AVAYA

Highlights of

Avaya Communication Manager

The LASER devices operate within the following parameters:

- Maximum power output: -5 dBm to -8 dBm
- Center Wavelength: 1310 nm to 1360 nm

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Klass 1 Laser Apparat

Use of controls or adjustments or performance of procedures other than those specified herein may result in hazardous radiation exposures. Contact your Avaya representative for more laser product information.

Electromagnetic Compatibility (EMC) Standards

This product complies with and conforms to the following international EMC standards and all relevant national deviations:

Limits and Methods of Measurement of Radio Interference of Information Technology Equipment, CISPR 22:1997 and EN55022:1998.

Information Technology Equipment – Immunity Characteristics – Limits and Methods of Measurement, CISPR 24:1997 and EN55024:1998, including:

- Electrostatic Discharge (ESD) IEC 61000-4-2
- Radiated Immunity IEC 61000-4-3
- Electrical Fast Transient IEC 61000-4-4
- Lightning Effects IEC 61000-4-5
- Conducted Immunity IEC 61000-4-6
- Mains Frequency Magnetic Field IEC 61000-4-8
- Voltage Dips and Variations IEC 61000-4-11
- Powerline Harmonics IEC 61000-3-2
- Voltage Fluctuations and Flicker IEC 61000-3-3

Federal Communications Commission Statement

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Part 68: Answer-Supervision Signaling

Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:

- answered by the called station,
- answered by the attendant, or
- routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.

This equipment returns answer-supervision signals on all direct inward dialed (DID) calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

Avaya attests that this registered equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

REN Number

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

This equipment complies with Part 68 of the FCC rules. On either the rear or inside the front cover of this equipment is a label that contains, among other information, the FCC registration number, and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

For G350 and G700 Media Gateways:

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the rear of this equipment is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. The digits represented by ## are the ringer equivalence number (REN) without a decimal point (for example, 03 is a REN of 0.3). If requested, this number must be provided to the telephone company.

For all media gateways:

The REN is used to determine the quantity of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0. To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

REN is not required for some types of analog or digital facilities.

Means of Connection

Connection of this equipment to the telephone network is shown in the following tables.

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Off premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
	02LS2	0.3A	RJ21X
Tie trunk	TL31M	9.0F	RJ2GX
Basic Rate Interface	02IS5	6.0F, 6.0Y	RJ49C
1.544 digital interface	04DU9-BN	6.0F	RJ48C, RJ48M
	04DU9-IKN	6.0F	RJ48C, RJ48M
	04DU9-ISN	6.0F	RJ48C, RJ48M
120A4 channel service unit	04DU9-DN	6.0Y	RJ48C

For G350 and G700 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Ground Start CO trunk	02GS2	1.0A	RJ11C
DID trunk	02RV2-T	AS.0	RJ11C
Loop Start CO trunk	02LS2	0.5A	RJ11C
1.544 digital interface	04DU9-BN	6.0Y	RJ48C
	04DU9-DN	6.0Y	RJ48C
	04DU9-IKN	6.0Y	RJ48C
	04DU9-ISN	6.0Y	RJ48C
Basic Rate Interface	02IS5	6.0F	RJ49C

For all media gateways:

If the terminal equipment (for example, the media server or media gateway) causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact the Technical Service Center at 1-800-242- 2121 or contact your local Avaya representative. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. It is recommended that repairs be performed by Avaya certified technicians.

The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Declarations of Conformity

United States FCC Part 68 Supplier's Declaration of Conformity (SDoC)

Avaya Inc. in the United States of America hereby certifies that the equipment described in this document and bearing a TIA TSB-168 label identification number complies with the FCC's Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA) adopted technical criteria.

Avaya further asserts that Avaya handset-equipped terminal equipment described in this document complies with Paragraph 68.316 of the FCC Rules and Regulations defining Hearing Aid Compatibility and is deemed compatible with hearing aids.

Copies of SDoCs signed by the Responsible Party in the U. S. can be obtained by contacting your local sales representative and are available on the following Web site: http://www.avaya.com/support.

All Avaya media servers and media gateways are compliant with FCC Part 68, but many have been registered with the FCC before the SDoC process was available. A list of all Avaya registered products may be found at: http://www.part68.org by conducting a search using "Avaya" as manufacturer.

European Union Declarations of Conformity

Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformité Europeénne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC). This equipment has been certified to meet CTR3 Basic Rate Interface (BRI) and CTR4 Primary Rate Interface (PRI) and subsets thereof in CTR12 and CTR13, as applicable.

Copies of these Declarations of Conformity (DoCs) can be obtained by contacting your local sales representative and are available on the following Web site: http://www.avaya.com/support.

Japan

This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective actions.

この装置は、情報処理装置等電波障害自主規制協議会（ＶＣＣＩ）の基準に基づくクラスＡ情報技術装置です。この装置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を講ずるよう要求されることがあります。

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About this book

Overview

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

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

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

Purpose of this book

This book describes the new and changed features and enhancements available with the most recent release of Communication Manager (Release 2.0) running on any of the following:

- Avaya media servers
    - DEFINITY® servers
    - S8100, S8300, S8500, or S8700 Media Servers
- Avaya media servers configured as a Local Survivable Processor (LSP).
- Avaya media gateways
    - MCC1 or SCC1 Media Gateways
    - G350, G600, G650, or G700 Media Gateways

      NOTE:
      This document does not contain information about prior releases of Communication Manager. For highlight 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 13 for more information).

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

Intended audience

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

Contents

This document includes the following chapters:

- Highlights — 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.

Conventions used

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

- Names of screens are printed in bold italic as follows: *screen name*.

  A "screen" is the display of fields and prompts that appear on a computer or terminal monitor. See page 39 for an example of a screen and how it is shown in this book.

- Keys on a keyboard, buttons that appear on a screen, and text that you need to type are printed in bold as follows: **Key**.

- Commands are printed in bold as follows: **command**.

  We show complete commands in this book, but you can usually type an abbreviated version of the command. For example, **list configuration station** can be typed as **list config sta**.

- Command variables are printed in bold italic as follows: *command variable*.

- Anything that displays on a screen in response to a command or other input is shown in a constant width font as follows: `Command successfully completed`.

- To move to a certain field, you can use the **Tab** key, **arrow** keys, or the **Enter** key (the **Enter** key may appear as the **Return** key or the **Submit** key on your keyboard).

- We show commands and screens from the most recent release of Communication Manager and refer to the most current books. Please substitute the appropriate commands for your system and refer to the manuals you have available.

- When a procedure requires you to press **Enter** to save your changes, the screen you were working on clears and the cursor returns to the command prompt.

  The message line shows "`Command successfully completed`" to indicate that the system accepted your changes.

- If you need help constructing a command or completing a field entry, remember to use **Help**.

  — When you press **Help** at any point on the command line, a list of available commands appears.

  — When you press **Help** with your cursor in a field on a screen, a list of valid entries for that field appears.

- The status line or message line can be found near the bottom of your monitor display. 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 our helpline.

Admonishments

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

> **NOTE:**
> Highlights information that you should pay attention to.

> **CAUTION:**
> Denotes possible harm to software, possible loss of data, or possible service interruptions.

> **WARNING:**
> Denotes possible harm to hardware or equipment.

> **DANGER:**
> **Denotes possible harm or injury to your body.**

> **SECURITY ALERT:**
> Denotes when system administration may leave your system open to toll fraud.

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How to obtain Avaya books on the Web

If you have internet access, you can view and download the latest version of Avaya documentation products. To view any book, you must have a copy of Adobe Acrobat Reader.

> **NOTE:**
> If you don't have Acrobat Reader, you can get a free copy at http://www.adobe.com.

For example, to access an electronic version of this book:

1   Access the Avaya Web site at http://www.avaya.com/support/.
2   Click **Product Documentation**.

    3   To find a specific book, type the document number (for example, 555-233-783 for this book) in the **Search Support** text box, and then click **GO**.

    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.

How to order documentation

In addition to this book, other description, installation and test, maintenance, and administration books are available.

This document and any other Avaya documentation can be ordered directly from the Avaya Publications Center toll free at 1-800-457-1235 (voice) and 1-800-457-1764 (fax). Customers outside the United States should use +1-410-568-3680 (voice) and +1-410-891-0207 (fax).

How to comment on this book

Avaya welcomes your feedback. Contact us through:

- e-mail: document@avaya.com
- fax: 1-303-538-1741
- Contact your Avaya representative

Mention this document's name and number, *Highlights of Avaya Communication Manager*, 555-233-783.

Your comments are of great value and help improve our documentation.

How to get help

If you suspect that you are being victimized by toll fraud and you need technical assistance or support in the United States and Canada, call the Technical Service Center's Toll Fraud Intervention Hotline at 1-800-643-2353.

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.

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

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

1   Highlights

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 a more complete overview of all the features of Communication Manager, see the *Overview for Avaya Communication Manager*, 555-233-767.

For more information on how to administer any of these features, see the *Administrator's Guide for Avaya Communication Manager*, 555-233-506.

Release 2.0 new features and enhancements

Avaya Communication Manager, Release 2.0, includes the following general telephony and system-wide features and enhancements.

Adjunct route support for network call redirection

This feature provides the capability to invoke network call redirection (NCR) through the route request response to an adjunct route vector step. This allows a CTI application to directly utilize NCR for redirecting an incoming call in the PSTN through the ASAI adjunct routing application.

The redirection request, along with the PSTN redirected to a telephone number, is included in the route select message from the adjunct. The redirect request invokes whatever format of network redirection that is assigned to the trunk group for the incoming call in the same manner as a vector invoked NCR. Information forwarding to the redirected destination is supported in the same manner as a vector invoked NCR.

This capability functions with either the network transfer type (the switch sets up the second leg of a call), or the network deflection type (the PSTN sets up the second leg of a call) of NCR protocols.

AES encryption algorithm for bearer channels

To provide privacy for media streams that are carried over IP networks, Communication Manager supports media encryption for the:

- H.248 signaling channel between the Media Server and the Media Gateway
- H.323 signaling channel between the Media Gateway and IP endpoints
- IP bearer (voice) channel

Digitally encrypting these channels can reduce the risk of electronic eavesdropping. IP packet monitors, sometimes called sniffers, are to VoIP calls what wiretaps are to circuit-switched (TDM) calls, except that an IP packet monitor can watch for and capture unencrypted IP packets and can play back the bearer channel conversation in real-time or store it for later playback. Comprehensive network security requires encrypting the signaling channels that might carry user-dialed authorization codes and passwords.

Communication Manager supports the Advanced Encryption Standard (AES) format of signal encryption for IP telephony. This encryption algorithm is in addition to Avaya's proprietary encryption protocol, the Avaya Encryption Algorithm (AEA).

AES encryption is a cryptographic algorithm developed by the U.S. Government to protect unclassified information. Communication Manager uses AES with 128 bit keys in counter mode (AES-128-CTR).

Administration is supported to select a combination of no encryption, AEA encryption, and/or AES encryption on a per codec set basis.

ASAI user to user information (UUI) to CMS

The ASAI user-to-user information (UUI) that is stored for a call is passed to the call management system (CMS). The ASAI UUI information (maximum of 96 bytes) is passed to the CMS in a message for storage in the external call history (ECH) call record.

This information can be used to record ASAI user data for the Variables for Vectors feature and for other applications. The UUI data is sent to CMS every time the call is routed to a VDN. This is particularly useful to supplement the Variables for Vectors feature where an "**asaiuui**" variable type is used to make decisions in vector processing for a call.

Avaya extension to cellular/OPS

Avaya extension to cellular and off-PBX stations (OPS) provides users with the capability to have one administered phone that supports Avaya Communication Manager features for both an office phone and one outside phone. Extension to cellular/OPS allows users to receive and place office calls anywhere, any time. People calling into an office phone can reach users even if they are not in the office. Users could receive the call on their cell phone, for example. This added flexibility also allows them to use certain Communication Manager features from a phone that is outside the phone network.

Extension to cellular

Previous versions of extension to cellular allowed for office calls to be extended to a user's cell phone. Also, calls from the cell phone would appear as if the call originated from the user's office phone when calling another phone on the same call server. Certain features within Communication Manager are available from the cell phone. These features are still available.

In previous versions of extension to cellular, cell phones had to be administered as XMOBILE stations. This is no longer necessary with Communication Manager Release 2.0.

If you had administered extension to cellular in earlier releases of Communication Manager, you do not have to change the administration to continue using extension to cellular features. It still works. However, users would not have the full range of features that are now possible with extension to cellular/OPS.

Off-PBX station (OPS)

New with Avaya Communication Manager Release 2.0, the off-PBX station (OPS) application type is used to administer a Session Initiation Protocol (SIP) phone. OPS cannot be disabled using the extension to cellular enable/disable feature button.

> **NOTE:**
> A SIP phone, such as the 4602 SIP phone, must register with the SIP proxy regardless of whether OPS is administered.

The extension to cellular/OPS application allows for many of the parameters used for the original extension to cellular application to be ported onto one of several DCP and IP station types. From a call processing perspective, extension to cellular/OPS is dealing with a multi-function phone, whereas the previous extension to cellular implementation utilized one or two XMOBILE stations that behaved like analog station types.

What's new in extension to cellular/OPS

In addition to features in previous releases of Communication Manager and extension to cellular, extension to cellular/OPS enhancements for Release 2.0 include the following:

- The ability to support Session Initiation Protocol (SIP) phones.

- Simplified administration of extension to cellular/OPS.

- The ability for the administrator to map certain Communication Manager features to phone extensions. You just dial one of those extensions – called a feature name extension (FNE) – from your cell phone to activate a Communication Manager feature.

- An extension to cellular feature button that allows you to extend a call that is received at the office phone, and extend it to your cell phone.

- An extension to cellular feature button that allows you to exclude anyone from joining in on your conversation from your office phone.

- The ability to administer the number of call appearances that are allowed to be mapped to the cell phone or SIP phone.

The extension to cellular feature buttons are available on office phones that support administrable feature buttons. This includes the feature button to enable and disable cell phones. You administer the buttons onto the office phone – the phone to which the extension to cellular is linked.

> **NOTE:**
> SIP-enabled phones cannot be enabled or disabled using a feature button.

Avaya integrated management

Avaya integrated management is a systems management software suite that contains applications to manage a converged voice and data network. The applications include:

- network management

- fault management

- performance management

- configuration management

- directory management
- policy management functionality

Avaya interactive response (IR)

The Avaya interactive response (IR) — formerly known as INTUITY Conversant® — voice information system is an interactive voice-response system that automates phone call transactions from simple tasks, like routing to the right department, to complex tasks, such as registering college students or providing bank balances. It communicates with customers in natural-sounding, digitally recorded speech, and performs 24-hours a day without the services of an operator.

The system can handle single or multiple voice-response applications simultaneously, and can serve up to 48 callers at once. It can operate by itself to dispense information or collect data, or it can work with a host computer to access a large database such as bank account records. With its speech-recognition capability, even rotary telephone users can have access to sophisticated phone-based services. Advanced telephone features provide intelligent call-transfer capabilities and allow you to use the system in your existing telephone environment.

Call admission control (CAC) bandwidth management

In order to ensure quality of service for voice over IP (VoIP) calls, there is a need to limit overall VoIP traffic on WAN links. The call admission control (CAC) bandwidth management feature of Communication Manager allows the customer to specify a VoIP bandwidth limit between any pair of IP network regions. The feature then denies calls that need to be carried over the WAN link that exceed that bandwidth limit.

Camp-on/Busy-out

A **camp-on/busy-out** command is commonly used by system technicians to busy-out system resources that need maintenance or repair. Without it, all active calls using those resources are indiscriminately dropped if the resource is physically removed from the system. This disruptive action causes problems for customers, especially when a large number of calls are torn down.

The camp-on/busy-out feature adds the ability to remove idle VoIP resources from the system's pool of available VoIP resources.

> **NOTE:**
> This feature is not supported by the G700 or G350 Media Gateway platforms.

The camp-on/busy-out feature enables the user to select the media processor to be busied-out while the media processor is still in service. After a call ends that was using resources within the specified media processor, the idled resource is removed from the system's pool of available resources. Once all of the media processor's resources are in a "busy-out" state, the associated board can be removed from the system without disrupting active calls.

CLAN load balancing

CLAN load balancing is the process of registering IP endpoints to CLAN circuit packs (TN799x). Load balancing occurs among CLANs within a network region.

IP endpoint registration among CLAN circuit packs is done through an algorithm. This algorithm tracks the number of sockets being used per TN799x circuit pack, and registers IP endpoints to the TN799x with the most available (unused) sockets. This algorithm applies to H.248, H.323 signaling groups, H.323 stations, and SIP endpoints. Sockets used by adjuncts are not included in the socket count.

Communication Manager API

Communication Manager 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.

Communication Manager 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, Communication Manager API does not run co-resident with Communication Manager.

Control network on customer LAN

This enhancement eliminates the need for a private IP control network for S8700 Multi-Connect configurations. In previous releases of Communication Manager, there is a stipulation that requires that S8700 servers, the Cajun Ethernet switches, and IPSI boards be the only components delivering traffic on this network. This enhancement eliminates this restriction and affords the option to implement the control network on customer LAN, as is done for the S8700 IP-Connect systems.

Converged communications server

The converged communications server adds a leading edge set of features to current enterprise customers. Through SIP, enterprise customers can send text instant messages through an instant message client integrated with the Avaya IP Softphone.

A presence-enabled buddy list gives enterprise users information on when colleagues are available. The converged communications server includes a SIP proxy, registrar, and presence server, as well as an instant message gateway to the popular consumer instant messaging networks (such as AOL, MSN, and Yahoo).

Disk survivability

The disk survivability feature allows call processing to continue in the event of a hard disk outage. Survivability of disk outages is provided by implementing the root (/) file system in a RAM disk partition, such that all critical files and data are physically stored in RAM disk space instead of on the hard disk media. Non-essential files and data are accessed indirectly from the RAM disk file system by means of links established by boot scripts.

For Communication Manager Release 2.0, this feature is only available with the S8500 Media Server.

E911 ELIN for wired IP extensions

This feature automates the process of assigning an emergency location information number (ELIN) through an IP subnetwork ("subnet") during an emergency 911 call. The ELIN is then sent over either CAMA or ISDN PRI trunks to the emergency services network when 911 is dialed. This feature properly identifies locations of wired IP telephones that call an emergency number from anywhere on a campus or location.

