Highlights of

Avaya™ Communication Manager

Release 1.3
555-233-783
Issue 2
May 2003

Notice

Every effort was made to ensure that the information in this document was complete and accurate at the time of printing. However, information is subject to change.

Warranty

Avaya Inc. provides a limited warranty on this product. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language as well as information regarding support for this product, while under warranty, is available through the following Web site:
http://www.avaya.com/support

Preventing Toll Fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there may be a risk of toll fraud associated with your system and that, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya Fraud Intervention

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.

How to Get Help

For additional support telephone numbers, go to the Avaya Web site:
http://www.avaya.com/support/

If you are:

- Within the United States, click *Escalation Lists*, which includes escalation phone numbers within the USA.
- Outside the United States, click *Escalation Lists* then click *Global Escalation List*, which includes phone numbers for the regional Centers of Excellence.

Providing Telecommunications Security

Telecommunications security (of voice, data, and/or video communications) is the prevention of any type of intrusion to (that is, either unauthorized or malicious access to or use of) your company's telecommunications equipment by some party.

Your company's "telecommunications equipment" includes both this Avaya product and any other voice/data/video equipment that could be accessed via this Avaya product (that is, "networked equipment").

An "outside party" is anyone who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf. Whereas, a "malicious party" is anyone (including someone who may be otherwise authorized) who accesses your telecommunications equipment with either malicious or mischievous intent.

Such intrusions may be either to/through synchronous (time-multiplexed and/or circuit-based) or asynchronous (character-, message-, or packet-based) equipment or interfaces for reasons of:

- Utilization (of capabilities special to the accessed equipment)
- Theft (such as, of intellectual property, financial assets, or toll facility access)
- Eavesdropping (privacy invasions to humans)
- Mischief (troubling, but apparently innocuous, tampering)
- Harm (such as harmful tampering, data loss or alteration, regardless of motive or intent)

Be aware that there may be a risk of unauthorized intrusions associated with your system and/or its networked equipment. Also realize that, if such an intrusion should occur, it could result in a variety of losses to your company (including but not limited to, human/data privacy, intellectual property, material assets, financial resources, labor costs, and/or legal costs).

Responsibility for Your Company's Telecommunications Security

The final responsibility for securing both this system and its networked equipment rests with you - Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:

- Installation documents
- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

To prevent intrusions to your telecommunications equipment, you and your peers should carefully program and configure:

- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products

TCP/IP Facilities

Customers may experience differences in product performance, reliability and security depending upon network configurations/design and topologies, even when the product performs as warranted.

Standards Compliance

Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment will be the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. could void the user's authority to operate this equipment.

Product Safety Standards

This product complies with and conforms to the following international Product Safety standards as applicable:

Safety of Information Technology Equipment, IEC 60950, 3rd Edition including all relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE) CB-96A.

Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-00 / UL 60950, 3rd Edition

Safety Requirements for Customer Equipment, ACA Technical Standard (TS) 001 - 1997

One or more of the following Mexican national standards, as applicable: NOM 001 SCFI 1993, NOM SCFI 016 1993, NOM 019 SCFI 1998

The equipment described in this document may contain Class 1 LASER Device(s). These devices comply with the following standards:

- EN 60825-1, Edition 1.1, 1998-01
- 21 CFR 1040.10 and CFR 1040.11.

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

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.

For the G700 Media Gateway:

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. 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. However, there is no guarantee that radio interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- **Reorient or relocate the receiving antenna.**
- **Increase the separation between the equipment and receiver.**
- **Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.**
- **Consult the dealer or an experienced radio/TV technician for help.**

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.

For MCC1, SCC1, G600, and CMC1 Media Gateways:
This equipment complies with Part 68 of the FCC rules. On the rear 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 the G700 Media Gateway:
This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. Located prominently on 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.

The REN is used to determine the quantity of devices which 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, G600, and CMC1 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Off/On 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
120A3 channel service unit	04DU9-DN	6.0Y	RJ48C

For the G700 Media Gateway:

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Ground Start CO trunk	02GS2	0.5A	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

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

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

For the G700 Media Gateway:
This Class B digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe B 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

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

For the G700 Media Gateway:
This is a Class B 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
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Haverhill, MA 01835 USA
Attention: Avaya Account Management

E-mail: totalware@gwsmail.com

For the most current versions of documentation, go to the Avaya Web site:
http://www.avaya.com/support/

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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.

