ) or globally ( <b>G</b> ).

**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
<b>1 to 16</b>	Enter the maximum length of digits to use in the variable.

**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
<b>1 to 96</b>	Enter the character position of the first digit to be stored in the variable.

## New and changed screens

**Assignment** - This field only allows entry when the **Type** field is set to **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
<b>digits</b>	Enter a number to pre-assign to the variable. This field displays the current value for global values

**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 FAC.

Valid entries	Usage
<b>1 to 9</b> or blank	Displays the Vector Variable Feature Access Code (FAC) to use for changing the value.

---

## Vector Directory Number

A new field, **Selection on BSR Ties**, is added to the **Vector Directory Number** screen.

To view the **Vector Directory Number** screen:

1. Type **change vdn n**, where **n** is the extension of the vector directory number (vdn). Press **Enter**.

The system displays the **Vector Directory Number** screen.

2. Click **Next** until you see the **Selection on BSR Ties** field ([Figure 54: Vector Directory Number screen](#) on page 135).

Figure 54: Vector Directory Number screen

```

change vdn 4567
VECTOR DIRECTORY NUMBER
AUDIX Name:
Return Destination:
VDN Timed ACW Interval:
BSR Application:
BSR Available Agent Strategy:
Selection on BSR Ties:
Observe on Agent Answer?
Display VDN for Route-To DAC?
VDN Override for ISDN Trunk ASAI Messages?
BSR Local Treatment?
Page 2 of x

```

### Field descriptions

**Selection on BSR Ties** - This field appears only when the **Vectoring (Best Service Routing)** field on the **Optional Features** screen is **y**.

Valid entries	Usage
<b>system</b>	The setting of the <b>Selection on BSR Ties?</b> field on the <b>Feature-Related System Parameters</b> screen applies. This is the default. <b>Note:</b> For more information, see <a href="#">Page 11</a> of the <b>Feature-Related System Parameters</b> screen.
<b>1st-found</b>	BSR uses the first selection for routing. That is, BSR uses the current best selected from the previous consider commands.
<b>alternate</b>	Allows alternating 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 active VDN, the selection from the consider step with the tie is used instead of the first selected split/skill or location to send the call. This helps balance the routing over the considered local splits/skills and remote locations when cost of routing remotely is not a concern.

## Release 3.0 changed screens

Avaya Communication Manager, release 3.0, includes the following changed screens. For a more complete explanation of the screens and their function, see the *Administrator Guide for Avaya Communication Manager*, 03-300509.

### Announcements/Audio Sources

The **Group/Port** field is modified in the **Announcements/Audio Sources** screen. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **Announcements/Audio Sources** screen:

1. Type `change announcements`. Press **Enter**.

The system displays the **Announcements/Audio Sources** screen ([Figure 55: Announcements/Audio Sources screen](#) on page 136).

**Figure 55: Announcements/Audio Sources screen**

change announcements										Page 1
ANNOUNCEMENTS/AUDIO SOURCES										
	Ext.	Type	COR	TN	Name	Q	QLen	Pro	Rate	Group/ Port
1:	55526__	integrated	1_	1_	business-hours	y	N/A	n	64	01B18
2:	55525__	integ-mus	1_	1_	Music4Queue__	b	N/A	n	64	G1
3:	33627__	integrated	1_	1_	PresWelcome__	y	N/A	n	64	G2
4:	_____	_____	1_	1_	_____	n				
5:	_____	_____	1_	1_	_____	n				

#### Field descriptions

**Group/Port** - The **Group/Port** field identifies the group number or the port location of an audio source.

Valid entries	Usage
<b>Gnn</b> port location	<b>Gnn</b> , where <b>nn</b> represents a one or two-digit group number. The location of the TN2501AP or the TN750 announcement circuit pack. In this example, the location is 01B18.
<b>ggvn</b>	<b>ggv9</b> for media gateway V VAL, where <b>gg</b> is the number of the media gateway, and <b>n</b> is the slot number.

---

## Configuration Set

The **Configuration Set** screen introduces the **Barge-in Tone** field. This change is prompted by the [Extension to Cellular](#) feature.

To view the **Configuration Set** screen:

1. Type `change off-pbx-telephone configuration-set n`, where *n* is the number that is assigned with a coverage path command. Press **Enter**

The system displays the **Configuration Set** screen ([Figure 56: Configuration Set screen](#) on page 137).

---

**Figure 56: Configuration Set screen**

```
change off-PBX-telephone configuration-set 1

                                CONFIGURATION SET: 1

Configuration Set Description: Standard
Calling Number Style: network
CDR for Origination: phone-number
CDR for Calls to EC500 Destination? y
Fast Connect on Origination? n
Post Connect Dialing Options: dtmf
Cellular Voice Mail Detection: none
Barge-in Tone? n
Identity When Bridging? station
```

---

### Field descriptions

**Barge-in Tone** - The **Barge-in Tone** field indicates whether the barge-in tone is enabled or disabled.

Valid entries	Usage
y/n	Type <b>y</b> to enable the barge-in tone. Type <b>n</b> to disable the barge-in tone.

---

## Extensions To Call Which Activate Features By Name

Three new fields are added to the **Extensions To Call which Activate Features By Name** screen. The new fields are **Automatic Call Back**, **Extended Group Call Pickup**, and **Whisper Page Activation**. This change is prompted by the [Extension to Cellular](#) feature.

To view the **Extensions To Call which Activate Features By Name** screen:

1. Type `change off-pbx-telephone feature-name-extensions`. Press **Enter**.

The system displays the **Extensions To Call which Activate Features By Name** screen ([Figure 57: Extensions To Call Which Activate Features By Name screen](#) on page 138).

---

**Figure 57: Extensions To Call Which Activate Features By Name screen**

```
change off-pbx-telephone feature-name-extensions                               Page 1 of 1
EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
Active Appearance Select: 31001                               Idle Appearance Select: 31020
Automatic Call-Back: _____                               Last Number Dialed: _____
Automatic Call-Back Cancel: _____                         Malicious Call Trace: _____
Call Forward All: 31002                                       Malicious Call Trace Cancel: _____
Call Forward Busy/No Answer: 31003                             Off-PBX Call Enable: _____
Call Forward Cancel: 31004                                    Off-PBX Call Disable: _____
Call Park: 31005                                             Priority Call: _____
Call Park Answer Back: _____                               Send All Calls: _____
Call Pick-Up: _____                                       Send All Calls Cancel: _____
Conference on Answer: _____                               Transfer On Hang-Up: _____
Calling Number Block: _____                               Transfer to Voice Mail: _____
Calling Number Unblock: _____                             Whisper Page Activation: _____
Directed Call Pick-Up: _____
Drop Last Added Party: _____
Exclusion (Toggle On/Off): _____
Extended Group Call Pick-up: _____
Held Appearance Select: _____
```

---

### Field descriptions

**Automatic Call Back - Automatic Call Back** allows users to choose whether they want an extension to automatically call them back. If a user places a call to a busy or unanswered telephone, the system calls the user back when the called telephone becomes available.

**Extended Group Call Pickup - Extended Group Call Pickup** allows a user to answer calls that were directed to another call pickup group.

**Whisper Page Activation - Whisper Page Activation** allows a user to make whisper pages. A whisper page is a low volume message. Users can send a whisper page when they want only one person on a conference call to hear a message.