> **NOTE:**
> This feature depends upon the customer having subnets that correspond to geographical areas.

This feature works on H.323 IP endpoints. For SIP IP endpoints, you must use the off-PBX station (OPS) option of the Avaya extension to cellular feature (see).

A caller who needs emergency assistance dials a Universal Emergency Number — for example, 911 in the United States, 000 in Australia, and 112 in the European community. The call routes through a local Central Office, through an emergency tandem office, to the appropriate Public Safety Answer Point (PSAP). The PSAP answers the call.

A typical tandem office can route the call to a PSAP within at most four surrounding areas. (In the US, that translates to four surrounding area codes.) If the PSAP that receives the call is not the correct one to handle the emergency, the PSAP might be able to transfer the call to the correct PSAP. Such transfers can only occur between geographically adjacent or nearby PSAPs.

Each PSAP usually covers one city or one rural county. At the PSAP, emergency operators determine the nature of the emergency and contact the appropriate agency: police, fire, ambulance, etc. A single PSAP is usually responsible for an area covering several independent police and fire departments in the United States.

With Enhanced 911 (E911), the system might send to the emergency services network the Calling Party Number (CPN) with the call over Centralized Automatic Message Accounting (CAMA) trunks or through the Calling Number IE over ISDN trunks. The PSAP uses the CPN to lookup the caller's documented street address location from the Automatic Location Information (ALI) database. The ALI database is usually owned and managed by Local Exchange Carriers. Many enterprise customers choose to contract with a third party to update the ALI database for them.

This depends on the assumption that a CPN always corresponds to the street address that the system owner arranged to have administered into the ALI database. This assumption is not always true.

- Users who have H.323 IP telephones can move them without notifying the system administrator.
- Users who have SIP IP telephones can use the same extension number simultaneously at several different telephones.

Without this feature, if these users dial 911, the emergency response personnel might go to the wrong physical location. With this feature, the emergency response personnel can now go to the correct physical location. In addition, emergency response personnel can now go to the correct physical location if a 911 emergency call comes from a bridged call appearance.

ETSI explicit call transfer (ECT) signaling

The network call redirection (NCR) support of the "ETSI Explicit Call Transfer" feature is desired by multi-site, non-U.S. Avaya call center customers who use various PSTN service providers for ISDN services. These non-U.S. call centers wish to accomplish call transfers between sites without holding the ISDN trunks of a transferred call at the call redirecting Communication Manager site.

The Network Call Redirection/Network Call Deflection (NCR/NRD) feature does not allow for announcement and call-prompting call-vectoring operations. Therefore, the ETSI ECT feature is for these call center customers who cannot use NCR/NRD since they wish to play an announcement to a caller and use Communication Manager call-prompting to allow the caller to determine the routing for the call.

H.248 link encryption

To provide privacy for media streams carried over IP networks, the H.248 signaling channel between the media server (media gateway controller) and the media gateways is encrypted. This signalling channel is used to distribute the media session keys to the media gateways, and may carry user-dialed authorization codes and passwords.

This feature protects our customer investments by encrypting the signaling channel between the gateway — including the G700 and G350 Media Gateways — and server (using proprietary technology). This feature also protects media encryption key, PINs, and account codes between the media gateway and the media gateway controller.

Encryption of the H.248 link to any given media gateway may be enabled or disabled through the Media Gateway screen. The encryption protocol cannot be disabled.

H.323 link recovery

This feature supports the detection of and recovery from a failed H.323 signaling channel link between an H.323 IP endpoint and an Avaya media server running Communication Manager Release 2.0.

The H.323 link between an Avaya Media Gateway and an H.323-compliant IP endpoint provides the signaling protocol for:

- Call setup
- Call control (user actions such as Hold, Conference, or Transfer) while the call is in progress
- Call tear-down

If the link goes down, H.323 Link Recovery preserves any existing calls and attempts to re-establish the original link. If the endpoint cannot reconnect to the original gateway, then H.323 Link Recovery automatically attempts to connect with alternate TN799DP (CLAN) circuit packs within the original server's configuration, or to a local spare processor (LSP).

H.323 Link Recovery does not diagnose or repair the network failure that caused the link outage, however it:

- Attempts to overcome any network or hardware failure by re-registering the IP endpoint with its original gateway

- Maintains calls in progress during the re-registration attempt

- Continues trying to reconnect if the call ends and the IP endpoint has not yet reconnected to its original gateway

- Attempts connecting to and registering with an alternate gateway if so configured

Hardware errors and alarms added to hard drive trace log

Hardware errors, alarms, and alarm resolutions have been added to the trace log. Software errors and alarms are already tracked on the trace log. The trace log is kept on the hard drive on Linux- and Windows-based servers so they can be retrieved for review and action.

ISDN calls to busy stations treated as ACA short calls

ISDN calls placed to user stations that are busy were labeled as ACA short calls.

> **NOTE:**
> ACA stands for automatic circuit assurance, a feature that tracks calls of unusual duration to help with troubleshooting. For example, a high number of very short calls, or a low number of very long calls, might indicate a problem with the trunk.

ACA short calls generate an notification to the system administrator through a lamp update and a referral call. There is also an ACA log to track events.

A customer might not want to treat short ISDN calls to busy stations as an ACA short call. Therefore, a new field has been added to the *Feature-Related System Parameters* screen. See "Feature-Related System Parameters screen" on page 63 for more information.

IP connections and disconnections

In order to improve the diagnosis of network problems, a log of all IP connections and disconnections is added to the "tracelog" for the Linux-based and Windows-based platforms. The new event types to be logged include IP endpoint registrations, IP endpoint unregistrations, Ethernet interfaces coming into service, and Ethernet interfaces going out of service.

These new events in the tracelog will be tagged as a new log type. This log will be viewable from server commands in Linux and Windows and will also be viewable from the Linux-based maintenance Web pages by selecting the new log type.

IP overload control

This enhancement more effectively manages processor occupancy overload situations. The enhancement applies selected overload mechanisms at a lower occupancy threshold in an effort to avoid more serious symptoms experienced at higher occupancy levels.

The IP overload control enhancement:

- fortifies the system against bursts of registration traffic
- provides a mechanism to alert the far-end to abstain from issuing registrations for some specified period of time
- records the event to maintain a history of potential performance problems
- optimizes the maximum number of simultaneous registrations the server can handle based on the available memory and CPU cycles
- reduces the frequency that a server might go into overload due to network problems

Local feedback for queued ACD calls

A cost saving trend used by many call centers is the movement of agent seats from locations in the US and EU to offshore locations. One detriment to achieving these savings is the increase in trunk costs by redirecting calls to these offshore locations.

When a call is rerouted to an alternate switch, it becomes the responsibility of the destination switch to provide audible feedback to the caller while that call remains in queue at the destination switch waiting for an available agent. Typically, such audible feedback takes the format of music interspersed with recorded announcements.

When the trunks between the sending and receiving switches are IP trunks, bandwidth is utilized when the music and recorded announcement packets are sent from the destination switch to the caller. Because of the continuous nature of music, the bandwidth required to provide this audible feedback to callers in queue is generally greater than that required to support a conversation between a caller and an agent.

Communication Manager allows vector processing to continue at the local sending switch, even after a call has been routed to a queue on an offshore destination switch. Vector processing at the sending switch can then continue to provide audible feedback to the caller while the call is in queue at the destination switch. No packets need be sent over the IP trunk during the queuing phase of the call.

Multiple level precedence and preemption (MLPP)

Multiple level precedence and preemption (MLPP) is an optional group of features that provide users the ability to interface to and operate in a Defense Switched Network (DSN). The DSN is a highly secure and standards-based communication system of the US Government's Department of Defense (DoD).

> **⚠ CAUTION:**
> MLPP is currently designed to meet only the US Government's Defense Switched Network Generic Switching Center Requirements (GSCR) for connection to a DSN by federal, state, or local government agencies. As such, MLPP is not currently designed for use in commercial enterprise environments. Activation of this feature in any other kind of network environment could result in unexpected and unwanted feature operations.

The MLPP features allow users to request priority processing of their calls during critical situations. The MLPP features include:

- Announcements for precedence calling
- Dual homing
- End office access line hunting
- Line load control
- Precedence call waiting
- Precedence calling
- Precedence routing
- Preemption
- Worldwide numbering and dialing plan (WNDP)

Announcements for precedence calling

In certain situations, precedence calls are blocked because of unavailable resources or improper use. When this occurs, recorded announcements are used to identify what went wrong. The announcements used for MLPP include:

- Blocked precedence call
- Unauthorized precedence level attempted
- Service interruption prevented call completion
- Busy, not equipped for preemption or precedence call waiting
- Vacant code

Dual homing

Dual homing allows a user to dial a telephone number and, if the initial route is unavailable, have the call route to its destination over alternate facilities.

End office access line hunting

End office access line hunting automatically hunts for an idle trunk over end office access lines, based on the precedence level of the call.

Line load control

Line load control is a feature that restricts a predefined set of station users from originating calls during a crisis or emergency. Through administration, users are assigned to a line load control level based on their relative importance. When an emergency occurs, the administrator manually enables the feature to restrict calling by users of lower importance. When the emergency is over, the administrator manually disables the feature.

For example, if a situation occurs that threatens national defense, station users in the defense department will not be restricted from originating calls, but stations in other departments, such as the accounting department, will be restricted. When the crisis is over, the system can be returned to normal operation by the administrator.

Precedence call waiting

After a precedence call is routed, the called party may already be busy on another call. Precedence call waiting allows the caller to "camp on" to the called party's line and wait for them to answer the call. The caller hears a special ringback tone and the called party hears a call waiting tone.

Depending on the type of telephone being used, the called party can put the current call on hold and answer the call, or the called party must hang up on their current call to answer the incoming call.

Precedence calling

Precedence calling is the centerpiece of the MLPP features. Precedence calling allows users, on a call-by-call basis, to select a level of priority for each call based on their need and importance (rank). The call receives higher-priority routing, whether the call is local or going around the world.

Users may access five levels of precedence when placing calls:

- Flash Override (the highest precedence level)
- Flash
- Immediate
- Priority
- Routine (the default, and lowest precedence level)

Each station user is administered with a maximum precedence level — the more important or higher in rank the user, the higher the precedence level. Users cannot originate calls at precedence levels higher than their maximum administered level. Non-MLPP calls are treated as routine level precedence calls.

Precedence routing

When precedence calls are destined for other switches in a private network, the precedence routing feature is used to route the calls. The precedence routing feature routes calls based on three main criteria:

- Routing based on the destination number
- Routing based on the precedence level
- Routing based on the time of day

These routing criteria are administrable and can be changed as required. Two related features are dual homing and end office access line hunting.

Preemption

Preemption works with the precedence routing feature to further extend the call routing capabilities of the MLPP features. Preemption, when allowed through administration, can actually tear down an existing lower priority call in order to complete a more important precedence call. Even non-MLPP calls are treated as routine level precedence calls and can be preempted.

When this occurs, the callers on the existing call hear a tone indicating that the call is about to be preempted. The callers have three seconds to end the call before the call is automatically disconnected. After the existing call is disconnected, the new call is placed using preempted facility.

Worldwide numbering and dialing plan (WNDP)

The worldwide numbering and dialing plan (WNDP) feature allows Communication Manager to conform to the standard numbering system established by the Defense Communications Agency (DCA). WNDP defines its own format for the precedence dialing. The capability to operate with this numbering plan must be incorporated into all new switches introduced into the Defense Switched Network (DSN).

Multi-location dial plans

When a customer migrates from a multiple voice server QSIG/DCS network to a single voice server whose gateways are distributed across a data network, it may initially seem as if some dial plan functions are no longer available.

This feature preserves dial plan uniqueness for extensions and attendants that were provided in a multiple QSIG/DCS network, but were lost when customers migrated to a single distributed network. This feature provides dial plan capabilities similar to those provided before the migration, including:

- extension uniqueness
- announcement per location
- local attendant access
- local ARS code administration

A major reason to migrate customers from a multiple QSIG/DCS environment to a single S8700 network is to provide a greater set of features and help reduce costs. Migrating to a single network reduces the number of systems a customer has to maintain. That in turn lowers administration costs — one switch to administer instead of multiple switches, one dial plan to maintain, and so on. With a single distributed network solution, some features no longer work transparently across multiple locations.

For example, in a department store with many locations, each location might have had its own switch with a QSIG/DCS network. That way, the same extension could be used to represent a unique department in all stores. For example, extension 123 might be the luggage department in all stores. If the customer migrates to a single distributed network, this functionality is not available without this feature.

In addition, an S8700 solution does not assure that a call that is routed to an attendant would terminate at the local attendant. Let us use an example of a public school district that previously was networked with a switch at each school. If the school district migrates to an S8700 network, dialing the attendant access code at your school may not route your call to the local attendant.

Instead of having to dial a complete extension, the multi-location dial plan feature allows a user to dial a shorted version of the extension. For example, a customer can continue to dial 4567 instead of having to dial 123-4567. Communication Manager takes the location prefix and adds those digits to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the *Dial Plan Parameters* screen.

Multiple QSIG voice mail hunt groups

Communication Manager provides for ten message center hunt groups to support QSIG integrated messaging. This feature allows customers to spread users in a single Communication Manager system over multiple messaging systems. This allows users to move among Communication Manager systems while retaining their same voice mailbox. Users do not lose voice messages.

This feature also enhances customer usability of Avaya messaging systems in the enterprise by allowing not only for growth, but the ability to migrate end users on a single Communication Manager system.

National ISDN (NI-1 and NI-2) BRI voice endpoint support

Similar to "Multiple level precedence and preemption (MLPP)" on page 23, national ISDN (NI-1 and NI-2) BRI voice endpoint support is a feature that provides users the ability to interface to and operate in a Defense Switched Network (DSN). The DSN is a highly secure and standards-based communication system of the US Government's Department of Defense (DoD).

> **CAUTION:**
> National ISDN (NI-1 and NI-2) BRI voice endpoint support is currently designed to meet only the US Government's Defense Switched Network Generic Switching Center Requirements (GSCR) for connection to a DSN by federal, state, or local government agencies. As such, it is not currently designed for use in commercial enterprise environments. Activation of this feature in any other kind of network environment could result in unexpected and unwanted feature operations.

Support of the NI-1 and NI-2 standards is guided by the requirements in the GSCR. As such, this does not assure support for all the commercially available NI-1 and NI-2 compliant BRI endpoints.

Since the NI-BRI station can be configured in several modes, with its own capability for feature button management, it is of utmost importance that the terminal configuration must match the NI-BRI station support provided by Communication Manager. This implies that:

- Non-initializing NI-BRI terminals (NIT) are *not* supported.

- Valid SPID length is from 3 to 12 digits.

   > **NOTE:**
   > The maximum length according to NI-BRI standards is 20. Communication Manager allows a maximum SPID length of 12, consisting of 10 digits for the USID (SPID on the NI-BRI Administration screen) and 2 digits for the TID (or endpoint ID on the Administration screen).

- Native support is for NI-BRI voice endpoints only. Native support for NI-BRI data endpoints is not offered. NI-BRI data endpoints must be administered as **wcbri** data-module type.

- The NI-BRI station must be programmed as CACH-EKTS mode only. Any other mode is *not* supported.

- The number of call appearances programmed on the terminal must be equal to the number of call appearances administered on the corresponding NI-BRI station form.

- Communication Manager does not support multiple directory numbers per endpoint – there is one station extension per endpoint.

- Conference, transfer, and drop feature buttons must be programmed in order to support features above and beyond the basic call services.

- Integrated voice/data endpoints must be configured to be either a voice-only NI-BRI endpoint (administered as **NI-BRI** station type), or as standalone data-only endpoint (administered as **wcbri** data-module type).

No shutdown when UPS looses battery power

If there's a power failure, an uninterruptible power supply (UPS) takes over to make sure call processing can continue. Prior to Release 2.0, when the UPS warns of low battery power remaining (3 minutes), the server initiated a shutdown.

With Release 2.0, the UPS continues to function until it has no battery power left. No shutdown is initiated. If power can be restored within the time that the UPS battery would have run out of battery power, there is no interruption to call processing. If the UPS runs out of battery power and the server loses power, no damage is done to the server.

Parsing capabilities for the history report

The history report provides details about every data command. You can use parsing options to limit the data returned in this report. The following table identifies the parsing options that are available.

> **NOTE:**
> You can display these options by entering the command **list history**, then clicking **HELP** or pressing **F5**.

Option	Description
date	Specify the month (**MM**) or day (**MM/DD**) for which to display history data.
time	Specify the hour (**HH**) or minute (**HH:MM**) for which to display history data.
login	Specify the login for which you wish to display history data.
action	Specify the command action (the first word of the command string) for which you wish to display history data. You can view the list of available command actions by clicking HELP or pressing **F5** at the command line.
object	Specify the command object for which you wish to display history data.
qualifier	Specify the command qualifier for which you wish to display history data.

To limit the data displayed in the history report, enter the command **list history** followed by a space and the appropriate parser and, if applicable, format. Only the data for the specified parsers will appear in the report.

You can include multiple parsers, but only a single instance of any parser (for example, you may parse for **date**, **time**, and **login**, but not for **date**, **time**, and two different **logins**).

Print option for 4425 terminal

A print option has been extended to 4425 terminal types, as currently exists with 4410 terminal types.

Russian MF shuttle tone level enhancements

A gain of 0 is applied to Russian multi-frequency (MF) shuttle trunks and outgoing Russian multi-frequency ANI. Now, the received and transmitted signal gains administered on the multi-frequency screen are applied to MF shuttle trunks. Also, the transmitted signal gain is applied to outgoing Russian multi-frequency ANI. This is in order to comply with recent Russian requirements.

Service level maximizer

The service level maximizer is a packaged capability that works with active Avaya Expert Agent Selection (EAS). It includes simplified administration to bring the benefits of advanced, predictive call distribution to the widest range of potential call center customers. This is done by meeting a customer's pre-defined service level targets for all skills within a call center.

Service level maximizer maximizes agent utilization to help a call center meet service level targets for all skills. It does require Avaya Expert Agent Selection (EAS) to be active. Unlike EAS, service level maximizer does not require call surplus decisions as to the use of either skill level (ensuring most expert agent agents) or greatest need for those skills.

Agent surplus decisions tied to Expert Agent Distribution also do not apply to those skills using service level maximizer. This means that, with service level maximizer active per skill (hunt group), there is no assurance that the most expert agents answer inbound ACD calls. The reason is skill level and expert agent distribution are not used. Those skills with service level maximizer active also do not consider agent fairness because the concept of least occupied agent is overridden in support of meeting service level targets.

SIP trunks

Session initiation protocol (SIP) trunking functionality allows a Linux server (S8300, S8500, or S8700) to function as a POTS gateway between traditional legacy endpoints (stations and trunks) and SIP endpoints. It also provides SIP-to-SIP routing. In the routing scenario, the server supports call routing similar to what a SIP proxy would provide.

SIP links can be secured using TLS to encrypt signaling, and use Digest Authentication to perform validation. When using TLS, the Media Encryption feature is also available to encrypt audio channels.

SIP trunking functionality:

- Provides access to less expensive local and long distance telephone services, plus other hosted services from SIP service providers
- Provides presence and availability information to members of the enterprise and authorized consumers outside the enterprise, including other enterprises and service providers
- Facilitates SIP-enabled converged communications applications within the enterprise, such as the Seamless Service Experience.

Allowing encryption of signaling and audio channel provides the customer with the option to provide a secure communications infrastructure.

Signaling encryption for SIP trunks

Signaling encryption for SIP trunks protects customer investments by encrypting the voice channel over SIP trunks.

SNMP setting of QoS parameters

In Communication Manager Release 2.0, the native SNMP agent allows setting of QoS parameters through SNMP. The check box on the administrator's *SNMP Agents* page has been changed from **Check to enable busyout/release setting** to **Check to enable busyout/release and qos parameter setting**.

TTY

People with hearing or speech disabilities often rely on a device known as a TTY in order to communicate on telephone systems. The term "TTY" is an abbreviation for Teletypewriter. (The term "TDD" – telecommunication device for the deaf – is also frequently used. The term TTY is generally preferred, however, because many people who use these devices are not deaf.)

TTY devices typically resemble small laptop computers, except that there is a one- or two-line alphanumeric display in place of the computer screen.

Connection to the telephone network is generally through an acoustic coupler into which the user places the telephone's handset, or through an analog RJ-11 tip/ring connections.

Reliable transmission of TTY signals is supported by Communication Manager. This complies with the requirements and guidelines outlined in United States accessibility-related laws. Those laws include:

- Titles II, III, and IV of the Americans with Disabilities Act (ADA) of 1990.

- Sections 251 and 255 of the Telecommunications Act of 1996.

- Section 508 of the Workforce Investment Act of 1998.

Communication Manager's TTY support is currently restricted to TTY devices that use the US standard TTY protocol. That protocol is specified by ANSI/TIA/EIA 825: "A 45.45 Baud FSK modem."

Important characteristics of this standard are:

- TTYs are silent when not transmitting — Unlike fax machines and computer modems, TTYs have no "handshake" procedure at the start of a call, nor do they have a carrier tone during the call. This approach has the advantage of permitting TTY tones, DTMF, and voice to be intermixed on the same call.

  > **NOTE:**
  > A large percentage of people who use TTY devices intermix voice and typed TTY data on the same call. The most common usage is by people who are hard of hearing, but nevertheless able to speak clearly. These people often prefer to receive text on their TTY device and then speak in response. This process is referred to as voice carry over (VCO).

- Operation is "half duplex" — TTY users must take turns transmitting and typically cannot interrupt each other. If two people try to type at the same time, their TTY devices might show no text at all or show text that is unrecognizable. Also, there is no automatic mechanism that lets TTY users know when a character they have correctly typed has been received incorrectly.

- Each TTY character consists of a sequence of seven individual tones — The first tone is always a "start tone" at 1800 Hz. This is followed by a series of five tones, at either 1400 or 1800 Hz, which specify the character. The final tone in the sequence is always a "stop tone" at 1400 Hz. The stop tone is a border that separates this character from the next.

The following types of systems support TTY communication:

- Analog telephones and trunks
- Digital telephones and trunks
- VoIP gateways
- Messaging systems
- Automated attendant systems
- IVR systems
- Wireless systems in which a TTY-compatible coder is used

As long as the user's TTY device supports the following, Communication Manager allows:

- Voice and TTY tones to be intermixed on the same call.
- DTMF and TTY (with or without voice) to be intermixed on the same call. This allows TTY users to access DTMF-based voice mail, auto-attendant, and IVR systems.
- The use of acoustically coupled and "direct connect" (RJ-11) TTY devices.

TTY over analog and digital trunks

Communication Manager supports TTY calls within a gateway or port network between two analog telephones. TTY calls from a gateway or port network over analog trunks or digital trunks is also supported.

TTY over Avaya IP trunks

Communication Manager supports calls over IP trunks, as well as inter-gateway calls (IGC).

For this feature to work, both the sender (near end) and the receiver (far end) of a TTY call must each be connected to Avaya IP trunks. This feature will not work if either telephone is an IP telephone.

Variables for Vectors

This enhancement to the call vectoring feature provides user assignable and changeable variables that can be tested in vectors using the **goto** command. Each variable is assigned a letter, such as A, B, C, and so on.

A particular variable can be used in many vectors. This allows a single user, caller, or CTI application change to effect a logic change in all vectors that the variable appears in. Local and global variables can be defined.

The threshold values and comparators that are applicable to a specific variable depend on the definition for the variable. The variables are defined in a table that is only administered through switch administration.

Each variable can be assigned as a certain type that specifies its characteristics and how it gets set or changed. Types include:

- collected digits assigned

- ASAI user information for the call

- a value [between 0–9] that can be set by dialing a feature access code

- calendar/time type [day-of-year, day-of-week, or time-of-day] – obtained from the switch system clock

- vdn – the active or latest extension of the VDN for the call

See for the new screens associated with this feature.

VoIP resource selection improved

A high percentage of calls between IP endpoints and non-IP endpoints (circuit-switched endpoints) involve more than one port network (PN). A call between an IP endpoint and a non-IP endpoint located in different PNs might require TMD resources that tie up both PNs. This enhancement serves to lessen the overall TDM requirements, and therefore, the required number of PNs involved in the call.

For a call between an IP endpoint and a non-IP endpoint, the IP phone is shuffled to the IP media processor in the non-IP phone's PN if all of the following are true:

- if the IP media processor of the originating IP endpoint is in a different PN as the destination non-IP phone

- if a IP media processor is available in the non-IP phone's PN

- if both IP media processors are in the same region

Capacity changes

Please see the *Avaya Communication Manager System Capacities Table*, 555-233-605, for the entire list of updated capacities. The most up-to-date system capacity information is not listed in Communication Manager documentation. Instead, this information is available online.

To view the System Capacities Table:

1 Go to http://www.avaya.com/support.

2 Type the document number **555-233-605** in the **Search Support** text box, and click **Go**.

3 Locate the latest version of the System Capacities Table document, and then click the format you want to use to download the information.

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).

Release 2.0 hardware additions

Avaya Communication Manager, Release 2.0, includes the following general hardware additions.

2402 DCP telephone

The 2402 digital communications protocol (DCP) telephone is a digital telephone with dual (two) call appearance capabilities. The 2402 DCP phone has both permanently-labeled feature buttons and administrable feature buttons. The 2402 does not support firmware downloads.

The 2402 telephone is aliased as a 6402 DCP telephone, with the following special instructions:

- The shifted dial pad # ("pound") key on the 6402 administration screen must be administered as an autodial button. Also, the dialup number for the voice mail system must be programmed into that autodial button. Since the Messages button on the 2402 telephone has the same button address as the 6402 shifted dial pad # ("pound") key, the 2402 Messages button now accesses the messages server.

- While aliased as a 6402, the 2402 shifted dial pad # ("pound") key cannot have an administered function. That is, the 2402 has 11, not 12, administrable feature buttons.

2420 DCP telephone firmware download

A new interface has been added for the 2420 DCP telephone firmware download that allows a user to initiate, schedule, and status the download. A user can also download multiple 2420 stations simultaneously.

4602SW IP telephone

The 4602SW telephones has improved functionality, including enhanced security and improved VLAN operation. The 4602SW telephone has a built-in Ethernet switch.

4602SIP telephone

A version of the 4602 telephone supports the session initiation protocol (SIP) for basic call control. Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network.

4610SW IP telephone

The 4610SW IP telephone has a built-in Ethernet switch. The 4610SW IP telephone provides advanced feature functionality with an intuitive and innovative user interface.

- a 168-by-80 pixel 4-gray scale display

- four softkeys

- six dynamically labeled call appearance/feature buttons

- four unique fixed feature buttons

The 4610SW IP telephone is aliased as a 4620 IP telephone.

4690 IP conference room speaker telephone

The 4690 IP SoundStation conference room speaker phone supports Avaya's H.323-based IP single-connect protocol, including registration and proprietary DCP/CCMS messages. It supports multiple call appearances and administrable features. The 4690 IP conference station is aliased as a 4620 telephone so that it can receive downloaded button labels.

Avaya IP Softphone

Avaya IP Softphone extends the level of Communication Manager services. This feature turns a PC or a laptop into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PCs.

> **NOTE:**
> R1 and R2 IP Softphone and IP Agent, which use a dual connect (two extensions) architecture, are no longer supported. R3 and R4 IP Softphone and IP Agent, which use a single connect (one extension) architecture, continue to be supported. This applies to the RoadWarrior configuration and the Native H.323 configuration for the IP Softphone.

The R5 release of the IP Softphone supports a number of enhanced features, including the following:

- Improved endpoint connection recovery algorithm

- AES media encryption (see "AES encryption algorithm for bearer channels" on page 15)

- Instant Messaging (see "Converged communications server" on page 19)

- Unicode support (see "Unicode support" on page 38)

- Softphone and Telephone Shared Control (see "IP Softphone and telephone — Shared Control mode" on page 35)

The IP Softphone provides a graphical user interface with enhanced capabilities when used with certain models of DCP telephones. Communication Manager supports a mode of H.323 registration that allows an IP Softphone to register for the same extension as a DCP telephone without disabling the telephone. It also allows the IP Softphone to send button-push messages and receive display and call progress messages in parallel with the telephone. In this mode, the Softphone does not terminate any audio.

IP Softphone and telephone — Shared Control mode

IP Softphone and telephone, Shared Control mode, enables users to have a telephone endpoint and an IP Softphone in service simultaneously on the same extension number. IP Softphone and an IP telephone can be integrated so that the IP softphone can control the IP telephone on a person's desk, and vice versa. This allows the power of the PC desktop (LDAP directories, TAPI PIMs/Contact Managers, etc.) to be used in conjunction with a desktop IP telephone.

An IP softphone can register to an extension number that is already assigned to an in-service telephone endpoint. From that point on, user actions carried out by either endpoint apply to calls to or from the extension. Only the telephone endpoint carries audio for the extension.

G350 Media Gateway

The G350 Media Gateway is a low-cost, modular device targeting the small branch office of a large enterprise. The G350 Media Gateway provides customers with an affordable converged solution for the remote/branch offices between 15–40 stations. Additionally, branch office customers do not have to sacrifice telephony features because this solution extends the power of Communication Manager in a survivable package.

Some of the highlights of the G350 Media Gateway include:

- VoIP media gateway with both trunk and line functions. A Communication Manager server acts as the call controller for the G350 Media Gateway. An S8300 ICC processor can be used in a G350 Media Gateway.
- The G350 Media Gateway can be controlled by an external S8700 or S8500 Media Server, or an internal S8300 Media Server.
- WAN (data) connectivity and routing for both H.248 connections to a call controller and for general inter/intranet connectivity, and LAN Switching, providing power over ethernet for IP phones and wireless access points.

The G350 supports a new high density form factor, and provides a set of voice and data infrastructure services. The G350 Media Module utilizes the G700 Media Module form factor media modules (MM).

The following G700 form factor media modules have been developed:

- MM714 4FXS + 4FXO analog media module
- MM722 2-port BRI media module
- MM340 E1/T1 WAN routing media module
- MM342 V.35/X.21 WAN routing media module

The following boards conform to the new high density form factor.

- MM312 24-port DCP expansion module
- MM314 24-port PoE expansion module

G650 Media Gateway

The G650 Media Gateway is an enhanced rack mount cabinet that provides customers with an expandable gateway for traditional and IP configurations, a data form factor, and scalability.

The G650 also includes:

- Support for up to 14 universal TN- circuit pack slots in a common carrier
- Support for up to 5 G650s (70 total TN- circuit pack slots) in a port network
- A new 655A power supply design that:
    — operates from either AC or DC input power
    — can be used in either a stand alone or redundant configuration in a single carrier
    — provides load sharing and improved system availability when used in the redundant mode
- Enhanced environmental status and control, including voltage, current, temperature, and fan speed with a new serial bus that interconnects all the power supplies with the TN2312BP
- Simplified carrier address setting at installation

The redundant power supplies, coupled with the enhanced environmental reporting, improve the reliability, availability, and maintainability of the carrier over the existing G600 design.

655A power supply

The 655A power supply provides power for the G650 Media Gateway. Each power supply can communicate environmental information on the new serial bus when polled by the TN2312BP. A G650 can be equipped with one or two 655A power supplies. When two power supplies are used, they work in a current sharing mode. Either is capable of supplying power for a fully equipped G650.

The 655A power supply is capable of operating on either AC or DC input power. If both types of power are supplied, the 655A uses AC power first, but switches to DC power if AC power is unavailable.

The 655A is capable of providing North American (20 Hz) or European (25 Hz) ringing through a selection switch on the power supply. A third ringing selection can be made to provide no ringing when an external ringing source is provided – for example, TN2202 French (50 Hz) ringing. When two 655A power supplies are equipped in a single G650, only one provides ringing. The other 655A is in a standby ready mode to provide ringing if needed.

TN2312BP IPSI functionality

The G650 Media Gateway includes an enhanced Internet Protocol Server Interface (IPSI) circuit pack – the TN2312BP IPSI circuit pack. The TN2312BP circuit pack provides additional maintenance capabilities and features. The TN2312BP is backward compatible with the TN2312AP, but provides greater capabilities when used in the G650. The TN2312BP in an A carrier acts as a bus master for the G650 or G650 stack.

The TN2312BP IPSI functionality consists of:
- the existing IPSI (TN2312AP circuit pack) functions
- G650 cabinet maintenance

Features requiring the new TN2312BP I/O adapter (700263502):
- Customer provided alarm device (CPAD) control
- Emergency transfer panel control and L.E.D. indicator

- Major/minor alarm inputs from a UPS or other equipment
- IPSI duplication for IP-connected systems

Upgrades from the TN2312AP IPSI to the TN2312BP IPSI cannot be done with a firmware download only because new hardware features are included on the new circuit pack.

TN2312BP IPSI support of G650 cabinet maintenance includes:

- power/fan temperature monitoring
- an enhanced bus interface from the TN2312BP IPSI to the 655A power supplies in the G650 carrier
- a lead from the TN2312BP IPSI to the 655A power supply that provides environmental monitoring of the power supply and fan assembly

External readable CD ROM for S8300 Media Server

The S8300 Media Server supports an external readable CD ROM drive through one of the two USB ports on the S8300. The CD ROM is used for system software upgrades.

S8500 Media Server

The Avaya S8500 Media Server is a rack mounted telephony server, running the Red Hat Linux 8.0 Operating System, and featuring Avaya Communication Manager. The S8500 is capable of supporting both Internet Protocol (IP) and traditional endpoints enabling new technology, and the ease of migration from legacy Avaya systems. The S8500 Media Server is a perfect solution for mid-sized customers, with growth of up to 3200 ports.

- Migration path for DEFINITY servers CSI, SI, and S8100
- Supports distributed environment in a smaller scale
- Going forward platform to support IP telephony

Disk survivability

See .

Linux 8.0 support

New purchases of S8300, S8500, and S8700 Media Servers operate with the Red Hat Linux 8.0 operating system. The S8500 and S8700 Media Servers are shipped with blank hard drives. The operating system and Release 2.0 of Communication Manager are shipped separately on a CD and installed on the media servers on site. The S8300 Media Server is shipped with the operating system and Release 2.0 of Communication Manager on the hard drives.

Upgrading Communication Manager to Release 2.0 on an existing S8300 Media Server or an S8700 Media Server requires upgrading the operating system to Red Hat Linux 8.0. The upgrade process requires remastering (reformatting) the hard drives, then loading the new software and operating system from a CD-ROM.

Unicode support

Communication Manager supports the display of non-English static and dynamic display text on Unicode-enabled endpoints. Non-English display information is entered into a Avaya Integrated Management application. Communication Manager processes, stores, and transmits the non-English text to endpoints that support Unicode displays. Unicode support is applicable to some firmware versions of the 4620 telephone, and the Softphone version of the 4620. It is not applicable to the 4690 telephone.

Unicode support provides the capability of supporting international and multi-national communications solutions. End-users are provided with a communications interface (delivered by an IP telephone or IP Softphone) in their own native language. This feature supports the Simplified Chinese, Japanese, and Korean (CJK) character sets.

3   New and changed screens

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

Release 2.0 new screens

Avaya Communication Manager, Release 2.0, includes the following new screens.

Change Station Extension screen

The Multi-Location Dial Plan feature introduces a new screen, ***Change Station Extension*** (Figure 1). Use the **change extension-station** *n* command, where *n* is the station you want to change. This screen allows an administrator to change a station extension on the switch without removing the station then adding it back.

> **NOTE:**
> You cannot use the **change extension-station** command to change the extension of a station if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- You cannot change the extension of station A using the **change extension-station** command unless you first change station B to assign a different emergency location extension.

- You can change the extension of station B. If you do, the ***Change Station Extension*** screen displays station A's extension in the **Emergency Location Ext.** field under the **From Extension** header.