---

## Feature Access Code (FAC)

The **Feature Access Code (FAC)** screen introduces two new fields:

- **EC500 Self Administration Access Code**
- **Enhanced EC500 Activation / Deactivation**

This change is prompted by the [Extension to Cellular](#) feature.

To view the **Feature Access Code (FAC)** screen:

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

The system displays the **Feature Access Code (FAC)** screen.

2. Click **Next** until you see the **EC500 Self Administration Access Code** field ([Figure 58: Feature Access Code \(FAC\) screen](#) on page 139).

---

**Figure 58: Feature Access Code (FAC) screen**

```

change feature-access-codes                                     Page 2 of 8
                                FEATURE ACCESS CODE (FAC)
                                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: *89      Deactivation:#89
                                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: *81      Deactivation: #81
                                ISDN Access Code:
                                Last Number Dialed Access Code: *54
                                Leave Word Calling Message Retrieval Lock: *48
                                Leave Word Calling Message Retrieval Unlock: #45
                                Leave Word Calling Send A Message: *60
                                Leave Word Calling Cancel A Message: #60
                                Malicious Call Trace Activation:      Deactivation:
                                Meet-me Conference Access Code Change:

```

---

### Field descriptions

**EC500 Self Administration Access Code** - The **EC500 Self Administration Access Code** field indicates the access code in accordance with an customer dial plan.

**Enhanced EC500 Activation / Deactivation** - The **Enhanced EC500 Activation / Deactivation** fields allow a user to activate and deactivate the enhanced EC500 features.

---

## Feature-Related System Parameters

**Note:**

There are numerous changes to the **Feature-Related System Parameters** screen for Communication Manager release 3.0. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **Feature-Related System Parameters** screen:

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

The system displays the **Feature-Related System Parameters** screen ([Figure 59: Feature-Related System Parameters screen](#) on page 140).

---

**Figure 59: Feature-Related System Parameters screen**

```
change system-parameters features                                     Page 1 of 15

                                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: Group 1_
                                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?
```

---

### Page 1

The **Music/Tone On Hold** field is modified to change the source type for music-on-hold. This change is prompted by the [Locally sourced announcements and music](#) feature.

**Field descriptions**

**Music/Tone On Hold** - The **Music/Tone On Hold** field assigns audio as music on hold or tone on hold.

Valid entries	Usage
<b>none</b> <b>music</b> <b>tone</b>	Type <b>none</b> for no music-on-hold. Type <b>music</b> for music-on-hold. Type <b>tone</b> for tone-on-hold.

If you type **music** or **tone**, the **Type** field appears.

**Type** - Use the **Type** field to assign music to an extension, a group, or a port.

Valid entries	Usage
<b>ext</b>	Type <b>ext</b> and the corresponding extension number of the integ-mus announcement/audio group.
<b>group</b>	Type <b>group</b> and the corresponding music-on-hold analog group number.
<b>port</b>	Type <b>port</b> and the corresponding port location of the music-on-hold analog/aux-trunk source.

- Click **Next** until you see the **TTI/PSA Parameters** area ([Figure 60: Feature-Related System Parameters screen](#) on page 141).

**Figure 60: Feature-Related System Parameters screen**

```

change system-parameters features                                     Page 3 of 15
                        FEATURE-RELATED SYSTEM PARAMETERS

TTI/PSA PARAMETERS

      WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE
      Terminal Translation Initialization (TTI) Enabled? y
      TTI State: voice      TTI Security Code: 123456
      Record CTA/PSA/TTI Transactions in History Log? n
      Enhanced PSA Location/Display Information Enabled? n
      Default COR for Dissociated Sets: 94
      CPN, ANI for Dissociated Sets: 1234567
      Customer Telephone Activation(CTA) Enabled? n

```

## Page 3

- The **COR for PSA Dissociated Sets** field in earlier releases of Communication Manager is renamed **Default COR for Dissociated Sets**.
- The **CPN, ANI for PSA Dissociated Sets** field in earlier releases of Communication Manager is renamed **CPN, ANI for Dissociated Sets**.

This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature. The function of these two fields did not change.

3. Click **Next** until you see the **Enable Inter-Gateway Alternate Routing?** field ([Figure 61: Feature-Related System Parameters screen](#) on page 142).

**Figure 61: Feature-Related System Parameters screen**

```
change system-parameters features Page 5 of 15
                                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

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: _____
```

## Page 5

- The **Emergency Extension Forwarding (min)** field is moved under the **System-Wide Parameters** area.
- A new field, **Enable Inter-Gateway Alternate Routing?**, is added to the **Feature-Related System Parameters** screen. This change is prompted by the [Inter-Gateway Alternate Routing](#) feature.

Valid entries	Usage
y/n	Type <b>y</b> to enable the Inter-Gateway Alternate Routing feature. The default is <b>n</b> .

4. Click **Next** until you see the **Maximum Ports per Expanded Meet-me Conf** field ([Figure 62: Feature-Related System Parameters screen](#) on page 143).

**Figure 62: Feature-Related System Parameters screen**

```

change system-parameters features                                     Page 7 of 15
                                                                    FEATURE-RELATED SYSTEM PARAMETERS
CONFERENCE/TRANSFER
    Abort Transfer?
    Transfer Upon Hang-Up?
    Abort Conference Upon Hang-Up?
    No Hold Conference Timeout:
ANALOG BUSY AUTO CALLBACK
    Without Flash?
AUDIX ONE-STEP RECORDING
    Apply Ready Indication Tone To Which Parties In The Call?
    Interval For Applying Periodic Alerting Tone (seconds):
                                                                    No Dial Tone Conferencing?
                                                                    Select Line Appearance Conferencing?
                                                                    Unhold?
    Maximum Ports per Expanded Meet-me Conf:
                                                                    Announcement:
                                                                    Voice Mail Hunt Group Ext:

```

## Page 7

A new field, **Maximum Ports per Expanded Meet-me Conf**, allows you to administer the maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit. You cannot administer the maximum number of conferees on a per Expanded-Meet-me VDN basis.

## New and changed screens

The **Maximum Ports per Expanded Meet-me Conf** field is hidden if the **Maximum Number of Expanded Meet-me Conference Ports** field on the [Optional Features screen](#) is set to **0**. This change is prompted by the [Expanded Meet-me Conferencing](#) feature.

Valid entries	Usage
<b>0 to 300</b> , in increments of <b>50</b>	Enter the maximum number of parties allowed for each conference on your system.

5. Click **Next** until you see the **International Call Routing Parameters** area ([Figure 63: Feature-Related System Parameters screen](#) on page 144).

**Figure 63: Feature-Related System Parameters screen**

change system-parameters features	Page 9 of 15
FEATURE-RELATED SYSTEM PARAMETERS	
CPN/ANI/ICLID PARAMETERS	
CPN/ANI/ICLID Replacement for Restricted Calls:	
CPN/ANI/ICLID Replacement for Unavailable Calls:	
INTERNATIONAL CALL ROUTING PARAMETERS	
Local Country Code: 1	
International Access Code: 011	
ENBLOC DIALING PARAMETERS	
Enable Enbloc Dialing without ARS FAC?	

## Page 9

The **SBS Parameters** area in earlier releases of Communication Manager is renamed **International Call Routing Parameters**. This change is prompted by the [Inter-Gateway Alternate Routing](#) feature.

---

## Gateway Status

A new value, **pd**, is added to the **Lk** field on the **Gateway Status** screen. This change is prompted by the [Auto fallback to primary for H.248 media gateways](#) feature.

To view the **Gateway Status** screen:

1. Type `status media-gateways`. Press **Enter**.

The system displays the **Gateway Status** screen ([Figure 64: Gateway Status screen](#) on page 145).

---

**Figure 64: Gateway Status screen**

```

status media-gateways                                     Page 1 of x

  ALARM SUMMARY          BUSY-OUT SUMMARY          H.248 LINK SUMMARY
  Major: 3              Trunks: 0              Links Down: 3      # Logins: 1
  Minor: 0              Stations: 0             Links UP: 0
  Warning: 0

                                GATEWAY STATUS

  Alarms                Alarms                Alarms                Alarms                Alarms
MG Mj Mn Wn Lk  MG Mj Mn Wn Lk  MG Mj Mn Wn Lk  MG Mj Mn Wn Lk  MG Mj Mn Wn Lk
1   1 | 0 | 0 |dn
2   1 | 0 | 0 |up
3   1 | 0 | 0 |pd

```

---

### Lk

The value **pd** indicates a pending link status when all three of these conditions are met:

- The media gateway has a recovery rule administered
- At least one registration request has been denied
- The media gateway has still not registered with the primary server

---

## Integrated Announcements/Audio

The **Group Number** field is added to the **Integrated Announcements/Audio** screen. This change is prompted by the [Locally sourced announcements and music](#) feature.

To view the **Integrated Announcements/Audio** screen:

1. Type `list integrated-annc-boards`. Press **Enter**.

The system displays the **Integrated Announcements/Audio** screen ([Figure 65: Integrated Announcements/Audio screen](#) on page 146).

---

**Figure 65: Integrated Announcements/Audio screen**

```
list integrated-annc-boards                                     Page 1
                                                                INTEGRATED ANNOUNCEMENTS/AUDIO
Board Location: 08A05                                         Time Remaining at 64 Kbps: 255
Annc.   Internal  Group  Announcement                                     Length in
Number  Number    Number Extension   Name                                               Rate      Seconds
1:      NA        6      658001      Collect_1_digit                                   64        4
2:      NA        31     658002      No_digits_coll_Goodbye                           64        5
3:      NA        11     658003      Collect_5_digits                                  64        5
4:      NA        10     658004      voal                                              64        6
5:      NA         9      658005      Music4                                           32       60
```

---

### Field descriptions

**Group Number** - The **Group Number** field identifies the audio group number of the audio sources on a specific VAL or V VAL.

---

## IP Interfaces

Numerous fields have either been added or changed on the **IP Interfaces** screen. These changes are prompted by the new [TN2602AP IP Media Resource 320 circuit pack](#).

To view the **IP Interfaces** screen:

1. Type `change ip-interface n`, where *n* is the slot location.

The system displays the **IP Interfaces** screen ([Figure 66: IP Interfaces screen](#) on page 147).

Figure 66: IP Interfaces screen

```

change ip-interface 1a03                                     Page 1 of 1

                                IP INTERFACES

                                Critical Reliable Bearer? y

                                Type: MEDRPRO
                                Slot: 01A03
                                Code/Suffix: TN2602
                                Node Name:
                                IP Address:
                                Subnet Mask: 255.255.255.0
                                Gateway Address:
                                Enable Ethernet Port? y
                                Network Region: 20
                                VLAN: n
                                VOIP Channel: xxx
                                Shared Virtual Assress: 255.255.255.255
                                Virtual MAC Table:

                                ETHERNET OPTIONS

                                Auto? n
                                Speed: 100Mbps
                                Duplex: Full

                                Slot: 01B03
                                Code/Suffix: TN2602
                                Node Name:
                                IP Address:
                                Subnet Mask: 255.255.255.0
                                Gateway Address:
                                Enable Ethernet Port? y
                                Network Region: 20
                                VLAN: n
                                VOIP Channel: xxx
                                Virtual MAC Table:

```

## Field Descriptions

### Critical Reliable Bearer?

The system displays this field when the:

- Board code of the slot location that you entered in the command is **TN2602**
- Software version is **V13** or greater

This field indicates that two TN2602AP circuit packs are **(y)** or are not **(n)** duplicated in a port network.

Valid entries	Usage
<b>n</b> (the default)	Indicates that two TN2602AP circuit packs are not duplicated in a port network.
<b>y</b>	Indicates that two TN2602AP circuit packs are duplicated in a port network. The fields on the right side of the screen appear. The system auto-fills the <b>Subnet Mask</b> and <b>Network Region</b> fields from the values on the left side of the screen.

## New and changed screens

**Slot** - The **Slot** field is display only. The system populates the left side of the screen with the slot location that you entered in the command. If the **Critical Reliable Bearer?** field is **y**, you must enter the slot location of the second TH2602 circuit pack in the **Slot** field on the right of the screen.

**Code/Suffix** - The **Code/Suffix** field is display only. The system populates the left side of the screen with the proper value based on the slot location that you entered in the command. If the **Critical Reliable Bearer?** field is **y**, the system populates the right side of the screen with the same value.

**Node Name** - Type the node name value on the left side of the screen that is associated with the **IP Address** of the TN2602 circuit pack. If the **Critical Reliable Bearer?** field is **y**, type the node name value of the second TN2602 circuit pack on the right side of the screen.

**IP Address** - Type the IP address on the left side of the screen that is associated with the **Node Name** of the TN2602 circuit pack. If the **Critical Reliable Bearer?** field is **y**, type the IP address of the second TN2602 circuit pack on the right side of the screen.

**Subnet Mask** - Type the subnet mask, if any, on the left side of the screen that is associated with the IP address. If the **Critical Reliable Bearer?** field is **y**, the system populates the right side of the screen with the same value.

**Gateway Address** - The **Gateway Address** field is display only. The system populates the left side of the screen with the gateway address that is associated with the IP address. If the **Critical Reliable Bearer?** field is **y**, the system populates the right side of the screen with the same value.

**Enable Ethernet Port?** - This field indicates that two TN2602AP circuit packs are (**y**) or are not (**n**) duplicated in a port network.

Valid entries	Usage
<b>n</b>	Indicates that the ethernet port that is associated with the TN2602AP circuit pack is disabled. <ul style="list-style-type: none"><li>● You cannot disable the ethernet port if the active TN2602AP circuit pack is part of an active/standby pair.</li><li>● You can disable the ethernet port if either:<ul style="list-style-type: none"><li>- There is no standby TN2602AP circuit pack</li><li>- The standby TN2602AP circuit pack has already been disabled</li></ul></li></ul>
<b>y</b> (the default)	Indicates that the ethernet port that is associated with the TN2602AP circuit pack is enabled.

**Network Region** - Type the network region on the left side of the screen that is associated with the TN2602AP circuit pack. If the **Critical Reliable Bearer?** field is **y**, the system populates the right side of the screen with the same value. You can override the value on the right side of the screen.

**VLAN** - This field is the virtual LAN (VLAN) that is associated with the TN2602AP circuit pack. If the **Critical Reliable Bearer?** field is **y**, override the default value, if necessary, on both sides of the screen.