Figure 1: Change Station Extension screen

```
change extension-station 1234567                                Page 1 of 1

                       CHANGE STATION EXTENSION

       Station Name: xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx    Port: xxxxxx

                          FROM EXTENSION                TO EXTENSION
                          --------------                ------------
                 Station: xxx-xxxx                       xxx-xxxx
            Message Lamp: xxx-xxxx                       xxx-xxxx
    Emergency Location Ext.: xxx-xxxx                    xxx-xxxx
IP Parameter Emergency Location: xxx-xxxx                See IP-Network Map Form
```

Function

When the screen is filled out and submitted, all administration that was associated with the current extension will now be associated with the new extension. Any administration references of the extension being changed, such as references used in a vector, coverage, etc., will now reference the new extension. Once the extension has been changed, all references to the previous extension will be removed from the switch.

If an extension is changed that is also administered on an adjunct (such as voice mail or an ASAI link), the extension on the adjunct must also be changed to ensure proper functionality.

Exceptions

Administration that is associated with most buttons on the current extension (for example, call forwarding digits and abbreviated dial) are not changed with the new extension through the **change extension-station** *n* command.

> **NOTE:**
> A forwarded extension administered as a button will not be handled by the **change extension-station** *n* command, The reason is the extension for the call forwarded button are stored as digits rather than as a UID. It is recommended that the administrator use the **list usage** command prior to changing any extensions.

Updating the **ISDN BRI SPID** field for BRI stations is omitted from this command. Changing BRI stations is a 2-step process — the change must be made both on the switch and the telephone. Updating this field remains a manual process.

Audits

If an attempt is made to change an extension that is administered on the same switch as an **emergency location extension** on the *Station* screen, or as an **emergency location extension** on the *IP Address Mapping* screen, the following warning message appears:

    Extension exists as Emergency Location. Continue?

Click **yes** to continue to process the change. Click **no** to stop the process.

The **change extension-station *n*** command will be denied under the following conditions:

— If the extension being changed is active on a call

— If the extension is being accessed by administration

- If an attempt is made to change an extension that is administered on the same switch as a **media complex extension** on the *Station* screen, the extension cannot be changed to a 6-digit or 7-digit number. The reason is the **media complex extension** field on the *Station* screen does not support 6-digit or 7-digit numbers.

  For example, if station 50002 is an H.323 station and is administered on station 50012 as the media complex, the **change extension-station *n*** command does not allow you to change station 50002 to 7050002, since a 7-digit media complex number is not supported. The command would permit changing 50002 to 52222, since the resulting extension is a 5-digit extension.

Field descriptions

Station Name   This field is read only, and displays the name of the existing extension (the extension that was typed in the **change extension-station** *n* command).

Port   This field is read only, and displays the port of the existing extension.

For the fields in the **From Extension** column (all fields are read only):

Station   This field displays the current extension that is being changed (the extension that was typed in the **change extension-station** *n* command).

Message Lamp   This field displays the **Message Lamp Extension** associated with the current extension.

Emergency Location Ext.   This field displays the **Emergency Location Ext**. from the *Station* screen associated with the current extension.

IP Parameter Emergency Location   This field displays the **Emergency Location Extension** from the *IP Address Mapping* screen associated with the current extension.

For the fields in the **To Extension** column:

Station   Type a new extension that you want the current extension changed to.

Valid entries	Usage
0 to **9**	Type up to seven numbers that make up a valid extension number for your dial plan.

Message Lamp   Type a new extension for the Message Lamp Extension.

Valid entries	Usage
0 to **9**	Type up to seven numbers that make up a valid extension number for your dial plan.

Emergency Location Ext.   Type a new extension for the **emergency location ext**. field that will appear on the *Station* screen.

Valid entries	Usage
0 to **9**	Type up to seven numbers that make up a valid extension number for your dial plan.

IP Parameter Emergency Location   The message `See Ip-Network Map Form` is displayed. The administrator can only change this field on the *IP Address Mapping* screen (using the **change ip-network-map** command).

CLAN (TN799x) Socket Usage screen

The CLAN socket load balancing feature introduces a new screen, *CLAN (TN799x) Socket Usage* (Figure 2). Use the **status clan-usage** command. This new screen lists information about the socket usage for all TN799x circuit packs administered on a system (up to 64). Page 2 of the screen has the same fields as page 1 – up to 32 C-LANs can be listed on each page.

Figure 2: CLAN (TN799x) Socket Usage screen

```
status clan-usage                                             Page 1 of 2

                       CLAN (TN799x) SOCKET USAGE

TN799        Socket   Net     TN799        Socket   Net    TN799        Socket   Net
Loc     Sfx Usage     Rgn     Loc     Sfx Usage     Rgn    Loc     Sfx Usage     Rgn
02A14   C   245/400   20      02C09   D   125/250   20     03C06   D   400/400   125
```

Field descriptions

TN799 Loc   This is the cabinet/carrier/slot location of the TN799x circuit pack.

Sfx   This is the suffix of the TN799 circuit pack (C or D).

Socket Usage   The first number shows the number of sockets in use on the specified TN799x circuit pack when the command was entered. The second number is the value administered in the **Number of CLAN Sockets Before Warning** field on the *add/change IP Interfaces* form. The socket usage number does not include sockets used by adjuncts.

Net Rgn   This is the network region to which the TN799x circuit pack is assigned on the *add ip-interface* screen.

The remaining fields are described in other documentation.

Events Report screen

The **reset ip-stations** command displays a new screen, *Events Report* ().

Figure 3: Events Report screen

```
display events

                         EVENTS REPORT

EVENT EVENT                    Event      Event     First        Last        Evnt
TYPE  DESCRIPTION              Data 1     Data 2    Occur        Occur       Cnt
2033 IP FURQ-Demand Unregister 0          0         03/15/09:51 03/15/09:51  1
```

Extensions to Call which Activate Features By Name screen

The extension to cellular feature introduces a new screen, *Extensions to Call which Activate Features By Name* ([Figure 4](#)). The *Extensions to Call which Activate Features By Name* screen is where you map a dialed extension to a feature within Communication Manager. These are called feature name extensions (FNE). These extensions are paired with feature access codes (FAC). When the extension is called, the feature access code is activated.

> **NOTE:**
> The Feature Access Codes are administered on the *Feature Access Code (FAC)* screen.

To map a dialed extension to a feature within Communication Manager:

1   Type **change off-pbx-telephone feature-name-extensions** and press **Enter**.

   The *Extensions to Call which Activate Features By Name* screen appears.

Figure 4: Extensions to Call which Activate Features By Name screen, page 1

```
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
            Call Forward All: _____          Last Number Dialed: _____
 Call Forward Busy/No Answer: _____         Malicious Call Trace: _____
         Call Forward Cancel: _____  Malicious Call Trace Cancel: _____
                   Call Park: _____               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: _____
          Directed Call Pick-Up: _____
        Drop Last Added Party: _____
   Exclusion (Toggle On/Off): _____
        Held Appearance Select: _____
```

Valid parameters

Action	Object	Qualifier
change	**off-pbx-telephone feature-name-extensions**	

Field descriptions

Extension   Each **Extension** field is an extension that matches your dial plan. A user would dial the extension from their cell phone, thereby activating a feature access code (FAC) administered for that feature.

Valid entries	Usage
blank **0–9**	Default = blank. Type an extension number, up to eight digits, of the Communication Manager feature you want to be able to access from the cell phone.

IP Interfaces screen

The CLAN socket load balancing feature introduces a new screen, *IP Interfaces*. Use the **list ip-interface** *<all/clan>* command (the default is *all*).

- Figure 5 shows the screen when the **list ip-interfaces** *clan* command is used.
- Figure 6 shows the screen when the **list ip-interfaces** *all* command is used with an S8500 or S8700 Media Server.
- Figure 7 shows the screen when the **list ip-interfaces** *all* command is used with an S8300 Media Server.

Figure 5: IP Interfaces screen

```
list ip-interface clan                                             Page 1 of x

                              IP INTERFACES

                                                           Num
                                                           Skts Net
On      Slot   Code  sfx Node Name       Subnet Mask    Gateway Address Warn Reg VLAN
y       01B04 TN799 D    clan1xxxxxxxxx 255.255.0 .0    172. 22. 22.254 400  1   1
```

Figure 6: IP Interfaces screen

```
list ip-interface all                                              Page 1 of x

                              IP INTERFACES

                                                                   Net
On Type   Slot  Code  sfx Node Name       Subnet Mask   Gateway Address Reg VLAN
y  C-LAN  01B04 TN799 D   clan1xxxxxxxx 255.255.0 .0 172. 22. 22.254 1   1
y  MEDPRO 01A05 TN2302 AP medpro1xxxxxx 255.255.0 .0 172. 22. 22.254 1   1
y  VAL    01B06 TN2501 AP val1xxxxxxxxx 255.255.0 .0 172. 22. 22.254 1   1
```

Figure 7: IP Interfaces screen

```
list ip-interface all                                              Page 1 of x

                              IP INTERFACES

                                                                   Net
On Type   Slot  Code  sfx Node Name       Subnet Mask   Gateway Address Reg VLAN
y  PROCR                  135.6.41.121 255.255.0 .0 172. 22. 22.254 1   1
```

Field descriptions

Num Skts Warn   The value administered in the **Number of CLAN Sockets Before Warning** field on the *add/change IP Interfaces* form. This field is displayed only when the *clan* option is entered. When the *all* option is entered, the **Type** field is displayed and **Num Skts Warn** is not displayed.

Type   For the S8700 and S8500 Media Servers, the possible values are **CLAN**, **MEDPRO**, and **VAL**. For the S8300 Media Server, the only possible value is **PROCR**.

Node Name   For the S8700 and S8500 Media Servers, the **Node Name** field displays the node name of the IP interface as administered on the *Node Names* form. For the S8300 Media Server, the **Node Name** field displays the IP address of the S8300, which is associated with the reserved node name procr on the *Node Names* form.

The remaining fields are described in other documentation for the *IP Interfaces* screen.

IP Interface Status screen

The camp-on/busy-out feature introduces a new screen, *IP Interface Status* (Figure 8). Use a new command, **status media-processor <cabinet/carrier/slot>**, to display this screen This new screen shows the busyout status of the specified MedPro or IPMedPro media processor board.

Figure 8: IP Interface Status

```
status media-processor 1c18                                     Page 1 of 1

                              IP INTERFACE STATUS

                          Link status: connected
                            Node Name: IPMEDPRO1
                    Source IP Address: 192.168.22.11
                          Subnet Mask: 255.255.255.0
                    Broadcast Address: 192.168.22.255
                     Physical Address: XX:XX:XX:XX:XX:XX
                             Enabled? yes

                             DSP Channel Status
      DSP1: in-service/idle                       DSP5: in-service/idle
      DSP2: in-service/active, 1 calls            DSP6: in-service,/active, 4calls
      DSP3: in-service/active, 3 calls            DSP7: in-service,/idle
      DSP4: in-service/active, 2 calls            DSP8: in-service,/idle
```

Valid parameters

Action	Object	Qualifier
status	**media-processor**	*<cabinet/carrier/slot>*

Field descriptions

> **NOTE:**
> All fields are display-only.

Link Status   **Link Status** indicates whether an Ethernet connection is being detected by the media processor. Values are **connected** or **disconnected**.

Node Name   This is the node name of the media processor administered with the **add/change ip-interfaces** command.

Source IP Address   This is the IP address of the media processor associated with the node name administered on the *IP Node Names* screen.

Subnet Mask   This is the subnet mask value administered with the **add/change ip-interfaces** command.

Broadcast Address   This is the broadcast address value administered with the **add/change ip-interfaces** command.

Physical Address   This is the physical IP address queried from the media processor.

Enabled?   This designates whether the IP interface is enabled or not. Values are **Yes** or **No**.

DSP1–DSP8F   This is the service state of the DSPs when the **status media-processor** command was issued. Values are:

- in-service/idle
- in-service/active, *x* calls (where *x* indicates the number of calls active on the media processor)
- craft busied out
- PENDING busy out, *x* calls (marked for busy-out but not performed)
- out-of-service (for another specified reason; for example, "no link")

Stations with Off-PBX Telephone Integration screen, page 1

The Extension to Cellular feature introduces a new screen, *Stations with Off-PBX Telephone Integration* (). The *Stations with Off-PBX Telephone Integration* screen is where you map a user's office (host) phone to an Extension to Cellular phone (for example, a cell phone). A person's outside phone number, like a cell phone, is mapped to an office phone in Communication Manager. The station may be a standard office number (presumably the user's primary extension), or may be an AWOH (administration without hardware) station.

> **NOTE:**
> When Extension to Cellular is administered, the initial state of the cell phone is disabled. You must enable the Extension to Cellular in order to receive calls on the cell phone from the Avaya server running Communication Manager.

To map a user's office (host) phone to your cell phone:

1   Type **add off-pbx-telephone station-mapping** and press **Enter**.

The *Stations with Off-PBX Telephone Integration* screen appears.

Figure 9: Stations with Off-PBX Telephone Integration screen, page 1

```
add off-pbx-telephone station-mapping                          Page 1 of 2

                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station      Application  Dial   Phone Number          Trunk      Configuration
Extension                 Prefix                        Selection  Set
43001        EC500        ____ - 9736831204            ars        1
43005        EC500        ____ - 6093451298            ars        2
43007        OPS          ____ - 12345                ars        3
43009        OPS          ____ - 67890                aar        4
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
_____      _____      ____ - _____         ____       __
```

Valid parameters

Action	Object	Qualifier
add	**off-pbx-telephone station-mapping**	
change	**off-pbx-telephone station-mapping**	*<station extension>*
display	**off-pbx-telephone station-mapping**	*<station extension>*
list	**off-pbx-telephone station-mapping**	*<variable>*

- The **add off-pbx-telephone station-mapping** command displays a blank screen of two pages. It allows for the addition of sixteen station extension to external phone number associations. For each **Station Extension**, the remaining fields on the first page (except for the **Dial Prefix** field) must be filled in. The **Station Extension** fields on the second page are defaulted based upon the values entered for the first page.

- The **change off-pbx-telephone station-mapping** *<station extension>* command displays a screen of two pages. It allows you to change one station extension to external phone number association. The first line contains the information for the station extension you entered as the command variable. Additional entries may be added after the changed one. The table is filled in and validated as described for the add command.

  If the specified station is a valid type, but has not been previously administered for Extension to Cellular, then the screen is blank except for the first **Station Extension** field.

- The **display off-pbx-telephone station-mapping** *<station extension>* command displays a screen of two pages. It lists up to sixteen entries, starting with the station extension you entered as the command variable. If this extension is not administered for Extension to Cellular, the display starts with the first administered Extension to Cellular extension following it.

- The extension may be omitted, in which case the display starts with the first extension administered for Extension to Cellular.

- The **list off-pbx-telephone station-mapping** *<variable>* command shows, on a single line, information about the association between a station and external phone number. The command variable specifies the phone number or numbers of interest. The command variable may be:

    — a complete phone number

    — a partial phone number followed by an asterisk (acting as a "wildcard" character)

    — blank

Field descriptions

Station Extension   The **Station Extension** field is an administered extension in your dial plan for the "host" or office phone.

Valid entries	Usage
a valid number in your dial plan	Default = blank. Type an extension number of the "host" office phone up to eight digits.

Application   The **Application** field is where you indicate what type of application is associated with this station.

Valid entries	Usage
blank	Default = blank
EC500	**EC500** = cell phone
OPS	**OPS** = other outside phone type (for example, the 4602 SIP-enabled phone)

Dial Prefix   The **Dial Prefix** field are any digits that will be prepended to the **Phone Number** field before dialing the outside phone.

Valid entries	Usage
blank **0–9, *, #**	Default = blank. Type up to four digits, including "*" or "#". If included, "*" or "#" must be in the first digit position.

Phone Number   The **Phone Number** field is the phone number of the outside phone.

Valid entries	Usage
0–9	Default = blank. Type up to fifteen digits.

Trunk Selection   The **Trunk Selection** field is where you define which outgoing trunk group you choose to use for outgoing calls.

Valid entries	Usage
ars aar	Default = blank.

Configuration Set   The **Configuration Set** field is used to administer the Configuration Set number that contains the desired call treatment options for this Extension to Cellular station. There are ten Configuration Sets.

Valid entries	Usage
1–10	Default = blank. Type the number of the Configuration Set.

Stations with Off-PBX Telephone Integration screen, page 2

The second page of the *Stations with Off-PBX Telephone Integration* screen (Figure 10) continues the administration of the phone mapping. The information you entered in the **Station Extension** field on the first page appears as read only information on the second page.