Valid entries	Usage
n (the default)	There is no VLAN ID that is associated with the TN2602AP circuit pack.
<b>0-4094</b>	A four-digit alpha-numeric character, between 0 and 4094, that indicates the VLAN ID and user priority (802.1p/Q).

**VOIP Channels** - The VOIP Channels field identifies the number of VoIP channels that are allocated to the associated TN2602AP circuit pack.

Valid entries	Usage
<b>0</b> (the default) <b>80</b> <b>320</b>	Type the number of VoIP channels - 0, 80, or 320 - that are associated with the TN2602AP circuit pack.

- If two TN2602AP circuit packs are administered (the **Critical Reliable Bearer?** field is **y**) in a load balanced configuration, you can give each VoIP channel setting the value of 320, for a total of 640.
- If two TN2602AP circuit packs are administered (the **Critical Reliable Bearer?** field is **y**) in a critically duplicated configuration, the system populates the right side of the screen with the value that you typed in the left side. The field on the right side of the screen is display only.

**Shared Virtual Address** - The system displays the **Shared Virtual Address** field only when the **Critical Reliable Bearer?** field is **y**. Type the virtual IP address for the duplicated bearer circuit packs.

**Virtual MAC Table and Virtual MAC Address** - The system displays the **Virtual MAC Address** field only when the **Critical Reliable Bearer?** field is **y**. The system populates the **Virtual MAC Address** field based on the value in the **Virtual MAC Table** field.

---

## IP Network Region

**Note:**

There are numerous changes to the **IP Network Region** screen for Communication Manager release 3.0. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **IP Network Region** screen:

1. Type **change ip-network-region n**, where **n** is the network region that you want to change. Press **Enter**.

The system displays the **IP Network Region** screen.

2. Click **Next** until you see the **Inter-Gateway Alternate Routing** area ([Figure 67: IP Network Region screen](#) on page 150).

---

**Figure 67: IP Network Region screen**

```
change ip-network-region 1 Page 2 of 19

                                IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING
Incoming LDN extension:_____
Conversion to Full Public Number - Delete:_ Insert:_____
Maximum Number of Trunks to Use:___

LSP NAMES IN PRIORITY ORDER          SECURITY PROCEDURES
1  _____                        1. strong___
2  _____                        2. challenge
3  _____                        3. _____
4  _____                        4. _____
5  _____
6  _____
```

---

### Page 2

A new field area, **Inter-Gateway Alternate Routing**, is added to the **IP Network Region** screen. This change is prompted by the [Inter-Gateway Alternate Routing](#) feature.

#### Incoming LDN Extension

Valid entries	Usage
Valid unused extension	Assign an unused Listed Directory Number for incoming IGAR calls.

### Conversion to Full Public Number - Insert

Valid entries	Usage
up to 13 digits	Enter up to 13 digits to insert. International numbers should begin with a plus sign (+), or blank. The optional plus sign (+) at the beginning of the inserted digits is an international convention indicating that you must dial the local international access code before the number.

### Maximum Number of Trunks to Use

It is necessary to impose a limit on the trunk usage in a particular PN/MG 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 alternate routing.

#### Note:

The S8500 supports up to 800 IP trunks (via license file limitations), which is less than the S8700 limit, but the overall maximum number of trunk members is the same as on the S8700: 8000.

A new field area, **Security Procedures**, is added to the **IP Network Region** screen. This change is prompted by the [Security of IP telephone registration/H.323 signaling channel](#) feature.

The permitted values for the **Security Procedures** area fields are:

**strong** - To permit use of any strong security profile. Only the **pin-eke** profile fits this category.

**pin-eke** - The mechanism used by this feature.

**challenge** - Includes the various methods of PIN-based challenge/response schemes in current use. These schemes support mechanisms prior to Communication Manager release 3.0.

**any-auth** - Includes any of the above.

By default, each network region has **challenge** enabled for a new installation, or when upgrading from a release prior to the implementation of this requirement.

## New and changed screens

IP telephones register based upon capability and administration:

- If you select either **strong** or **pin-cke**, IP telephones that do not support this feature cannot register or make calls.
  - If you select **challenge**, all IP telephones use the registration mechanism prior to Communication Manager release 3.0.
  - If you select **any-auth**, all telephones use whatever registration mechanism they support to register.
3. Click **Next** until you see the **Inter Network Region Connection Management** area ([Figure 68: IP Network Region screen](#) on page 152).

**Figure 68: IP Network Region screen**

```
change ip-network-region 1 Page 3 of 19
```

Inter Network Region Connection Management

src rgn	dst rgn	codec set	direct WAN	WAN-BW limits	Intervening-regions	Dynamic CAC 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						

## Page 3

A new column, **IGAR**, is added to the **IP Network Region** screen. This change is prompted by the [Inter-Gateway Alternate Routing](#) feature.

## IGAR

This field allows pair-wise configuration of Inter-Gateway Alternate Routing (IGAR) 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.

Valid entries	Usage
<b>y</b>	Enter <b>y</b> to enable IGAR capability between this network region pair.
<b>n</b>	IGAR capability between this network region pair is disabled. The default is <b>n</b> , except when <b>codec set</b> is <b>pstn</b> . When <b>codec set</b> is <b>pstn</b> , this field defaults to <b>y</b> .
<b>f</b>	Forced. This option can be used during initial installation to verify the alternative PSTN facility selected for a network region pair. This option may 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.

---

## IP-Options System Parameters

A new field, **Periodic Registration Timer (min)**, is added to the **IP-Options System Parameters** screen.

To view the **IP-Options System Parameters** screen:

1. Type `display system-parameters ip-options`. Press **Enter**.

The system displays the **IP-Options System Parameters** screen ([Figure 69: IP-Options System Parameters screen](#) on page 154).

**Figure 69: IP-Options System Parameters screen**

```
display system-parameters ip-options Page 1 of 2
      IP-OPTIONS SYSTEM PARAMETERS

IP MEDIA PACKET PERFORMANCE THRESHOLDS
  Roundtrip Propagation Delay (ms)      High: 120      Low: 75
      Packet Loss (%)                   High: 10       Low: 5
      Ping Test Interval (sec): 10
  Number of Pings Per Measurement Interval: 10

RTCP MONITOR SERVER
  Default Server IP Address: 192.168.15 .25
      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): 1
      Primary Search Time (sec): 15
      Periodic Registration Timer (min): 20
```

**Field descriptions**

The range for the **Periodic Registration Timer (min)** field is 1 to 120 minutes. The default value is 20 minutes. The purpose of this field is to set the time, in minutes, when a previously-registered IP telephone attempts to reregister. For example, a registered IP telephone becomes unregistered because another telephone has taken over the extension. After the number of minutes in the **Periodic Registration Timer (min)** field have elapsed, the unregistered telephone attempts to reregister.

If the reregistration is unsuccessful, the telephone again attempts to reregister. For example, if the **Periodic Registration Timer (min)** field is set to 20 minutes, the telephone attempts to reregister after 20 minutes has elapsed. If unsuccessful, the system waits another 20 minutes and tries again. This process continues until the system can reregister the telephone.

---

## Language Translations

Message 4, **^-party conference in progress**, is added to the **Language Translations** screen for transfer/conference. This change is prompted by the [Expanded Meet-me Conferencing](#) feature.

To view the **Language Translations** screen for transfer/conference:

1. Type **change display-messages transfer-conference**. Press **Enter**.

The system displays the **Language Translations** screen ([Figure 70: Language Translations screen](#) on page 155).

---

**Figure 70: 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: *****

```

---

**Note:**

You manually must change the character “^” in your user-defined language. Communication Manager does not update automatically. The character “^” is a place holder character.

**Message 4**

English Text	Replacement Info
^-party conference in progress	Communication Manager replaces the “^” character with the number of parties that are currently on the conference call.

---

## List Usage Report

The **List Usage Report** screen displays an IP telephone that is in TTI service. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

To view the **List Usage Report** screen:

1. Type `list usage ip-address n`, where `n` is the IP address that you want to review. Press **Enter**.

The system displays the **List Usage Report** screen ([Figure 71: List Usage Report screen](#) on page 156).

---

**Figure 71: List Usage Report screen**

```
list usage ip-address 135.9.159.130

                                LIST USAGE REPORT

Used By
TTI Station                      Port Number s00074                Station IP Addr
```

---

## Location Parameters

Two new fields, **International Access Code** and **Local E.164 Country Code**, are added to the **Location Parameters** screen. This change is prompted by the [Inter-Gateway Alternate Routing](#) feature.

To view the **Location Parameters** screen:

1. Type `change location-parameters n`, where `n` is the location code. Press **Enter**.

The system displays the **Location Parameters** screen ([Figure 72: Location Parameters screen](#) on page 157).

**Figure 72: Location Parameters screen**

```

change location-parameters 1                                     Page 1 of 4

                                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

```

**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.

**Media-Gateway**

A new field, **Recovery Rule**, is added to the **Media-Gateway** screen. This change is prompted by the [Auto fallback to primary for H.248 media gateways](#) feature.

## New and changed screens

To view the **Media-Gateway** screen:

1. Type `change media-gateway n`, where *n* is the number of the gateway. Press **Enter**.

The system displays the **Media-Gateway** screen ([Figure 73: Media-Gateway screen](#) on page 158).

**Figure 73: Media-Gateway screen**

```
change media-gateway 1                                     Page 1 of 1

                                MEDIA-GATEWAY

Number: 1                                                    IP Address:
Type: g250                                                    FW Version/HW Vintage:
Name: _____                                             MAC Address:
Serial No: _____                                         Encrypt Link? y
Network Region: _____                                     Location:
Registered? n                                                Controller IP Address:
Recovery Rule: _____                                     Site Data: _____
                                                            Name: _____

Slot   Module Type
V1:   _____
V2:   _____
V3:   _____
V4:   _____
V5:   _____
V6:   _____
V7:   _____
V8:   _____
V9:   _____

Max Survivable IP Ext:
```

### Recovery Rule

The **Recovery Rule** field allows you to associate an auto-fallback recovery rule with this media gateway. By associating the recovery rule to the **Media Gateway** screen, you can use the `list media-gateway` command to see what media gateways have the same recovery rules. All the administration parameters for the media gateways are consolidated on a single screen. The actual logic of the recovery rule is separate, but an administrator can start from the **Media Gateway** screen and find the recovery rule.

For more information, see the [System Parameters Media Gateway Automatic Recovery Rule](#) screen.

Valid entries	Usage
1 to the server maximum, or none	Type a valid recovery rule number. Type <b>none</b> to disable.

---

## Media-Gateway Report

Two new fields, **RecRule** and **Reg?**, are added to the **Media-Gateway Report** screen. These changes are prompted by the [Auto fallback to primary for H.248 media gateways](#) feature.

To view the **Media-Gateway Report** screen:

1. Type `list media-gateway`. Press **Enter**.

The system displays the **Media-Gateway Report** screen ([Figure 74: Media-Gateway Report screen](#) on page 159).

---

**Figure 74: Media-Gateway Report screen**

```
list media-gateway
```

MEDIA-GATEWAY REPORT						
Num	Name	Serial No/ FW Ver/HW Vint	IP Address/ Cntrl IP Addr	Type	NetRgn RecRule	Reg?
5	MG5	01DR12310234 220.66 .0 ./0	172.154.658.251	g700	205 none	n
7	MG5	01DR12310234 220.66 .0 ./0	172.154.658.251	g700	205 none	y
11	MG5	01DR12310234 220.66 .0 ./0	172.154.658.251	g700	205 1	n
19	MG5	01DR12310234 220.66 .0 ./0	172.154.658.251	g700	205 none	p

---

### RecRule

The **RecRule** field displays the auto-fallback recovery rule that is associated with the media gateway.

For more information, see the [System Parameters Media Gateway Automatic Recovery Rule](#) screen and the [Media-Gateway](#) screen.

### Reg?

The **Reg?** field displays the registration status of the media gateway. A new value, **p**, appears during a pending registration state.

---

## Optional Features

**Note:**

There are numerous changes to the **Optional Features** screen for Communication Manager release 3.0. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **Optional Features** screen:

1. Type `display system-parameters customer-options`. Press **Enter**.

The system displays the **Optional Features** screen ([Figure 75: Optional Features screen](#) on page 160).

---

**Figure 75: Optional Features screen**

```
display system-parameters customer-options                               Page 1 of 10

                                OPTIONAL FEATURES

G3 Version: V13
Location: 1                      RFA System ID (SID): 1
Platform: 6                      RFA Module ID (MID): 1

                                USED
Platform Maximum Ports: 44000 9605
Maximum Stations: 36000 7807
Maximum XMOBILE Stations: 100 1
Maximum Off-PBX Telephones - EC500: 5 1
Maximum Off-PBX Telephones - OPS: 5 3
Maximum Off-PBX Telephones - SCCAN: 0 0

(NOTE: You must logoff & logon to effect the permission changes.)
```

---

### Page 1

- A new field, **Maximum Stations**, is added to the **Optional Features** screen. This change is prompted by the [Station licensing](#) feature. This field tracks station licenses only. Customers can easily identify the number of station licenses on the system.
2. Click **Next** until you see page 2 ([Figure 76: Optional Features screen](#) on page 161).

Figure 76: Optional Features screen

```

display system-parameters customer-options
Page 2 of 10

OPTIONAL FEATURES

IP PORT CAPACITIES
Maximum Administered H.323 Trunks: 1500 419
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
Max Concur Registered Unauthorized H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 0 0
Maximum Video Capable IP Softphones: 0 0
Maximum Administered SIP Trunks: 450 20

Maximum Number of DS1 Boards with Echo Cancellation: 0 0
Maximum TN2501 VAL Boards: 1 0
Maximum G250/G350/G700 VAL Sources: 0 0
TN2602 80-VoIP Channel Licenses: 20 12
TN2602 320-VoIP Channel Licenses: 4 3

Maximum Number of Expanded Meet-me Conference Ports: 50 0

(NOTE: You must logoff & logon to effect the permission changes.)

```

## Page 2

- The **Maximum G250/G350/G700 VAL Sources** is renamed to include the new [G250 Media Gateway](#).
  - Two new fields, **TN2602 80-VoIP Channel Licenses** and **TN2602 320-VoIP Channel Licenses**, are added. These changes are prompted by the new [TN2602AP IP Media Resource 320 circuit pack](#).
  - A new field, **Maximum Number of Expanded Meet-me Conference Ports**, is added. This change is prompted by the [Expanded Meet-me Conferencing](#) feature.
3. Click **Next** until you see the **Enterprise Survivable Server?** field ([Figure 77: Optional Features screen](#) on page 162).

Figure 77: Optional Features screen