Figure 10: 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
43005          2          both         all          both
43007          2          both         all          both
43009          2          both         all          both
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
               __         _____   _____     _____
```

Call Limit   The **Call Limit** field sets the maximum number of Extension to Cellular calls that may be simultaneously active at the station.

Valid entries	Usage
blank	Default = **2**.
1–10	

Mapping Mode   The **Mapping Mode** field contains the mode of operation in which the cell phone operates. See "Mapping Modes" on page 50 for information on the possible entries.

Valid entries	Usage
termination origination both none	Default = **both** (if the **Phone Number** field has not been previously administered for another extension with a **Mapping Mode** of **both** or **origination**. Otherwise, the default = **termination**).

Calls Allowed   The **Calls Allowed** field is used to identify the call filter type for an Extension to Cellular station. See "Calls Allowed" on page 51 for information on the possible entries.

Valid entries	Usage
internal external all none	Default = **all**.

Bridged Calls   The **Bridged Calls** field is used to determine if bridged call appearances, that may be administered on this *Stations with Off-PBX Telephone Integration* screen, should also be extended to the Extension to Cellular phone. See "Mapping Modes" on page 50 for information on the possible entries.

Valid entries	Usage
termination origination both none	Default = **both.**

Mapping Modes

There are four modes in which an Extension to Cellular phone can be mapped to the user's main office phone.

- termination
- origination
- both
- none

These modes are used to control the degree of integration between their cell phone and main office number. The modes are valid for Extension to Cellular calls only – calls to the user's main office number when Extension to Cellular is enabled, and calls from the cell phone into the user's switch when Extension to Cellular is enabled.

Mapping modes are administered on the second page of the *Stations with Off-PBX Telephone Integration* screen (Figure 10).

Calls terminating to a cell phone

The termination mode is enabled by setting the **Mapping Mode** field on the second page of the *Stations with Off-PBX Telephone Integration* screen to **termination**. In termination mode, the cell phone may only be used to terminate (receive) calls from its associated host phone. The cell phone may not be used to originate calls from its associated host phone.

Calls originating from the cell phone are completely independent of Extension to Cellular, and behave exactly as before enabling Extension to Cellular.

Calls originating from a cell phone

The origination mode is enabled by setting the **Mapping Mode** field on the second page of the *Stations with Off-PBX Telephone Integration* screen to **origination**. In origination mode, the cell phone may only be used to originate calls from its associated host phone. The cell phone may not be used to terminate (receive) calls from its associated host phone.

Calls both to and from a cell phone

This mode is enabled by setting the **Mapping Mode** field on the second page of the *Stations with Off-PBX Telephone Integration* screen to **both**. In this mode, the cell phone may be used to originate calls and to terminate calls from its associated host phone.

No calls to or from a cell phone

This mode is enabled by setting the **Mapping Mode** field on the second page of the *Stations with Off-PBX Telephone Integration* screen to **none**. In this mode, In this mode, the cell phone may not be used to originate calls or to terminate calls from its associated host phone.

Calls Allowed

There are four values to define what kind of calls can be associated with the cell phone.

- internal
- external
- all
- none

These entries are used to filter the calls to the cell phone.

The **Calls Allowed** field is administered on the second page of the *Stations with Off-PBX Telephone Integration* screen (Figure 10).

Internal

When **internal** is chosen as the call filter type, the cell phone receives only internal calls. External calls are not delivered to the cell phone.

- When **all** is chosen as the call filter type, the cell phone receives both internal and external calls.
- When **none** is chosen as the call filter type, the cell phone does not receive Extension to Cellular calls.

> **NOTE:**
> Regular calls placed to the cell phone number are still received.

External

When **external** is chosen as the call filter type, the cell phone receives only external calls. Internal calls are not delivered to the cell phone.

All

When **all** is chosen as the call filter type, the cell phone receives both internal and external calls.

None

When **none** is chosen as the call filter type, the cell phone does not receive Extension to Cellular calls.

> **NOTE:**
> Regular calls placed to the cell phone number are still received.

Variables for Vectors screen

The Variables for Vectors feature introduces a new screen, *Variables for Vectors* (Figure 11). Use the **change variables** command. On these two screens, you can assign variable definitions for vectors. For this feature to be activated, the **Vectoring (Variables)** field must be set to **y** on the *Call Center Optional Features* screen (use the **display system-parameters customer-options** command – see page 59). For more information on the Variables for Vectors feature, see "Variables for Vectors" on page 31.

Figure 11: Variables for Vectors screen

```
change variables                                                  Page 1 of 2

                            VARIABLES FOR VECTORS

Var Description                    Type    Scope Length Start Assignment       VAC
A   _____ _____ _      __    __   _____ ___
B   _____ _____ _      __    __   _____ ___
C   _____ _____ _      __    __   _____ ___
D   _____ _____ _      __    __   _____ ___
E   _____ _____ _      __    __   _____ ___
F   _____ _____ _      __    __   _____ ___
G   _____ _____ _      __    __   _____ ___
H   _____ _____ _      __    __   _____ ___
I   _____ _____ _      __    __   _____ ___
J   _____ _____ _      __    __   _____ ___
K   _____ _____ _      __    __   _____ ___
L   _____ _____ _      __    __   _____ ___
M   _____ _____ _      __    __   _____ ___
N   _____ _____ _      __    __   _____ ___
O   _____ _____ _      __    __   _____ ___
```

Var   The **Var** field lists the variable letters **A**–**Z**, one for each row.

For the remaining fields, the following applies:

Table 1: Field values for the Variables for Vectors screen *1 of 2*

Type	Scope – **L** (for local) or **G** (for global)	Length – the maximum number of digits to use	Start – the character position of the digit string to start with	Assignment – a number to be preassigned to the variable	VAC (variable access code) – the vector variable (VV) feature access code to use for changing the value
collect	Type either **L** or **G**. This is a required field.	Type a number between 1–16 to be used from the up to 16-digit collected digit string. This is a required field. The default is **16**.	Type a number between 1–16 for the start position. This is a required field. The default is **1**.	For **Type L**, this does not apply. For **Type G**, type a number between 0–9999999999999999 (sixteen digits long). This is an optional field. The default is blank if null, or displays the current value when not null.	Does not apply.
asaiuui	Entry of **L** is pre-entered.	Type a number between 1–16 to be used from the up to 96-digit ASAI user information digit string. This is a required field. The default is **16**.	Type a number between 1–96 for the start position. This is a required field. The default is **1**.	Does not apply.	Does not apply.
value	Entry of **G** is pre-entered.	Entry of **1** is pre-entered.	Does not apply.	Type a number between 0–9. This is an optional field. The default is blank if null, or displays the current value when not null.	Type a value of **VVx**, where **x** is a number between 1–9. This is an optional field. The default is blank if null or if **VVx** is not assigned. (To assign **VVx** values, see page 63.)
tod	Entry of **G** is pre-entered.	Does not apply.	Does not apply.	Displays current value from switch clock.	Does not apply.

1 of 2

Table 1: Field values for the Variables for Vectors screen *2 of 2*

Type	Scope – **L** (for local) or **G** (for global)	Length – the maximum number of digits to use	Start – the character position of the digit string to start with	Assignment – a number to be preassigned to the variable	VAC (variable access code) – the vector variable (VV) feature access code to use for changing the value
doy	Entry of **G** is pre-entered.	Does not apply.	Does not apply.	Displays current value from switch clock.	Does not apply.
dow	Entry of **G** is pre-entered.	Does not apply.	Does not apply.	Displays current value from switch clock.	Does not apply.
vdn	Entry of **L** is pre-entered.	Does not apply.	Does not apply.	Entry of **active** or **latest** is required. The default is blank.	Does not apply.

2 of 2

Release 2.0 changed screens

Avaya Communication Manager, Release 2.0, includes the following changed screens.

Fields moved or changed

A number of fields have been moved from one screen to a different screen. The purpose was to make a more logical grouping and for future features. In addition, other fields have been renamed – again, for a more logical explanation. The function of these fields, however, remains the same.

Table 2 shows a list of the fields that have moved or changed.

> **NOTE:**
> The screen "name" is actually the command you type to access the screen. For example, the **location-parameters** screen means that you should type the command **change location-parameters** to access the proper screen.

Table 2: Fields that have moved or changed names *1 of 2*

Field name	Changed to	From screen	Moved to screen
Analog Ringing Cadence	——	system-parameters country-options	location-parameters
Analog Line Transmission	——	system-parameters country-options	location-parameters
Companding Mode	——	system-parameters country-options	location-parameters
Digital Loss Plan	2 Party Loss Plan	system-parameters country-options	location-parameters
Digital Tone Plan	Tone Loss Plan	system-parameters country-options	location-parameters
End-to-End total loss (db) in a n-party conference	——	system-parameters country-options	location-parameters
Customize (associated with the End-to-End total loss (db) in a n-party conference field)	——	(new)	location-parameters
Base Tone Generator Set	——	system-parameters country-options	tone-generation
440Hz PBX-dial Tone	——	system-parameters country-options	tone-generation
440Hz secondary-dial Tone	——	system-parameters country-options	tone-generation
Tone Name	——	system-parameters country-options	tone-generation
Cadence Step	——	system-parameters country-options	tone-generation
Tone (Frequency/Level)	——	system-parameters country-options	tone-generation
Flashhook Interval	——	system-parameters features	location-parameters
Disconnect Timing	——	system-parameters features	location-parameters
RECALL TIMING Upper Bound	——	system-parameters features	location-parameters
RECALL TIMING Lower Bound	——	system-parameters features	location-parameters
			1 of 2

Table 2: Fields that have moved or changed names *2 of 2*

Field name	Changed to	From screen	Moved to screen
Forward Disconnect Timer	——	**system-parameters features**	**location-parameters**
ani for pbx	**Default ANI**	**system-parameters multifrequency-signaling**	——
(all fields except the following three fields)	——	**system-parameters multifrequency-signaling**	**multifrequency-signaling**
MF Interdigit Timer	——	**system-parameters multifrequency-signaling**	**location-parameters**
Outgoing Shuttle Exchange Cycle Timer	——	**system-parameters multifrequency-signaling**	**location-parameters**
MF Test Call Extension	——	**system-parameters multifrequency-signaling**	**system-parameters maintenance**
(all fields)	——	**terminal-parameters 302/603/606**	**terminal-parameters** (page 1)
(all fields)	——	**terminal-parameters 6400/607A1/4600**	**terminal-parameters** (page 2)
(all fields)	——	**terminal-parameters 8400**	**terminal-parameters** (page 3)
default parameter set	**Base Parameter Set**	**terminal-parameters**	——

2 of 2

Administration Changes screen

If the **station set port type** field is **IP**, and if the **Record IP Registrations in History Log?** field on the *System Parameters Features* screen is set to **y**, the history log already records IP registrations, just not unregistrations. Changes to the *Administration Changes* screen now records IP unregistrations (). This change is prompted by the E911 emergency location information number (ELIN) for wired IP feature.

This log is viewable through the **list history** and **notify history** commands. This log contains the following information:

- Date and time of feature use
- Extension number
- Port number

Unregistrations will be logged under login "ip-u". The "-u" part of the login is for consistency with PSA and ACTR, which use "-a" for associations and "-u" for unassociations.

Figure 12: Administration Changes screen

```
notify history


                            ADMINISTRATION CHANGES

Date Time  Login   Actn  Object      Qualifier
2/15 10:18 actr-d  cha   station     2005
2/15 10:18 actr-u  cha   station     2004
2/15 10:17 init    cha   system-param features
2/15 10:15 tti-m   cha   station     2004
2/15 10:15 tti-m   cha   station     2005
2/15 10:15 tti-s   cha   station     2005
2/15 10:12 actr-a  cha   station     2005
2/15 10:12 ip-u    cha   station     2006
2/15 10:12 ip-u    cha   station     2007
2/15 10:10 ip-u    cha   station     2008
2/15 10:10 ip-a    cha   station     2009
```

Attendant Console screen

On the *Attendant Console* screen (Figure 13), the field **IP Emergency Calls** in earlier releases of
Communication Manager has been renamed to **Remote Softphone Emergency Calls** in release 2.0. This
newly-renamed field also allows a new value — **as-on-local**. The value **extension** is no longer allowed
for this field. Use the **change attendant** *xx* command, where *xx* is the number of the attendant console
station. This change is prompted by the E911 emergency location information number (ELIN) for wired
IP feature.

This field only appears if the **IP Softphone** field in the *Station* screen is set to **y**. The restriction (current
in previous releases of Communication Manager) that this field can only be changed if the station is
unregistered remains in force for release 2.0.

Figure 13: Attendant Console screen, page 2

```
change attendant 2                                             Page 2 of 4

                           ATTENDANT CONSOLE

VIS FEATURE OPTIONS
            Auto Start? y
      Echo Digits Dialed? y

EMERGENCY CALL HANDLING
Remote Softphone Emergency Calls: as-on-local
         Emergency Location Ext: 1001
```

> **NOTE:**
> The fields are not accessible if the set type is not one of the IP telephone or IP softphone
> types.

Field descriptions

Remote Softphone Emergency Calls   This field has been renamed. In earlier releases of Communication Manager, this field was named **IP Emergency Calls**.

Valid entries	Usage
as-on-local	Type **as-on-local** to achieve the following results: • If the administrator chooses not to use the new feature outlined in the next bullet item and chooses to leave the **Emergency Location Extension** fields (that correspond to this station's IP address) on the *IP Address Mapping* screen blank, the value **as-on-local** sends the extension entered in the **Emergency Location Extension** field in the *Station* screen to the Public Safety Answering Point (PSAP). • If the administrator does choose to use the new feature and populates the *IP Address Mapping* screen with emergency numbers, the value **as-on-local** functions as follows: — If the **Emergency Location Extension** field in the *Station* screen is the same as the **Emergency Location Extension** field in the *IP Address Mapping* screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP). — If the **Emergency Location Extension** field in the *Station* screen is different from the **Emergency Location Extension** field in the *IP Address Mapping* screen, the value **as-on-local** sends the extension in the *IP Address Mapping* screen to the Public Safety Answering Point (PSAP).
block	Type **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
cesid	Type **cesid** to allow Communication Manager to send the CESID information supplied by the IP softphone to the PSAP. The end user enters the emergency information into the IP softphone. Use this entry for IP softphones with Road Warrior service that are near enough to the media server or switch that an emergency call routed over the trunk reaches the PSAP that covers the server or switch. If the media server or switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP softphone. If the media server or switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP softphone location, based on advice from the local emergency response personnel.

Valid entries	Usage
option	Type **option** to allow the user to select the option (**as-on-local**, **block**, or **cesid**) that the user selected during registration and the IP softphone reported. Use this entry for extensions that may be switched back and forth between IP softphones and a telephone with a fixed location.
	The user chooses between **as-on-local**, **block** and **cesid** on the softphone. A DCP telephone in the office use the **Emergency Location Extension** field in the *Station* screen.
	During registration, users and administrators see the following choices on the IP softphone user interface:

`This is the softphone interface equivalent to `**`as-on-local`**`:`

The box "Use the following information to identify your location to a PSAP should you need to make a 911 call: your extension number" is checked.

`This is the softphone interface equivalent to `**`block`**`:`

The box "Enable Emergency Call Handling Feature" is not checked.

`This is the softphone interface equivalent to `**`cesid`**`:`

The box "Use the following information to identify your location to a PSAP should you need to make a 911 call: telephone number _____" has a phone number entered.

Call Center Optional Features screen

The *Call Center Optional Features* screen (Figure 14) is screen 6 using the **display system-parameters customer-options** command.

For the Variables for Vectors feature to work, the **Call Center Release** field must be set to **12.0** or later. In addition, your license file must set both the **Vectoring (Basic)** and **Vectoring (Prompting)** fields to **y**. Once these conditions are met, the **Vectoring (Variables)** field can also be set to **y**.

For the Service level maximizer feature to work, the **Service Level Maximizer** field must be set to **y**.

Figure 14: Call Center Optional Features screen

```
display system-parameters customer-options                    Page 6 of 10

                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 12.0

                          ACD? n        PASTE (Display PBX Data on Phone)? n
                 BCMS (Basic)? n                          Reason Codes? n
       BCMS/VuStats Service Level? n             Service Level Maximizer? y
  BSR Local Treatment for IP & ISDN? n           Service Observing (Basic)? n
               Business Advocate? n      Service Observing (Remote/By FAC)? n
                 Call Work Codes? n            Service Observing (VDNs)? n
     DTMF Feedback Signals For VRU? n                         Timed ACW? n
               Dynamic Advocate? n                    Vectoring (Basic)? y
       Expert Agent Selection (EAS)? n                Vectoring (Prompting)? y
                        EAS-PHD? n             Vectoring (G3V4 Enhanced)? n
               Forced ACD Calls? n      Vectoring (ANI/II-Digits Routing)? n
            Least Occupied Agent? n      Vectoring (G3V4 Enhanced Routing)? n
           Lookahead Interflow LAI)? n                  Vectoring (CINFO)? n
  Multiple Call Handling (On Request)? n     Vectoring (Best Service Routing)? n
      Multiple Call Handling (Forced)? n            Vectoring (Holidays)? n
                                                    Vectoring (Variables)? y
            (NOTE: You must logoff & login to effect the permission changes.)
```

Dial Plan Analysis Table screen

The **Dialed String** fields on the *Dial Plan Analysis Table* screen (Figure 15) allows an entry up to four digits. Use the **display dialplan analysis** command. By widening the **Dialed String** column to four digits, customers can allocate blocks of 1000 numbers. This change is prompted by the Multi-Location Dial Plan feature.

The screen has also been increased to twelve pages, with three columns of fifteen entries on each page.

Figure 15: Dial Plan Analysis Table screen, page 1

```
display dialplan analysis                                         Page 1 of 12

                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full: 7

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

The **Call Type** allows a **Dialed String** value of either a given first digit (for example, **7**), a set of two-digit pairs (for example, **70**, **71**, **72**, ...), a set of three-digit pairs (for example, **701**, **702**, **703**, ...), or a set of four-digit pairs (for example, **7001**, **7002**, **7003**, ...).

The following apply:

- When a three-digit entry exists, a new three-digit entry with the same dial string cannot be added if the total length is different.

- When a four-digit entry exists, a new four-digit entry with the same dial string cannot be added if the total length is different.

- The **Dialed String** column on the *Dial Plan Analysis Table* screen allows mixed (different) length dial strings for the same first digit. This allows a mix of 1-digit, 2-digit, 3-digit, and 4-digit entries having the same first digit. This removes previous screen validations.

  The **Dialed String** column on the *Dial Plan Analysis Table* screen does not allow mixed (different) lengths for 2-digit, 3-digit, or 4-digit dial strings.

- A new entry cannot be administered if it causes an existing extension (ext), FAC, or TAC entry to become inaccessible. For example, let us say a customer has the following range administered:

```
    Dialed  Total   Call
    String  Length  Type
      4       4      ext
```

If the customer adds this entry:

```
    Dialed  Total   Call
    String  Length  Type
     41       3      fac
```

The system ensures that no extensions of the format **41xx** are administered. If any are administered, the new "fac" entry is blocked.

Feature Access Code (FAC) screen

A new field, **Attendant Access Code**, is added to page 1 of the *Feature Access Code (FAC)* screen (Figure 16). Use the **change feature-access-codes** command. This field only appears and is valid if an **attd** entry does not exist on the *Dial Plan Analysis* screen. You cannot have an entry in both the *Dial Plan Analysis* screen and the *Feature Access Code (FAC)* screen. This change is prompted by the Multi-Location Dial Plan feature.

Figure 16: Feature Access Code (FAC) screen, page 1

```
change features-access-codes                              Page 1 of 8

                        FEATURE ACCESS CODE (FAC)

          Abbreviated Dialing List1 Access Code:
          Abbreviated Dialing List2 Access Code:
          Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                    Announcement Access Code:
                    Answer Back Access Code:
                     Attendant Access Code: 00
       Auto Alternate Routing (AAR) Access Code: *88
       Auto Route Selection (ARS) - Access Code 1: 9        Access Code 2:
               Automatic Callback Activation:              Deactivation:
Call Forwarding Activation Busy/DA:       All: *20          Deactivation: #20
                      Call Park Access Code:
                    Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
               CDR Account Code Access Code:
                  Change COR Access Code:
              Change Coverage Access Code:

              Data Origination Access Code:
                 Data Privacy Access Code:
```

Attendant Access Code   The **Attendant Access Code** field on the *Feature Access Code (FAC)* screen takes on the same characteristics as the **attd** call type administered on the *Dial Plan Analysis* screen. Entries in this field must conform to the existing *Feature Access Code (FAC)* screen rules.