```
display system-parameters customer-options                               Page 4 of 10

                                OPTIONAL FEATURES

Emergency Access to Attendant? y                                     IP Stations? y
  Enable 'dadmin' Login? y                                           Internet Protocol (IP) PNC? n
  Enhanced Conferencing? y                                           ISDN Feature Plus? y
    Enhanced EC500? y                                               ISDN Network Call Redirection? n
Enterprise Survivable Server? n                                       ISDN-BRI Trunks? y
  Enterprise Wide Licensing? n                                       ISDN-PRI? y
    ESS Administration? y                                           Local Survivable Processor? n
  Extended Cvg/Fwd Admin? y                                           Malicious Call Trace? y
  External Device Alarm Admin? y                                       Media Encryption over IP? y
Five Port Networks Max Per MCC? n                                     Mode Code for Centralized Voice Mail? n
  Flexible Billing? n
Forced Entry of Account Codes? n                                       Multifrequency Signaling? n
  Global Call Classification? y   Multimedia Appl. Server Interface (MASI)? n
  Hospitality (Basic)? y       Multimedia Call Handling (Basic)? y
Hospitality (G3V3 Enhancements)? y   Multimedial Call Handling (Enhanced)? y
  IP Trunks? y

IP Attendant Consoles? y

(NOTE: You must logoff & logon to effect the permission changes.)
```

## Page 4

Two new fields, **Enterprise Survivable Server?** and **ESS Administration?**, are added to the **Optional Features** screen. This change is prompted by the [Enterprise Survivable Servers](#) feature.

### Field descriptions

For the main server:

- Make sure that the **Enterprise Survivable Server?** field is set to **n**.
- Make sure that the **ESS Administration?** field is set to **y**.

For each Enterprise Survivable Server:

- Make sure that the **Enterprise Survivable Server?** field is set to **y**.
- Make sure that the **ESS Administration?** field is set to **y**.

For a complete set of screens that apply to ESS, see the *Avaya Enterprise Survivable Server (ESS) Users Guide*, 03-300428.

---

## Port Information

The IP address of an IP telephone that is in TTI service appears in the **Identification** field on the **Port Information** screen. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

To view the **Port Information** screen:

1. Type `display port n`, where `n` is the port number. Press **Enter**.

The system displays the **Port Information** screen ([Figure 78: Port Information screen](#) on page 163).

---

**Figure 78: Port Information screen**

```
display port s00001

                                PORT INFORMATION

                                Port: S00001
                                Equipment Type: TTI port
                                Identification: 172.22.22.122
```

---

## Security-Related System Parameters

A new field, **Receive Unencrypted from IP Endpoints?**, is added to the **Security-Related System Parameters** screen. This change is prompted by the [Security of IP telephone registration/H.323 signaling channel](#) feature.

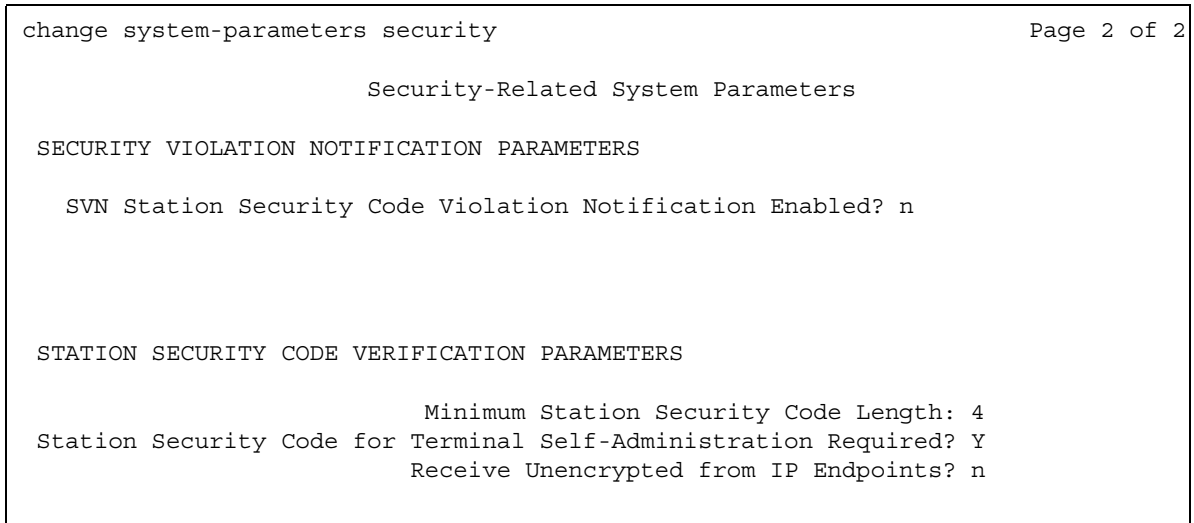
To view the **Security-Related System Parameters** screen:

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

The system displays the **Security-Related System Parameters** screen.

2. Click **Next** until you see the **Receive Unencrypted from IP Endpoints?** field ([Figure 79: Security-Related System Parameters screen](#) on page 164).

**Figure 79: Security-Related System Parameters screen**



**Field descriptions**

The system can transmit the user PIN in the clear for IP telephones that do not support encryption. System administrators have the option to enable or disable this capability.

---

## System Capacity

**Note:**

There are numerous changes to the **System Capacity** screen for Communication Manager release 3.0. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **System Capacity** screen:

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

The system displays the **System Capacity** screen.

2. Click **Next** until you see the **TN2602 80-VoIP Channel Licenses** field ([Figure 80: System Capacity screen](#) on page 165).

Figure 80: System Capacity screen

```

display capacity
                                                    Page 6 of 12

                SYSTEM CAPACITY

                Used   Available   Limit
                -----
Recorded Announcement Analog Queue Slots:    0      1000    1000
                TN2501 VAL Boards:           0       0*      0
                G700 VAL Sources:            1       9       10
                TN2602 Boards with 80-VoIP Channels: 12       8       20
                TN2602 Boards with 320-VoIP Channels: 3       1       4
TEMPORARY SIGNALING CONNECTIONS (TSC)
                Administered TSCs:           1      127     128
                NCA-TSC Calls:               0      256     256

'' Available VAL Boards limited by other inserted integ type annc boards

```

## Page 6

Two new fields, **TN2602 80-VoIP Channel Licenses** and **TN2602 320-VoIP Channel Licenses**, are added to the **System Capacity** screen. These display-only fields indicate the current usage, license limit, and available 80-VoIP and 320-VoIP channel license capacities that are associated with the system. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

3. Click **Next** until you see the **Concurrent Registration Counts** area ([Figure 81: System Capacity screen](#) on page 165).

Figure 81: System Capacity screen

```

display capacity
                                                    Page 10 of 12

                SYSTEM CAPACITY

                CONCURRENT REGISTRATION COUNTS

                Currently
                Registered   Available   System
                -----
                IP Stations:    1009      4091    5100
                IP Stations in TTI State:    0         -         -
                IP Attendant Consoles:      0         1         1
                Remote Office Stations:      3        997    1000

```

## Page 10

A new field, **IP Stations in TTI State**, is added to the **System Capacity** screen. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

---

## Vector Directory Number

**Note:**

There are numerous changes to the **Vector Directory Number** screen for Communication Manager release 3.0. The changes are all listed here, by page. The information also mentions what feature prompted the change.

To view the **Vector Directory Number** screen:

1. Type `change vdn n`, where *n* is a valid vector directory number. Press **Enter**.

The system displays the **Vector Directory Number** screen ([Figure 82: Vector Directory Number screen](#) on page 166).

---

**Figure 82: Vector Directory Number screen**

```
change vdn 5000                                     Page 1 of x
                                                    VECTOR DIRECTORY NUMBER
                                                    Extension: 5000
                                                    Name:
                                                    Vector Number: 234
Attendant Vectoring: n
Meet-me Conferencing? n
Allow VDN Override? n
                                                    COR: 59
                                                    TN: 1
                                                    Measured: none
Acceptable Service Level (sec):
VDN of Origin Annc. Extension: 301
                                                    1st Skill:
                                                    2nd Skill:
                                                    3rd Skill:
```

## Page 1

A new field, **Meet-me Conferencing?**, is added to the **Vector Directory Number** screen. This change is prompted by the [Expanded Meet-me Conferencing](#) feature.

The **Meet-me Conferencing?** field appears only if the **Enhanced Conferencing** field is set to **y** on the **Optional Features** screen. This field determines if the VDN is a Meet-me Conferencing VDN.

### Note:

If the VDN extension is part of your DID block, external users can access the conference VDN. If the VDN extension is not part of your DID block, only internal callers on your network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Enter <b>y</b> to enable Meet-me Conferencing for this VDN. If Meet-me Conferencing is <b>y</b> , only <b>Extension, Name, Vector Number, Meet-me Conference, COR,</b> and <b>TN</b> fields display and the fields for page 2 change. Both <b>Attendant Vectoring</b> and <b>Meet-me Conferencing</b> cannot be enabled at the same time. If <b>Enhanced Conferencing</b> is <b>y</b> , but no other vectoring options are enabled, only Meet-me Conferencing vectors can be assigned.

- Click **Next** until you see the **Meet-me Conference Parameters** area ([Figure 83: Vector Directory Number screen](#) on page 167).

**Figure 83: Vector Directory Number screen**

change vdn 5000	VECTOR DIRECTORY NUMBER	Page 2 of x
MEET-ME CONFERENCE PARAMETERS		
Conference Access Code:123456		
Conference Controller:		
Conference Type: expanded		
Route-to Number:		

## Page 2

### Conference Access Code

To ensure conference security, you should always assign an access code to a Meet-me Conferencing VDN.

Valid entries	Usage
6-digit number or blank	Enter a 6-digit access code for the Meet-me Conferencing 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 Conferencing VDN using a feature access code. This can be a local user or someone dialing in through 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
<b>6-party</b> or <b>expanded</b>	Type <b>expanded</b> to enable the Expanded Meet-me Conferencing feature. The default is <b>6-party</b> .

**Route-to Number**

The **Route-to Number** 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. Or you can enter the unique UDP extension. The <b>Route-to Number</b> must be unique across all Expanded Meet-me Conferencing VDNs.

## **New and changed screens**

# Chapter 4: New and changed commands

This chapter displays the new and changed commands for Avaya Communication Manager.

---

## Release 3.1 new commands

Avaya Communication Manager, release 3.1, includes the following new commands.

---

### add off-pbx-telephone station-mapping

A new command, `add off-pbx-telephone`, displays a new screen, **Stations with Off-PBX Telephone Integration**. For more information, see the [Stations with Off-PBX Telephone Integration](#) on page 119.

#### Valid parameters

Action	Object	Qualifier
<code>add</code>	<code>off-pbx-telephone station-mapping</code>	
<code>change</code>	<code>off-pbx-telephone station-mapping</code>	<code>&lt;station extension&gt;</code>
<code>display</code>	<code>off-pbx-telephone station-mapping</code>	<code>&lt;station extension&gt;</code>
<code>list</code>	<code>off-pbx-telephone station-mapping</code>	<code>&lt;variable&gt;</code>

- 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.

## New and changed commands

- 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 displays 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

---

## Release 3.0 new commands

Avaya Communication Manager, release 3.0, includes the following new commands.

---

### add audio-group

A new command, `add audio-group n`, where `n` is the audio group number, displays a new screen, **Audio Group**. For more information, see the [Audio Group](#) on page 77. This change is prompted by the [Locally sourced announcements and music](#) feature.

#### Valid parameters

Action	Object	Qualifier
<code>add</code>	<code>audio-group</code>	<code>n   next</code>
<code>change</code>	<code>audio-group</code>	<code>n</code>
<code>remove</code>	<code>audio-group</code>	<code>n</code>
<code>display</code>	<code>audio-group</code>	<code>n   n to n</code>

---

## add moh-analog-group

A new command, `add moh-analog-group n`, where `n` is the music-on-hold group number, displays a new screen, **MOH Group**. For more information, see the [MOH Group](#) on page 79. This change is prompted by the [Locally sourced announcements and music](#) feature.

### Valid parameters

Action	Object	Qualifier
add	moh-analog-group	number <code>n</code>   next
change	moh-analog-group	number <code>n</code>
remove	moh-analog-group	number <code>n</code>
display	moh-analog-group	number <code>n</code>   to-number <code>n</code>
list	moh-analog-group	[1-Max] (number <code>n</code>   (to-number <code>n</code> )   (count <code>n</code> )

---

## change system-parameters mg-recovery-rule

A new command, `change system-parameters mg-recovery-rule n`, where `n` is the recovery rule number, displays a new screen, **System Parameters Media Gateway Automatic Recovery Rule**. For more information, see the [System Parameters Media Gateway Automatic Recovery Rule](#) on page 81. This change is prompted by the [Auto fallback to primary for H.248 media gateways](#) feature.

---

## display virtual-mac-address

A new command, `display virtual-mac-address n`, where `n` is the MAC address table number, displays a new screen, **Virtual MAC Addresses**. For more information, see the [Virtual MAC Addresses](#) on page 86. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

---

## enable filexfer

Two new commands, `enable filexfer n` and `disable filexfer n`, where `n` is the slot location of a circuit pack, act as a toggle switch, and enable a file transfer protocol (ftp) session.

**Note:**

These commands are intended for use on the bearer 10/100 Ethernet port on the back plane. These commands are not intended for use on the services port.

This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

**Valid parameters**

Action	Object	Qualifier
<code>enable</code>	<code>filexfer</code>	<code>n</code>
<code>disable</code>	<code>filexfer</code>	<code>n</code>

---

## enable session

Two new commands, `enable session` and `disable session <circuit pack slot location>`, act as a toggle switch, and enable a telnet session.

**Note:**

These commands are intended for use on the bearer 10/100 Ethernet port on the back plane. These commands are not intended for use on the services port.

This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

**Valid parameters**

Action	Object	Qualifier
<code>enable</code>	<code>session</code>	
<code>disable</code>	<code>session</code>	<code>&lt;circuit pack slot location&gt;</code>

---

## list audio-group

A new command, `list audio-group`, displays a new screen, **Audio Groups**. For more information, see the [Audio Groups](#) on page 77. This change is prompted by the [Locally sourced announcements and music](#) feature.

### Valid parameters

Action	Object	Qualifier
<code>list</code>	<code>audio-group</code>	

---

## list ip-interface medpro

A new command, `list ip-interface medpro`, displays a new screen, **IP Interfaces**. For more information, see the [IP Interfaces](#) on page 78. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

---

## list moh-analog-group

A new command, `list moh-analog-group`, displays a new screen, **Music-On-Hold Groups**. For more information, see the [Music-on-Hold Groups](#) on page 81. This change is prompted by the [Locally sourced announcements and music](#) feature.

### Valid parameters

Action	Object	Qualifier
<code>list</code>	<code>audio-group</code>	

---

## list tti-ip-stations

A new command, `list tti-ip-stations`, displays a new screen, **TTI Service IP Stations**. This new screen lists IP telephones that are in TTI service. For more information, see the [TTI Service IP Stations](#) on page 86. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

---

## list usage integ-annc-board

A new command, `list usage integ-annc-board n`, where `n` is the 5-character circuit pack location number, displays a new screen, [Announcement Group Board Usage](#). This change is prompted by the [Locally sourced announcements and music](#) feature.

### Valid parameters

Action	Object	Qualifier
<code>list</code>	<code>usage integ-annc-board</code>	<code>n</code>

---

## reset media-gateway

`reset media-gateway [ n | all ] level [ 1 | 2 | 3 ]`

Use the `reset media-gateway` command from the primary server to reset one or more media gateways. This change is prompted by the [Connection preserving failover/failback for H.248 media gateways](#).

Action/Object	Qualifier	Qualifier Description
<code>reset media-processor</code>	<code>n</code>	The specific media gateway number to reset. This command drops all connections to the specified media gateway.
	<code>all</code>	All registered media gateways. This command drops all connections to all media gateways.
	<code>level 1</code>	The <code>reset media-gateway level 1</code> command forces a reset of the entire platform and is destructive to user connections. The media gateway attempts to register with the media gateway controllers on its MGC list.
	<code>level 2</code>	The <code>reset media-gateway level 2</code> command resets the H.248 link and does not tear-down calls. The media gateway attempts to register with the media gateway controllers on its MGC list. Use <code>reset media-gateway level 2</code> to force a media gateway off of an LSP.
	<code>level 3</code>	The <code>reset media-gateway level 3</code> command resets all media modules and tears down all calls.

## set media-processor