The *Dial Plan Analysis* screen allows administration of only one **attd** code that connects to one attendant. To allow more than one attendant access code in a single distributed network, the **Attendant Access Code** field has been added to the *Feature Access Code (FAC)* screen.

Valid entries	Usage
0–9	Attendant — Defines how users call an attendant. Attendant access numbers can start with any number from 0 – 9 and contain 1 or 2 digits. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code.

Another page of the *Feature Access Code (FAC)* screen has been changed, prompted by a different feature.

New fields for the Variables for Vectors feature have been added to page 6 of the *Feature Access Code (FAC)* screen (Figure 17). Use the **change feature-access-codes** command. These fields all you to assign nine unique vector variable feature access codes, in the format **VVx** (where **x** is a number from 1-–9), for **value** type variables. For an explanation of **value** type variables, see "Variables for Vectors screen" on page 52.

Figure 17: Feature Access Code (FAC) screen, page 6

```
change features-access-codes                                    Page 6 of 8

                        FEATURE ACCESS CODE (FAC)

                    Call Vectoring/Call Prompting Features

    Converse Data Return Code:  ____

Vector Variable 1 (VV1) Code:  ____
Vector Variable 2 (VV2) Code:  ____
Vector Variable 3 (VV3) Code:  ____
Vector Variable 4 (VV4) Code:  ____
Vector Variable 5 (VV5) Code:  ____
Vector Variable 6 (VV6) Code:  ____
Vector Variable 7 (VV7) Code:  ____
Vector Variable 8 (VV8) Code:  ____
Vector Variable 9 (VV9) Code:  ____
```

Feature-Related System Parameters screen

A new field, **Treat ISDN Call to Busy User as ACA Short Holding Time Call**, has been added to page 1 of the *Feature-Related System Parameters* screen (Figure 18). Use the **change system-parameters features** command. This field allows the administrator to indicate whether ISDN calls placed to user stations that are busy should be considered ACA short calls (see page 22).

> **NOTE:**
> The **Automatic Circuit Assurance (ACA) Enabled** field must be set to **y** for this field to appear.

Figure 18: Feature-Related System Parameters screen, page 4

```
change system-parameters features                              Page 4 of 12

                        FEATURE-RELATED SYSTEM PARAMETERS

                             Self Station Display Enabled? n
                                  Trunk-to-Trunk Transfer: none
    Automatic Callback - No Answer Timeout Interval (rings): 3
                      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
                               AAR/ARS Dial Tone Required? y
                                       Music/Tone on Hold: none
           Music (or Silence) on Transferred Trunk Calls? no
                          DID/Tie/ISDN Intercept Treatment: attd
   Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                 Automatic Circuit Assurance (ACA) Enabled? y
                                        ACA Referral Calls: local
                                  ACA Referral Destination: 8102
           ACA Short Holding Time Originating Extension: 8122
           ACA Long Holding Time Originating Extension: 8123
Treat ISDN Call to Busy User as ACA Short Holding Time Call? y
                Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
                   Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

Another page of the *Feature-Related System Parameters* screen has been changed, prompted by a different feature.

If an emergency call should drop (get disconnected), the public safety personnel will attempt to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, this feature automatically forwards that return call to the set that placed the emergency call for an administered period of time.

This Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same PBX as the extension that dialed 911. Customers who have several PBXs in a campus should assign emergency location extensions accordingly.

A new field, **Emergency Extension Forwarding (min)**, has been added to page 4 of the *Feature-Related System Parameters* screen (). Use the **change system-parameters features** command. This field sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off (drops). This change is prompted by the E911 emergency location information number (ELIN) for wired IP feature.

> **NOTE:**
> It is important to realize that the call forwarding timer, once it is running, applies to *all* incoming trunk calls and is not limited to just incoming calls from an emergency care provider. The reason is there is no way for the system to know that an incoming trunk call is actually a return call from an emergency care provider.

Figure 19: Feature-Related System Parameters screen, page 4

```
change system-parameters features                              Page 4 of 12

                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
              System Printer Endpoint:                    Lines Per Page: 60
                 EIA Device Bit Rate: 9600

 Emergency Extension Forwarding (min): 10

SYSTEM-WIDE PARAMETERS
                      Switch Name:
        Emergency Numbers - Internal:          External: 911
      No-License Incoming Call Number:

MALICIOUS CALL TRACE PARAMETERS
             Apply MCT Warning Tone? n       MCT Voice Recorder Trunk Group:

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:
```

Field descriptions

Emergency Extension Forwarding (min)

Valid entries	Usage
0–999	Type a number between 0–999 that represents the time (in minutes) that an incoming trunk call will forward to the extension that made the initial 911 call. The default value for both new installs and upgrades is **10**.
	If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes might remain on the timer.

While the emergency extension forwarding timer is running, the feature turns on the lamp on the call forwarding button, and appears in the **CF Destination Ext** field on the first page of the *General Status* screen. Use the **status station *n*** command, where *n* is the station number. Also, the Send All Calls (SAC) and Do Not Disturb (DND) features, if on, will temporarily be turned off. Once the Emergency Extension Forwarding timer expires, the Send All Calls feature will be turned on again.

> **NOTE:**
> The Do Not Disturb (DND) feature remains turned off and must be reset manually.

The status of the call forwarding, and of SAC and DND, appears on the first page of the *General Station* screen for the emergency location extension. Use the **status station *n*** command, where *n* is the station number.

Hunt Group screen

A new **Group Type** of **slm** is allowed on page 1 of the *Hunt Group* screen (Figure 20). Use the **change hunt-group *n*** command (where *n* is the hunt group number). This change is prompted by the Service level maximizer feature.

Figure 20: Hunt Group screen, page 1

```
change hunt-group 2                                               Page 1 of 3

                                  HUNT GROUP

            Group Number: 2                                        ACD? y
              Group Name: HNT-2                                  Queue? y
         Group Extension: 7330002                               Vector? y
              Group Type: slm
                      TN: 1
                     COR: 1                       MM Early Answer? n
           Security Code:
      ISDN Caller Display:

            Queue Length: 5
 Calls Warning Threshold: 2      Port:
  Time Warning Threshold: 15     Port:
```

Click **Next** to go to page 2 of the *Hunt Group* screen (Figure 21). The **Maximum Auto Reserve Agents** field only displays if the **Group Type** (on page 1) is **slm**.

Figure 21: Hunt Group screen, page 2

```
change hunt-group 2                                               Page 2 of 3

                                  HUNT GROUP

                  Skill? y           Expected Call Handling Time (sec): 180
                    AAS? n           Target Service Level (% in sec): 80 in 20
               Measured: both
      Supervisor Extension:


      Controlling Adjunct: none


        VuStats Objective:
 Timed ACW Interval (sec):                Maximum Auto Reserve Agents: 0
    Multiple Call Handling: none

                                        Redirect on No Answer (rings): 4
                                                    Redirect to VDN: 52002
                     Forced Entry of Stroke Counts or Call Work Codes? n
```

IP Address Mapping screen

A new column and fields, **Emergency Location Extension**, has been added to the first page of the *IP Address Mapping* screen (Figure 22). Use the **change ip-network-map** command. This change is prompted by the E911 emergency location information number (ELIN) for wired IP feature.

Figure 22: IP Address Mapping screen, page 1

```
change ip-network-map                                            Page 1 of X

                              IP ADDRESS MAPPING
                                                              Emergency
                                     Subnet            802.1Q  Location
 FROM IP Address   (TO IP Address or Mask)   Region     VLAN   Extension
  1.__2.__3.__0   1.__2.__3.255    24        __1       ___3    _____
  1.__2.__4.__4   1.__2.__4.__4    32        __2       ___0    _____
  1.__2.__4.__5   1.__2.__4.__5    __        __3       ___0    _____
  1.__2.__4.__6   1.__2.__4.__9    __        __4       ___4    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
 ___.___.___.___  ___.___.___.___  __        ___       ____    _____
```

The administrator should fill out the *IP Address Mapping* screen with emergency location extensions for every IP address range in their LAN's data network. Telephones that are physically close together but have different IP addresses could have the same ELIN. Once completed, the administrator would not need to update the *IP Address Mapping* screen again unless they re-wire the building.

If users tell the system administrator that they have moved their extensions, then the system administrator should make sure that the ALI database will have their telephone's new physical locations. After doing so, the administrator should update each user's *Station* screen.

Field descriptions

Emergency Location Extension   When updating Communication Manager, Release 2.0, from an earlier version of Communication Manager, the **Emergency Location Extension** fields take on the same values as the **Emergency Location Ext** field on the *Station* screen.

Valid entries	Usage
0–9	Enter the emergency location extension for this station — up to 7 digits.

Similar to the **Emergency Location Ext** field on the *Station* screen, these entries have to be blank or of a type extension in the dial plan. They do not have to be extensions on the local system. They can be UDP extensions on another PBX. The entries default to blank. A blank entry typically would be used for an IP softphone dialing in through PPP from somewhere outside the customer's network, and used outside the USA and Canada.

New and changed screens
Release 2.0 changed screens

IP Interfaces screen

Changes to the **add/remove ip-interface** command have prompted changes to the *IP Interfaces* screen (Figure 23). The layout of the fields on the *IP Interfaces* screen is changed to a vertical display for a single circuit pack rather than the previous horizontal display for multiple circuit packs.

Figure 23: IP Interfaces screen

```
add ip-interface 02c08                                            Page 1 of X

                            IP Interfaces

                      Type: CLAN
                      Slot: 02c08
                  Code Sfx: TN799 D
                 Node Name: clan1xxxxxxxxxx
                IP Address: 123.456.789.012
               Subnet Mask: ___.___.___.___
           Gateway Address: ___.___.___.___
        Enable Ethernet Pt? n
            Network Region: 20
                      VLAN: 0


 Number of CLAN Sockets Before Warning: 400
```

Field descriptions

Number of CLAN Sockets Before Warning   This field appears only if the value in the **Type** field is **CLAN**.

Valid entries	Usage
1–499	Default = **400**. Type a threshold number of CLAN sockets that can be used (for registering endpoints) before a warning is issued.

Type   This field is display-only, based on the board location entered on the command line.

Valid entries	Usage
CLAN, MEDPRO, VAL	Valid for S8700 and S8500 Media Servers.
PROCR	Valid for S8300 Media Server.

Slot   This field is display-only, based on the slot number entered on the command line.

Valid entries	Usage
2-6 alpha-numeric characters or (blank)	For the S8300 Media Server, the value is (blank).

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Code Sfx   This field is display-only, based on the circuit pack in the location specified on the command line.

Valid entries	Usage
7 characters or (blank)	The first 6 alpha-numeric characters are for the TN code, followed by 1 alpha character Code Suffix. If the board has an A suffix, the value in the **Sfx** portion of the field is (blank). For an S8300 Media Server, this field is (blank).

Node Name   This field is user-specified, based the name administered on the *IP Node Names* screen.

Valid entries	Usage
15 alpha-numeric characters	User-specified, based the name administered on the *IP Node Names* screen.

IP Address   This field is display only, from the *IP Node Names* screen, based on the node name entered.

Valid entries	Usage
valid IP address	Displays an IP address from the *IP Node Names* screen, based on the node name entered.

IP Network Region screen

With the increase to 250 possible network regions the *IP Network Region* screen was changed. Use the **change ip-network-region *n*** command (where *n* is the number of the network region you want to change).

The *Ip Network Region* screen shows specific IP region information on page 1 ([Figure 24](#)), which includes not only the codec set, but also shows bandwidth management information, such as:

- Whether the regions are directly or indirectly connected
- The WAN bandwidth limits (if any), the units for those bandwidth limits
- The intervening regions when the regions are indirectly connected

> **NOTE:**
> Only one path is allowed to be administered at this time for the non-adjacent regions.

Figure 24: IP Network Region screen, page 1

```
add ip-network-region 1                                          Page 1 of 19

                              IP NETWORK REGION

  Region: 1
Location: 1              Home Domain:
    Name: Rosebud NR 1
                                   Intra-region IP-IP Direct Audio: yes
AUDIO PARAMETERS                   Inter-region IP-IP Direct Audio: yes
   Codec Set: 1                              IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028                        RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS            Use Default Server parameters: y
 Call Control PHB Value: 34
       Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 6     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                RSVP Enabled? n
   H.323Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

For Linux platforms in Release 2.0, page 2 (Figure 25) is the LSP names page. This page covers the information for the LSP names in priority order.

Figure 25: IP Network Region screen, page 2

```
add ip-network-region 1                                          Page 2 of 19

                              IP NETWORK REGION

 LSP NAMES IN PRIORITY ORDER
 1
 2
 3
 4
 5
 6
```

To accommodate the maximum of 250 regions for the Linux platforms, seventeen additional pages are now needed for each region's inter-region connectivity information. Page 3 (Figure 26) to page 19 are called the *Inter Network Region Connection Management* screens. These seventeen pages show the inter-region connectivity, numbered 1–250, for 15 region pairs.

Figure 26: Inter Network Region Connection Management screen, page 3

```
add ip-network-region 1                                              Page 3 of 19

                  INTER NETWORK REGION CONNECTION MANAGEMENT

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

Field descriptions

Codec-set   Indicates which codec set is to be used between the two regions. If the two regions are not connected at all, this field should be blank. When the codec set is blank, the **direct-WAN**, **WAN-BW-limits**, and **Intervening-regions** entry fields are not displayed. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions.

Direct-WAN   Indicates whether the two regions (source and destination) are directly connected by a WAN link. The default value is y(es) if the **codec-set** field is not blank. If so, the **WAN-BW-limits** field displays, but the **Intervening-regions** fields do not. If the **direct-WAN** field is set to n(o), then the **WAN-BW-limits** field does not display, but the **Intervening-regions** fields are displayed.

WAN-BW-limits   The **WAN-BW-limits** field is divided into two parts – values and units.

- The value portion of the field allows entry of the bandwidth limits for direct WAN links. This field may be entered in the number of connections, bandwidth in Kbits/sec, bandwidth in Mbits/sec, or left blank. The default is blank.

    > **NOTE:**
    > For Release 2.0, the number must be less than or equal to **65** when the units part of the field is set to **Mbits/sec**.

- The unit portion of the field allows entry of the units for the bandwidth limits for direct WAN links. This field may represent connections, Kbits/sec, Mbits/sec, or NoLimit.

Intervening-regions   Four fields allowing entry of up to four intervening region numbers between the two indirectly-connected regions.

> **NOTE:**
> Entry is not allowed for indirect region paths until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

The **status ip-network-region** *n* command (where *n* is the network region you want to review) displays the modified *Inter Network Region Bandwidth Status* screen. This screen was modified to show statistics on the Call Admission Control-Bandwidth Limitation feature. The following two figures show two different command options.

The first command option (an existing option) is **status ip-network-region** *n* (Figure 27). Each line on the screen shows the connection that region *n* has to either a direct or indirect region. For the direct regions, the bandwidth and number of connections being used is shown in both the transmit and receive directions. For indirect regions, only the status of the connection is shown.

> **NOTE:**
> Regions not connected either directly or indirectly to region *n* are not shown.

Figure 27: Inter Network Region Bandwidth Status screen

```
 status ip-network-region 2                                     Page 1 of 1
                   INTER NETWORK REGION BANDWIDTH STATUS


                                                               #Times
 Src Dst   Conn    Conn      BW-limits  BW-Used (Kbits)  #-of-Connections  BW-Limit
 Rgn Rgn   Type    Stat                   Tx      Rx       Tx      Rx      Hit Today

   2   1   direct  pass      128 Kbits     0       0        0       0        0
   2   3   indirect pass
   2   4   indirect pass
   2   5   indirect pass
```

The second command option (a new option) is **status ip-network-region** *n/m* (Figure 28), where *n* and *m* are two separate and connected regions. Use this option for more details on the indirect regions. This command option shows all the bandwidth being used between region *n* and region *m*, which in this example includes all the intermediate regions because region *n* and region *m* are indirectly connected.

Figure 28: Inter Network Region Bandwidth Status screen

```
 status ip-network-region 2                                     Page 1 of 1
                   INTER NETWORK REGION BANDWIDTH STATUS


                                                               #Times
 Src Dst   Conn    Conn      BW-limits  BW-Used (Kbits)  #-of-Connections  BW-Limit
 Rgn Rgn   Type    Stat                   Tx      Rx       Tx      Rx      Hit Today

   2   1   direct  pass      128 Kbits     0       0        0       0        0
   1   3   direct  pass      256 Kbits     0       0        0       0        0
```

Field descriptions

Src Rgn   Source region

Dst Rgn   Destination region

Conn Type   Connection type (direct or indirect)

Conn Stat   Connection status (pass or fail)

BW-limits   Bandwidth and limits as administered on the *IP Network Region* screen

BW-Used (Kbits) Tx   Transmit bandwidth used (for direct connections only)

BW-Used (Kbits) Rx   Receive bandwidth used (for direct connections only)

#-of-Connections Tx   Transmit connection count (for direct connections only)

#-of-Connections Rx   Receive connection count (for direct connections only)

#Times BW-Limit Hit Today   Daily count of how many times the CAC threshold limits have been reached (for direct connections only)

> **NOTE:**
> This log is cleared at midnight (server time).

IP-Options System Parameters screen

An additional option has been added to a field on the *IP-Options System Parameters* screen (Figure 29). Use the **change system-parameters ip-options** command. The additional option for the **Intra-System IP DTMF Transmission Mode** field is **rtp-payload**. The option **rtp-payload** allows for dual-tone multi-frequency (DTMF) signals – in other words, touchtones – to be transmitted over SIP trunks.

When the **rtp-payload** option is applied to an IP connection, the VoIP audio resource (on TN2302 Medpros, or on G700 or G350 Media Gateways), is configured to implement RFC2833. RFC2833 is an internet standard for sending and receiving telephony events.

The VoIP resource detects speech on an internal bus. Normal speech is encoded according to the specified voice coder (for example, G711). The VoIP resources places the encoded audio into packets and identifies the audio packets with a well known payload type value.

When the other end receives the packets, based on this payload type value, the receiver knows the packets contain encoded audio and that the encoding was G711.

When the **rtp-payload** option is set, the VoIP resource not only detects speech, but it is also detecting DTMF tones. When a DTMF tone is detected, it is not encoded using the selected voice encoder. Instead, the VoIP resource produces a special packet with a special payload type. The special packet includes the dynamic payload type, the digit detected, duration, and volume. This packet is sent along in the same RTP stream as the encoded audio, but since it has a different payload type, the receiver handles it differently. The receiver regenerates a DTMF tone based on the data in the received packet.

Figure 29: IP-Options System Parameters screen