```
set media-processor location [ lock | unlock ] [ override ]
```

Use the `set media-processor` command to request a demand interchange of IP Media Resource 320 circuit packs on duplicated circuit packs only. Use `set media-processor location lock` to manually specify the active TN2602AP IP Media Resource 320 circuit pack when duplicated TN2602AP circuit pack exist in a port network. For more information, see the [TTI Service IP Stations](#) on page 86. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

Action/Object	Qualifier	Qualifier Description
<code>set media-processor</code>	<code>location</code>	location of the Media Resource to be active
	<code>lock</code>	circuit packs remain in their current state (active or standby)
	<code>unlock</code>	clear the locked state
	<code>override</code>	force an interchange to a less-healthy board

When the `set media-processor` command does not produce an interchange, an error message appears.

SAT error message	Description
Command only supported by a TN2602AP and greater board	The board location specified is not a TN2602 IP Media Resource. Use <code>list config</code> to verify the TN code and identify the board in this location.
Duplication not administered for this media-processor	This IP Media Resource is not administered as a duplicated board. Use <code>display ip-interface</code> to verify administration of the board.
Invalid duplication state for this media-processor pair	This pair of duplicated IP Media Resources has not transitioned to a state where one is active and one is standby. Use <code>status media-processor</code> to verify the duplication status of the IP Media Resources.
Standby media-processor is not refreshed; use "override"	The standby IP Media Resource does not have the same set of calls up as the active board. An interchange making the standby active would cause a loss of some or all of the calls. Use <code>set media-processor location override</code> to ignore the warning and continue the interchange.

---

## Release 3.1 changed commands

Avaya Communication Manager, release 3.1, includes the following changed commands.

---

### change public-unknown-numbering

`change public-unknown-numbering n [ext-digits]`

Use the `change public-unknown-numbering n` command to administer the desired digits for name and number display on display-equipped stations in an ISDN network.

Action/Object	Qualifier	Qualifier Description
<code>change public-unknown-numbering</code>	<code>n</code>	Number of digits (extension length, <b>Ex -Len</b> ) in the extension being administered. Enter <b>0</b> for attendant.
	<code>ext-digits</code>	Displays the first extension on the screen.  Example: <code>change public-unknown-numbering 5</code> <code>change public-unknown-numbering 5 ext-digits 10010</code>

See *Administrator Guide for Avaya Communication Manager, 03-300509*, for a screen example and field descriptions, and for more information on ISDN Call Identification Display and Numbering-Public/Unknown.

---

### list registered-ip-stations

The `list registered-ip-stations` command has a new option, `count n`, where `n` is a number of extensions that you want to list.

When used with the `ext` option, the system administrator can now list IP telephones within a specific range and “count” a specific number of consecutive extensions from that beginning number. Previously, the user had to list all registered IP telephones and scroll until the user found the range of extensions.

For example, if you want to list three consecutive registered IP endpoints beginning with extension 1000000, type `list registered-ip-stations ext 1000000 count 3`. The system displays information on the **Registered IP Stations** screen ([Figure 84: Registered IP Stations screen](#) on page 179):

Figure 84: Registered IP Stations screen

```
list registered-ip-stations ext 1000000 count 3
```

REGISTERED IP STATIONS								
Station Ext	Set Type	Product ID	Prod Rel	Station IP Address	Net Rgn	Orig Port	Gatekeeper IP Address	TCP Skt
1000000	MDW9000	1234567890	9.123	100.101.102.103	250	01A0404	123.124.125.126	y
1000001	MDW9000	1234567890	9.123	1.2.3.4	1		16.17.18.19	y
1000001	MDW9000	2222222222	2.222	2.2.2.2	1		32.33.34.35	n
1000002	8434D	1212121212	8.888	5.6.7.8	2	250V101	16.17.18.21	n
1000002	8434D	2222222222	2.222	2.2.2.2	2	250V101	32.33.34.35	y

## Release 3.0 changed commands

Avaya Communication Manager, release 3.0, includes the following changed commands.

### get boot-image

The system now recognizes a **Code** type of **TN2602AP** to support the [TN2602AP IP Media Resource 320 circuit pack](#).

### reset ip-stations

The `reset ip-stations` command has a new option, `tti`. The complete command is now:

```
reset ip-stations [ip-phones | all | tti] [network-region(1-250) |
all-regions].
```

The system administrator can use the `reset ip-stations tti` command to unregister IP telephones that are in TTI service without unregistering other IP telephones. This change is prompted by the [Emergency calls from unnamed IP endpoints](#) feature.

## set boot-image

The system now recognizes a **Code** type of **TN2602AP** to support the [TN2602AP IP Media Resource 320 circuit pack](#).

---

## status ip-board

The system now recognizes and provides status for a TN2602AP circuit pack. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

---

## status media-processor

The system now recognizes the argument **a11** to display all media processor circuit packs in a system. This change is prompted by the [TN2602AP IP Media Resource 320 circuit pack](#).

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