```
change system-parameters ip-options                                  Page 1

                      IP-OPTIONS SYSTEM PARAMETERS

IP MEDIA PACKET PERFORMANCE THRESHOLDS
   Roundtrip Propagation Delay (ms)      High: 800      Low: 400
                    Packet Loss (%)      High: 40       Low: 15
               Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10

RTCP MONITOR SERVER
                   Default Server IP Address: ___.___.___.___
                          Default Server Port: 5005
          Default RTCP Report Period (secs): 5

IP DTMF TRANSMISSION MODE
      Intra-System IP DTMF Transmission Mode: rtp-payload
      Intra-System IP DTMF: See Signaling Group Forms

H.248 MEDIA GATEWAY                  H.323 IP ENDPOINT
 Link Loss Delay Timer (min): 5       Link Loss Delay Timer (min): 60
                                        Primary Search Time (sec): 75
```

List Trace screen

Previously, the G700 Media Gateway call controller did not provide packet loss and jitter statistics for IP stations or trunks. The screens for displaying the information were available within Communication Manager, but were not populated with measurements obtained from the G700 VoIP engines.

With Communication Manager Release 2.0, VoIP engine statistics are collected from the G700 Media Gateway, and displayed in a manner consistent with existing gateway reporting formats for other platforms. The commands used for this change are:

- **list trace station** *n* (Figure 30)
- **list trace tac** *n* (Figure 31)
- **status station** *n* (Figure 32)
- **status trunk** *n* (Figure 33)

In each of these commands, *n* is the variable for the specific component you want to review.

The **list trace station** *n* command displays call progress, as well as jitter and packet loss statistics, for a specified IP station (Figure 30).

Figure 30: List Trace screen

```
list trace station 2152                                           Page 1
                              LIST TRACE

time              data
10:02:43     active station     2152 cid 0xa2
10:02:43     G711MU ss:off ps:20 rn:1/1 135.9.78.43:62210 135.9.77.174:5736
10:02:46     dial 2301
10:02:46     ring station    2301 cid Oxa2
10:02:46     alert station   2301 cid Oxa2n
10:02:54     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:1 Avg:0
10:02:54     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:02:55     active station     2301 cid 0xa2
10:03:05     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:1 Avg:0
10:03:05     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:03:15     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:1 Avg:0
10:03:15     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:03:25     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:1 Avg:0
10:03:25     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:03:35     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:1 Avg:0

 press CANCEL to quit -- press NEXT PAGE to continue.
```

The **list trace tac *n*** command displays call progress, as well as jitter and packet loss statistics, for a specified IP trunk (Figure 31).

Figure 31: List Trace screen

```
list trace tac 415                                                Page 1
                              LIST TRACE

time              data
10:05:32     dial 95676002
10:05:32     route-pattern  15 preference 1  cid Oxa3
10:05:32     seize trunk-group 15 member 3  cid Oxa3
10:05:32     Setup digits 6002
10:05:32     Calling Number & Name NO-CPNumber NO-CPName
10:05:32     Proceed trunk-group 15 member 3  cid Oxa3
10:05:33     Alert trunk-group 15 member 3  cid Oxa3
10:05:33     G711MU ss:off ps:20 rn:1/1 135.9.78.41:30588 135.9.77.174:5804
10:05:42     active trunk-group 15 member 3  cid Oxa3
10:05:43     Jitter:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:05:43     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:05:54     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:0 Avg:0
10:05:54     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0
10:06:04     Jitter:0 0 0 0 0 0 0 0 0 0: Buff:8 UC:0 Avg:0
10:06:04     Pkloss:0 0 0 0 0 0 0 0 0 0: Oofo:0 UC:0 Avg:0

 press CANCEL to quit -- press NEXT PAGE to continue.
```

The **status station** *n* command displays loss statistics for a specified IP station (Figure 32).

Figure 32: Station Status screen

```
status station 2152                                               Page 4
                            STATION STATUS

 NETWORK STATUS
       Average Jitter (ns)           Packet Loss per Second
        Last Ten Seconds               Last Ten Seconds
        # - more than 255ns            x - 100% loss

             0                              0           Per Call Info
             0                              0
             0                              0       Out of Order Counter: 0
             0                              0       SSRC Change for Call: 0
             0                              0        Last Rx Sequence #: 43394
             0                              0        Last Tx Sequence #: 62421
             0                              0
             0                              0
             0                              0
             0                              0

 SUMMARY
 Worst Case this Call (ns): 1      Worst Case this Call: 1
    Average this Call (ns): 0         Average this Call: 0
  Current Buffer Size (ns): 9
```

The **status trunk** *n* command displays jitter and loss statistics for a specified IP trunk (Figure 33).

Figure 33: Trunk Status screen

```
status trunk 1573                                                 Page 2
                            TRUNK STATUS

 NETWORK STATUS
       Average Jitter (ns)           Packet Loss per Second
        Last Ten Seconds               Last Ten Seconds
        # - more than 255ns            x - 100% loss

             0                              0           Per Call Info
             0                              0
             0                              0       Out of Order Counter: 0
             0                              0       SSRC Change for Call: 0
             0                              0         Last Rx Sequence #: 43394
             0                              0         Last Tx Sequence #: 62421
             0                              0
             0                              0
             0                              0
             0                              0

 SUMMARY
 Worst Case this Call (ns): 1      Worst Case this Call: 1
    Average this Call (ns): 0         Average this Call: 0
  Current Buffer Size (ns): 9
```

Locations screen

Three new columns are added to the *Locations* screen (Figure 34):

- **ARS FAC** field
- **Attd FAC** field
- **Prefix** field

> **NOTE:**
> These three columns only appear if the **Multiple Locations** field on the third page of the *Optional Features* screen is set to **y**. Otherwise, the three columns and the column headers do not appear.

This change is prompted by the Multi-Location Dial Plan feature.

Figure 34: Locations screen, page 1

```
change locations                                              Page 1 of 1
                                    LOCATIONS

                    ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc. Name              Timezone   Rule    NPA   ARS   Attd        Pre-   Foreign
 No                     Offset                   FAC   FAC         fix    Rte pat
  1:   Main             + 00:00    0
  2:   _____ _ __:__     __      ___   ___   __         ____   _____
  3:   _____ _ __:__     __      ___   ___   __         ____   _____
  4:   _____ _ __:__     __      ___   ___   __         ____   _____
  5:   _____ _ __:__     __      ___   ___   __         ____   _____
  6:   _____ _ __:__     __      ___   ___   __         ____   _____
  7:   _____ _ __:__     __      ___   ___   __         ____   _____
  8:   _____ _ __:__     __      ___   ___   __         ____   _____
  9:   _____ _ __:__     __      ___   ___   __         ____   _____
 10:   _____ _ __:__     __      ___   ___   __         ____   _____
 11:   _____ _ __:__     __      ___   ___   __         ____   _____
 12:   _____ _ __:__     __      ___   ___   __         ____   _____
 13:   _____ _ __:__     __      ___   ___   __         ____   _____
 14:   _____ _ __:__     __      ___   ___   __         ____   _____
```

The following applies for all fields:

- If the **Multiple Locations** field on the *Optional Features* screen is set to **y**, 10, 50, or 250 rows appear where data can be entered.

- If the **Multiple Locations** field on the *Optional Features* screen is set to **n**, only one row appears where data can be entered.

ARS FAC   The **ARS FAC** field is controlled by the **Multiple Locations** field on the *Optional Features* screen (use the **display system-parameters customer-options** command). Administration of this field must follow the same rules that exist for administering an ARS code on the *Feature Access Code (FAC)* screen.

Valid entries	Usage
0 to **9**	Any valid FAC format is acceptable, up to four digits. Characters ***** or **#** are permitted, but only in the first position. Many locations are expected to share the same access code.

Attd FAC   The **Attd FAC** field is controlled by the **Multiple Locations** field on the *Optional Features* screen (use the **display system-parameters customer-options** command). Administration of this field must be a **Call Type** of **FAC/DAC** on the *Dial Plan Analysis Table* screen for first digit **0–9**.

> **NOTE:**
> Within a dial plan, **FAC/DAC** codes and extensions cannot both start with the same first digits.Either the **FAC/DAC** entries or the block of extensions must be changed to have a different first digit.

A user cannot administer an **Attd FAC** unless an **Attendant Access Code** has first been administered on either the *Dial Plan Analysis Table* screen or the *Feature Access Code (FAC)* screen.

Valid entries	Usage
0 to **9**	Values up to two digits are permitted. Characters ***** or **#** are not permitted. Many locations are expected to share the same access code.

Prefix   The **Prefix** field is used for prepending the leading digits for *Uniform Dial Plan Table* screen entries for all calls. This field is controlled by the **Multiple Locations** field on the *Optional Features* screen (use the **display system-parameters customer-options** command). Administration of this field must be a **Call Type** of **FAC/DAC** on the *Dial Plan Analysis Table* screen for first digit **0–9**.

Valid entries	Usage
0 to **9**	Values from one to five digits (0–99999) are permitted.

Optional Features screen

Before you can administer Extension to Cellular/off-PBX station (OPS) extensions, certain settings must be enabled for your system, as determined by the installed license file. These settings are on the *Optional Features* screen. This change is prompted by enhancements to the Extension to Cellular feature.

To make sure the system is set up to administer Extension to Cellular/OPS extensions:

1   Type **display system-parameters customer-options** and press **Enter**.

The *Optional Features* screen appears (Figure 35).

Figure 35: Optional Features screen, page 1

```
display system-parameters customer-options                    Page   1 of 10
                              OPTIONAL FEATURES

 G3 Version: V12                               RFA System ID (SID): 123456789012
   Location: 2                                 RFA Module ID (MID): 123456
   Platform: 2
                                                             USED
                                          Maximum Ports: 300   174
                              Maximum XMOBILE Stations: 30    28
                      Maximum Off-PBX Telephones - EC500: 1200  0
                      Maximum Off-PBX Telephones -   OPS: 1200  0
                      Maximum Off-PBX Telephones - SCCAN: 0     0



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

The following settings must be enabled for your system, as determined by the installed license file.

On page 1 of the *Optional Features* screen:

- the **G3 Version** field must be set to **V12** or greater.
- either the **Maximum Off-PBX Telephones - EC500** field, or the **Maximum Off-PBX Telephones - OPS** field must be greater than zero. Either or both of these fields must be set to the number of stations that are to be used for Extension to Cellular/OPS.

> **NOTE:**
> The **Maximum Off-PBX Telephones - SCCAN** field is not used at this time.

Field descriptions

Maximum Off-PBX Telephones - EC500   The **Maximum Off-PBX Telephones - EC500** field is a new field with Release 2.0 having following parameters:

Valid entries	Usage
0 to system limit	Default = 0. Type a number between **1**–**99999**. The "system limit" is defined as follows:
	On DEFINITY® systems, the upper limit is 1/2 of the maximum number of administrable stations.
	On S8300, S8500, and S8700 systems, the upper limit is the maximum number of administrable stations.
	Stations that are administered for any Extension to Cellular/OPS application count against this limit.

To be usable, the system must also have the **IP Trunks**, **ISDN-BRI Trunks**, or **ISDN-PRI** field enabled. Those fields are on page 4 of the *Optional Features* screen.

Maximum Off-PBX Telephones - OPS   The **Maximum Off-PBX Telephones - OPS** field is a new field with Release 2.0 having following parameters:

Valid entries	Usage
0 to license max	Default = 0. Type a number between **1–99999**. The "license max' is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations.

- On S8300, S8500, and S8700 systems, the upper limit is the maximum number of administrable stations.

Stations that are administered for any Extension to Cellular/OPS application count against this limit.

To be usable, the system must also have the **IP Trunks**, **ISDN-BRI Trunks**, or **ISDN-PRI** field enabled. Those fields are on page 4 of the *Optional Features* screen.

Registered IP Stations screen

The Registered IP Stations screen (<u>Figure 36</u>) has been modified, based on changes to the **list registered-ip-stations** command. (For more information on the changes to the command, see <u>"list registered-ip-stations" on page 94</u>.)

The screen has been reformatted, and the following new fields were added:

- **Station Type**

- **Product ID**

- **Product Release**

- **Stations IP Address/Port**

- **Network Region**

Figure 36: Registered IP Stations screen

```
list registered-ip-stations gatekeeper-address 135.9.49.156

                        REGISTERED IP STATIONS

Command Options: gatekeeper-address 135.9.49.156

Station   Station  Product       Product  Stations IP      Network    Original
Ext       Type     ID            Release  address/port     Region     Port

3030      4624     IP_Phone      1.700    135.9.44.56         1
3031      4624     IP_Soft       1.800    135.9.44.56         1
3032      6408D+   IP_Soft       1.800    135.9.44.57         2         1E0303
3033      console  IP_eCons      1.500    135.9.44.57         2
3034      606A1    IP_Agent      1.800    135.9.44.57         2
3035      8410D    IP_ROMax      1.100    135.9.44.58         3
```

Signaling Group screen

An additional option has been added to a field on the *Signaling Group* screen (Figure 37). Use the
change signaling-group *n* command (where ***n*** is the number of the signaling group you want to change).
The additional option for the **DTMF Over IP** field is **rtp-payload**. The option **rtp-payload** allows for
dual-tone multi-frequency (DTMF) signals – in other words, touchtones – to be transmitted over
SIP trunks.

For more information on how this option works, see "IP-Options System Parameters screen" on page 73.

Figure 37: Signaling Group screen, page 1

```
change signaling-group 1                                           Page 1 of 5

                             SIGNALING GROUP

 Group Number: 1              Group Type: h.323
                           Remote Office? n          Max number of NCA TSC: 0
                                  SBS? n              Max number of CA TSC: 0
                                                    Trunk Group for NCA TSC:
        Trunk Group for Channel Selection: 1
           Supplementary Service Protocol: a         Network Call Transfer: n

           Near-end Node Name: clan1b09         Far-end Node Name: Newclan1c20
          Near-end Listen Port: 1720            Far-end Listen Port: 1720
                                               Far-end Network Region:
                  LRQ Required? n              Calls Share IP Signaling Connection? n
                  RRQ Required? n
                                               Bypass If IP Threshold Exceeded? n

                  DTMF over IP: rtp-payload       Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
                                                    Interworking Message: PROGress
```

Station screen

On the *Station* screen (Figure 38), the field **IP Emergency Calls** in earlier releases of Communication
Manager has been renamed to **Remote Softphone Emergency Calls** in release 2.0. This newly-renamed
field also allows a new value — **as-on-local**. The value **extension** is no longer allowed for this field. Use
the **change station *n*** command, where ***n*** is the number of the station.

This field only appears if the **IP Softphone** field in the *Station* screen is set to **y**. The restriction (current
in previous releases of Communication Manager) that this field can only be changed if the station is
unregistered remains in force for release 2.0. This change is prompted by the E911 emergency location
information number (ELIN) for wired IP feature.

Figure 38: Station screen, page 3

```
change station 1001                                            Page 3 of 5
                                  STATION

              IP Audio Hairpinning? y
   Direct IP-IP Audio Connections? y

EMERGENCY CALL HANDLING
Remote Softphone Emergency Calls: as-on-local
          Emergency Location Ext: 1001
```

NOTE:
The fields are not accessible if the set type is not one of the IP telephone or IP softphone types.

Field descriptions

Remote Softphone Emergency Calls   This field has been renamed. In earlier releases of Communication Manager, this field was named **IP Emergency Calls**.

Valid entries	Usage
as-on-local	Type **as-on-local** to achieve the following results: • If the administrator chooses not to use the new feature outlined in the next bullet item and chooses to leave the **Emergency Location Extension** fields (that correspond to this station's IP address) on the *IP Address Mapping* screen blank, the value **as-on-local** sends the extension entered in the **Emergency Location Extension** field in the *Station* screen to the Public Safety Answering Point (PSAP). • If the administrator does choose to use the new feature and populates the *IP Address Mapping* screen with emergency numbers, the value **as-on-local** functions as follows: — If the **Emergency Location Extension** field in the *Station* screen is the same as the **Emergency Location Extension** field in the *IP Address Mapping* screen, the value **as-on-local** sends the extension to the Public Safety Answering Point (PSAP). — If the **Emergency Location Extension** field in the *Station* screen is different from the **Emergency Location Extension** field in the *IP Address Mapping* screen, the value **as-on-local** sends the extension in the *IP Address Mapping* screen to the Public Safety Answering Point (PSAP).
block	Type **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.

Valid entries	Usage
cesid	Type **cesid** to allow Communication Manager to send the CESID information supplied by the IP softphone to the PSAP. The end user enters the emergency information into the IP softphone.
	Use this entry for IP softphones with Road Warrior service that are near enough to the media server or switch that an emergency call routed over the trunk reaches the PSAP that covers the server or switch.
	If the media server or switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP softphone. If the media server or switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP softphone location, based on advice from the local emergency response personnel.
option	Type **option** to allow the user to select the option (**as-on-local**, **block**, or **cesid**) that the user selected during registration and the IP softphone reported. Use this entry for extensions that may be switched back and forth between IP softphones and a telephone with a fixed location.
	The user chooses between **as-on-local**, **block** and **cesid** on the softphone. A DCP telephone in the office use the **Emergency Location Extension** field in the *Station* screen.
	During registration, users and administrators see the following choices on the IP softphone user interface:
	This is the softphone interface equivalent to **as-on-local**:
	The box "Use the following information to identify your location to a PSAP should you need to make a 911 call: your extension number" is checked.
	This is the softphone interface equivalent to **block**:
	The box "Enable Emergency Call Handling Feature" is not checked.
	This is the softphone interface equivalent to **cesid**:
	The box "Use the following information to identify your location to a PSAP should you need to make a 911 call: telephone number _____" has a phone number entered.

Station screen (for NI-BRI)

A new station type, **NI-BRI**, has been added to the *Station* screen. (For more information on NI-BRI, see "National ISDN (NI-1 and NI-2) BRI voice endpoint support" on page 27.) Use the **change station** *n* command, where *n* is the number of the station.

When the **NI-BRI** station type is selected, the remaining fields on the *Station* screen reflect only those fields that are relevant to NI-BRI (Figure 39).

Figure 39: Station screen, page 1

```
add station next                                              Page 1 of 3
                                STATION

Extension: 22878                        Lock Message: n            BCC: 0
     Type: NI-BRI                       Security Code:             TN: 1
     Port:                           Coverage Path 1:             COR: 1
     Name:                           Coverage Path 2:             COS: 1
                                     Hunt-to Station:

STATION OPTIONS
            Loss Group: 3

                                                  Message Lamp Ext: 22878

       Display Language: english
                  Model: other



BRI OPTIONS
                XID? n        Fixed TEI? n
                            Endpt Init? y     SPID: 22878       Endpt ID: 01
```

Click the second tab to go to the next screen (<u>Figure 40</u>).

Figure 40: Station screen, page 2

```
add station next                                              Page 2 of 3
                                STATION

FEATURE OPTIONS
         LWC Reception: spe
          LWC Activation? y                    Coverage Message Retrieval? y
  LWC Log External Calls? n
           CDR Privacy? n
   Redirect Notification? y


                                                    Restrict Last Appearance? y

                                       Per Station CPN – Send Calling Number?


    MWI Served User Type:
           AUDIX Name:
                                                    Coverage After Forwarding? y
```

Click the third tab to go to the next screen (<u>Figure 41</u>).

Figure 41: Station screen, page 3

```
add station next                                              Page 3 of 3
                               STATION

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




ABBREVIATED DIALING
     List1:                     List2:                     List3

BUTTON ASSIGNMENTS
 1: call-appr                          6:
 2: call-appr                          7:
 3: call-appr                          8:
 4:                                    9:
 5:                                   10:
```

Station Status screen

Two new fields – **Product ID and Release** and **Native NAT Address** – are added to the third page of the *Station Status* screen, *Call Control Signaling* (Figure 42). Use the **status station _n_** command (where _n_ is the station number).

Figure 42: Call Control Signaling screen, page 3

```
status station 3032                                           Page 3 of 3
                          CALL CONTROL SIGNALING

                  Switch            IP                   IP
                  Port       Switch-End Addr:Port   Set-End Addr:Port
     IP Signaling: 01E0617   172. 28.208. 21 :1720  172. 28. 3. 77:1673
          H.245:
       Node Name:            clan
   Network Region:           1                       1
                            AUDIO CHANNEL
                  Switch            IP                   IP
                  Port       Switch-End Addr:Port   Set-End Addr:Port
         Audio:
       Node Name:
   Network Region:
    Audio Connection Type: ip-tdm
    Product ID and Release: IP_Phone 1.700
  H.245 Tunneled in Q.931? does not apply
      Registration Status: registered-authenticated
    MAC Address: 00:60:1d:24:47:38         Native NAT Address: 10.0.0.1
```

Field descriptions

Product ID and Release   A ten-character field containing the product identifier of the endpoint that is registered with the associated extension, plus a five-character field, in the format **9.999**, that contains the release number of the endpoint as it provided to the gatekeeper upon registration.

Native NAT Address   A fifteen-character field, in the format **999.999.999.999**, that specifies the NAT IP address of the endpoint when a network device is providing the Network Address Translation function on behalf of the endpoint. The endpoint's NAT address is provided to the gatekeeper upon registration.

Also see Figure 32, Station Status screen, on page 76.

Trunk Status screen

See Figure 33, Trunk Status screen, on page 76.

System Capacity screen

A new field, **Meet-me Conference vectors per system**, has been added on the third page of the *System Capacity* screen (Figure 43). Use the **display capacity** command. The *System Capacity* screen displays:

- used meet-me conferencing vectors
- available meet-me conferencing vectors
- the system limit for meet-me conferencing vectors

Also, the group heading on page 10 of the *System Capacity* screen (not shown) is changed to **Concurrent Registration Counts**.

Figure 43: System Capacity screen, page 3

```
display capacity                                                   Page 3 of 12
                               SYSTEM CAPACITY

                                                                System
                                                  Used Available  Limit
                                                  ----------------------

                 CALL COVERAGE
                         Coverage Answer Groups:   0    1000      1000
                                 Coverage Paths:   0    9999      9999

                             Call Pickup Groups:   0    5000      5000

                                   Call Records:   -      -      15424

                 CALL VECTORING/CALL PROMPTING
                    Total Vector Directory Numbers:  0  20000     20000
               Meet-me Conference VDNs per system:   0   1800      1800
                         Total Vectors per system:   0    999       999
         Meet-me Conference vectors per system:     0    999       999
          BSR Application-Location Pairs per system: 0   2560      2560
```

Uniform Dial Plan Table screen

A new value, **L***x* (where *x* indicates a number of digits), is allowed in the **Insert Digits** field in the *Uniform Dial Plan Table* screen ([Figure 44](#)). Use the **change uniform-dialplan *x*** command, where *x* is the number of the dial plan. This change is prompted by the Multi-Location Dial Plan feature.

Figure 44: Uniform Dial Plan Table screen, page 1

```
change uniform-dialplan 3                                                Page 1 of 2
                           UNIFORM DIAL PLAN TABLE

                                                              Percent Full: 0

  Matching            Insert             Node    Matching            Insert             Node
  Pattern    Len Del Digits Net Conv Num  Pattern    Len Del Digits Net Conv Num
  3            3   0  L1     ext  y                                              n
  5            5   1  005    aar  y                                              n
  538          7   0         aar  y                                              n
  6            5   1  006    aar  y                                              n
  852          7   0         aar  y                                              n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
                                      n                                          n
```

The new value **Lx** represents taking the location prefix from the *Locations* screen and prepending the digits to the dialed string. The value *x* on the *Uniform Dial Plan Table* screen must match the number of digits in the location prefix.

Use of this new value requires the **Multiple Locations** field to first be set to **y** on the *Optional Features* screen.

Field descriptions

Insert Digits .The variable *x* in the value **L***x* in the **Insert Digits** field is only valid when the **Call Type** on the *Dial Plan Analysis Table* screen (the **Net** field on the *Uniform Dial Plan Table* screen) is **ext**.

Valid entries	Usage
1 to **5**	Type a number between 1 and 5 that represents the number of digits that should be prepended to (added to the front of) the dialed string. The number of digits should be equal to the number of digits of the location prefix.

4 New and changed commands

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

Release 2.0 new commands

Avaya Communication Manager, Release 2.0, includes the following new commands.

G650 Media Gateway

The addition of the new G650 Media Gateway – with the new TN2312BP IP Server Interface (IPSI) circuit pack, and the new 655A power supply – has produced numerous new or changed commands. The new and changed commands are:

- **add cabinet**
- **add fiber-link**
- **add/change/list ipserver-interface**
- **change circuit-packs**
- **change system-parameters duplication**
- **display system-parameters customer-options**
- **get power-shutdown**
- **list configuration hardware-group**
- **list configuration power-supply**
- **list registered-ip-stations**
- **recycle carrier**
- **set emergency-xfr**
- **status environment**
- **test customer-alarm**
- **test environment**

The many commands and accompanying screens relating to the G650 are not shown in this document. They are described in the *Maintenance Commands Reference*, 555-245-101.

add off-pbx-telephone station-mapping

The Extension to Cellular feature introduces a new command, **add off-pbx-telephone station-mapping**. This command displays a new screen, the *Stations with Off-PBX Telephone Integration* screen.

In prior releases of Extension to Cellular, cell phones had to be administered as XMOBILE stations. If there were two line appearances allowed on your cell phone (for example, if Call Waiting was a feature that was supported on your cell phone), two separate XMOBILE station settings had to be administered.

With Communication Manager release 2.0, this is no longer needed. See Stations with Off-PBX Telephone Integration screen, page 1 on page 46 for information on this new screen and its function.

change extension-station

The Multi-Location Dial Plan feature introduces a new command, **change extension-station** *n* (where *n* is a valid telephone extension number to be changed). This command allows an administrator to change a station extension on the switch without removing the station then adding it back.

> ⚠️ **CAUTION:**
> The **change extension-station** command changes most instances of an extension as it appears in the system. However, if an extension is used as an **emergency location extension** on the *Station* screen or the *IP Address Mapping* screen, the change is blocked.

When you use the **change extension-station** *n* command, a new screen — the *Change Station Extension* screen — appears. See Change Station Extension screen on page 39 for information on this new screen.

Action	Object	Qualifier
change	**extension-station**	*n*

change off-pbx-telephone feature-name-extensions

The Extension to Cellular feature introduces a new command, **change off-pbx-telephone feature-name-extensions**. This command displays a new screen, the *Extensions to Call which Activate Features By Name* screen.

See Extensions to Call which Activate Features By Name screen on page 43 for information on this new screen and its function.

change/display variables

The Variables for Vectors feature introduces two new commands, **change variables** and **display variables**. The **change variables** command allows an administrator to define variables to be used in vector steps for call vectoring. The **display variables** command displays a list of already defined variables.

When you use the **change variables** command, a new screen — the *Variables For Vectors* screen — appears. See Variables for Vectors screen on page 52 for information on this new screen.

list ip-interface

The new **list ip-interface** *<all/clan>* command displays the *IP Interfaces* screen. This new screen lists information about the IP interfaces administered on the system.

- When the *clan* option is used, all the administered C-LANs are listed showing the new **Num Skts Warn** field and not showing the **Type** field.

- When the *all* option is used, all the administered IP interfaces are listed showing the **Type** field and not showing the **Num Skts Warn** field.

The IP interface types depend on the media server – **C-LAN**, **MEDPRO**, and **VAL** types for the S8700 and S8500 Media Servers, and the **PROCR** type for the S8300 Media Server.

The new **list ip-interface** *all* command is identical to the current **display ip-interfaces** command. The new **change/display ip-interface** command is changed for Communication Manager Release 2.0 as described in Changed Commands/Screens.

Valid parameters

Action	Object	Qualifier
list	**ip-interface**	*<clan	all>*

See IP Interfaces screen on page 44 for information on this new screen and its function.

list multi-media ip-unregistered

A new option, **ip-unregistered**, has been added to the existing **list multi-media** command. This command shows a list of all unregistered IP telephones. This new option allows an administrator to quickly identify what IP telephones are unregistered and to resolve any related network problems.

> **NOTE:**
> This new command option does not list IP softphones that are unregistered, since in most instances, the end-user has purposely unregistered their IP softphones to regain control back at their office telephone set.

Valid parameters

Action	Object	Qualifier
list	**multi-media ip-unregistered**	

reset ip-stations

Use the **reset ip-stations** command to simultaneously reset all IP endpoints on a system, or a certain group of IP stations. You can limit the reset to only IP phones, to IP phones in a specific network region, or to all IP endpoints in a specific network region.

Use **reset ip-stations** to reset H.323 stations including:

- IP phones
- IP Softphones
- IP Agents
- IP E-consoles
- All endpoints that appear as IP stations to Communication Manager, including stations on R300 Remote Office and stations on G350 Media Gateways

Use **reset ip-stations** to initiate simultaneous firmware upgrades to many IP stations, or a certain group of IP stations. You can reset IP stations on one IP network region to prevent overloading a system with large numbers of IP station resets.

When **reset ip-stations** is submitted, each IP station receives a reset message and is unregistered. When **reset ip-stations** is run, a new event is logged. See the Events Report screen on page 42 for information on this screen and its function.

Valid parameters

Action	Object	Qualifier
reset	**ip-stations**	*<ip-phones	all-ip>* and *<ip-network-region (1-250)	all-regions>*

status clan-usage

The new **status clan-usage** command displays the *IP Interfaces* screen.

See CLAN (TN799x) Socket Usage screen on page 41 for information on this new screen and its function.

status media-processor

The new **status media-processor** *<cabinet/carrier/slot>* command displays a new screen showing the busyout status of the specified MedPro or IPMedPro media processor board. This command displays a new screen, the *IP Interface Status* screen.

See IP Interface Status screen on page 45 for information on this new screen and its function.

status ip-network-region

The **status ip-network-region** command has a new option: *n/m*, where *n* and *m* are two separate and connected regions. See page 72 under "IP Network Region screen" for more information.

test media-gateway

A new command, **test media-gateway**, allows customers to run an H248 link audit. The H248 link audit might be necessary if, for instance, there is an alarm against a media gateway for being unregistered, even though the media gateway is registered. For another example, the **status media-gateway** and **list media-gateway** commands might show that a media gateway is unregistered, when it is registered and processing phone calls.

These conflicts can arise if there has been a server interchange while a media gateway is registering or un-registering, because it is possible for an inter-process message to become lost.

The **test media-gateway** *xxx* command (where *xxx* is the number of the gateway being tested) has one test:

Test #1527 – Link Audit. This test verifies that the link states between various system processes are consistent, and clears up any erroneous alarms.

Two symptoms of a problem can be resolved by this test:

- A server interchange or other system operation occurs that results in a registration alarm for a media gateway, although the media gateway is registered and functioning properly
- The display events form shows that a media gateway is trying to register, but is being denied access (usually the result of a lost message because of abnormal server operation)

The test result is always PASS. Therefore, there are no error codes associated with the test.

Release 2.0 changed commands

Avaya Communication Manager, Release 2.0, includes the following changed commands.

add/remove ip-interface

The existing **change/display ip-interface** command is changed to include the **add** and **remove** actions, and a specification of a single IP interface circuit pack location. The full command syntax is **add/change/remove/display ip-interface** *<cabinet/carrier/slot | procr>*

- For the S8700 and S8500 Media Servers, you must specify the slot location of a C-LAN, Medpro, or VAL circuit pack.
- For the S8300 Media Server, **procr** is the only valid argument. In this case, the IP interface is embedded in the S8300 processor.

See IP Interfaces screen on page 68 for information on this new screen and its function.

add station next

The command **add station next** ignores the new entries in the *Uniform Dial Plan Table* screen when calculating the next extension number to use. This is consistent with the *Uniform Dial Plan Table* screen and the **add station next** command today. The **add station next** command ignores entries in the *Uniform Dial Plan Table* screen if the type is other than **ext**.

For example, if the dial plan has an entry for leading digit **7**, length **5**, as an extension (**ext**), and the *Uniform Dial Plan Table* screen has an entry for matching pattern **72**, length **5**, as **aar**, the **add station next** command would skip all instances of 72000–72999 as valid extensions that can be used.

> **NOTE:**
> If someone explicitly entered an extension, say **72111**, the **add station next** command would allow it. The same is true for the new *Uniform Dial Plan Table* screen entry.

campon-busyout media-processor

A new object key word **media-processor** has been added to the **campon-busyout** command. The syntax is now **campon-busyout media-processor** *<cabinet/carrier/slot>*. The command produces a *Command Results* report that shows the port address, the maintenance object name, any alternate name for the maintenance object, and any result and error codes.

list registered-ip-stations

Four new options have been added to the **list registered-ip-stations** command that enhance the sorting capabilities of the command, including the ability to sort by network region. A new field, **Command Options**, has been added to the *Registered IP Stations* screen that repeats the qualifiers specified after the **list registered-ip-stations** command. See Registered IP Stations screen on page 80 for information on this new screen and its function.

Using the options allows you to see only the extensions that have the attribute you specify:

- **list registered-ip-stations gatekeeper-address** *xxx.xxx.xxx.xxx* – This option lists the CLAN's or processor's IP address.
- **list registered-ip-stations network-region** *xx* – This option lists the network region of the extension.
- **list registered-ip-stations product-id** *xxxxxx* – This option lists the product type that is registered to the extension. For example, if a softphone is registered to a hardphone's extension, this command displays the release information of the registered endpoint. The **Product Release** field is the release information passed from the endpoint during registration. It contains one major character (**x**) and three minor characters (**y**): **x.yyy**.
- **list registered-ip-stations station-type** *xxxxxx* – This option lists the administered set type. This is needed since the firmware loads may be different between IP phone types.

list usage

The **list usage** command allows short extensions as a qualifier for the command. The *List Usage Report* that results includes all instances of a short extension's administration.

list usage ip-address

It is often necessary to determine the extension number associated with an IP address. Other than checking the status of one station at a time by extension number, there had been no way to get a cross-reference of IP address to extension.

The **list usage ip-address** *n* command (where *n* is the IP address) has been enhanced. The *List Usage Report* that results now shows station/extension usage.

save translation

Prior to Release 2.0, the **save translation** command caused **filesync** to run to all LSPs. Since many administrators execute the **save translation** command many times a day, this can cause a heavy load on the network. This also can cause the SAT to lock for many hours, depending on the size of the network, the size of translations, and the number of LSPs in the system.

Four changes to the **save translation** command are added to help in lessening this issue:

- **New qualifier:** A new optional qualifier *[lsp]* is added to the **save translation** command. This option specifies the IP address of the desired LSP. An IP address cannot be entered without the *[lsp]* qualifier. The specified IP address must also exist on the **node-names** IP screen, or it will not be accepted. The message passed to begin **filesync** is modified to include the information about the type of **save translation** being issued.

- **Filesync change:** If the message sent to **filesync** indicates that the qualifier was not present, **filesync** will only be run to the standby server. If the message sent to **filesync** indicates that the qualifier was present, **filesync** will run to the standby server, as well as to all LSPs.

- **Scheduled maintenance:** There is an option on the *System Maintenance* screen (use the **change system-maintenance** command): **Update LSPs when saving translations?**. This option indicates whether the **filesync** to LSP is on or off during the scheduled maintenance save translations. This field defaults to **y**.

- **Registering LSPs:** The main server issues a **filesync** command when an administered LSP registers with the main server.

The **save translation** command by itself (with no qualifier) causes **filesync** to run to the active and standby server. It does not initiate a **filesync** to any administered LSPs.

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