This book describes the new and changed features and enhancements available with the most recent release of Communication Manager (that is, release 1.3) running on any of the following:

- An Avaya DEFINITY Server.
- An Avaya S8300 Media Server with an Avaya G700 Media Gateway.
- An Avaya S8700 Media Server with either an Avaya G600 Media Gateway (for IP Connect configurations), or with an MCC1 or SCC1 Media Gateway (for Multi-Connect configurations).

- An Avaya S8700 Media Server configured to control a remote Avaya G700 Media Gateway. Typically, the G700 media gateway contains an Avaya S8300 Media Server configured as a Local Survivable Processor.

This document also contains information about prior releases of Communication Manager. In the first two chapters of this book, information about this and prior releases of Communication Manager is divided by the release number as follows:

- Release 1.3
- Release 1.2
- Release 1.1.2

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 releases of Communication Manager.
- Hardware — describes hardware that is introduced or changed with the most recent releases of Communication Manager.
- New and changed administration forms — provides information about new administration forms, and changes to existing forms due to the most recent releases of Communication Manager.
- New and changed commands — provides information about non-administration commands (such as display, list, or status commands) that are new or have changed for the most recent releases of Communication Manager.

Trademarks

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 Customer Support 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 Product Documentation search 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.

Call: Avaya Publications Center
Voice: 1-800-457-1235 or 1-207-866-6701
Fax: 1-800-457-1764 or 1-207-626-7269

NOTE:
Customers outside the United States should use +1-410-568-3680 (voice) and +1-410-891-0207 (fax).

Write: Globalware Solutions
Attn: Avaya Account Management
200 Ward Hill Ave
Haverhill, MA 01835 USA

E-mail: totalware@gwsmail.com

Order: Document No. 555-233-783, Issue 2, May 2003

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How to comment on this document

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.

- If you are within the United States, go to the Avaya Web site at http://www.avaya.com/support/ for support telephone numbers. Click *Escalation Lists*, which includes escalation phone numbers within specific regions of the United States.
- For all international resources, contact your local Avaya authorized dealer for any additional help and questions.

Highlights

1

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 1.3 features and enhancements

Avaya™ Communication Manager, Release 1.3, includes the following general telephony and system-wide features and enhancements.

Agent-loginID skill pair increase

Since the LINUX platform supports 20,000 administered agent-loginIDs, the administered agent-loginID skill pairs has been increased from 65,000 to 180,000.

With this enhancement, customers could administer an average of 9 skills per agent for the 20,000 agent-loginIDs (180,000/20,000). Customers could also administer 9,000 agents with 20 skills each (180,000/20). The number of skill pairs is administered on the `DISPLAY CAPACITY SAT` form using the `Administered Logical Agent-Skill Pairs` field.

NOTE:
This capacity increase applies only to the S8700 Media Server and other configurations that have the S8700 capacities.

Alphanumeric field designation

In addition to numeric designations for key system lists and groups of related information, the system administrator can specify alphanumeric designations, 0-15 characters in length, for the following:

- abbreviated dial lists
- abbreviated dial groups
- call pickup groups
- call routing patterns

Announcement sources for the G700 Media Gateway

This feature provides an announcement source for each G700 Media Gateway registered to either an S8300 or S8700 server.

With this feature, the S8700 Media Server supports 10 integrated announcement boards (TN750, TN2501, CWY1), plus an additional 250 G700 announcement sources (for a total of 260). The S8300 Media Server supports 50 G700 announcement sources.

The S8300 does not support standard port networks and TN-type boards. Also, the software resources for integrated boards and G700 sources are separated. The G700 announcement sources are counted separately towards its own limit of 50 on the S8300, and 250 on the S8700.

ARS/AAR dialing without FAC

The Automatic Route Selection (ARS) version of this feature allows users to place calls by dialing the full public-network numbers without first having to dial a Feature Access Code (FAC) — such as the number "9" to access an outside line. The system recognizes the call as an ARS call and uses the ARS digit analysis and digit conversion tables to manipulate the digits to route the call.

The Automatic Alternate Routing (AAR) version of this feature is similar except that the call is routed as an AAR call and therefore uses the AAR digit analysis and digit conversion tables.

AUDIX one-step recording

Users can record conversations by pressing a single button. This feature uses AUDIX as the recording device. This feature is not available with INTUITY AUDIX through Mode Codes or remote AUDIX. Only one AUDIX recording is allowed for each call.

NOTE:
Certain countries, states, and localities have laws or regulations regarding the recording of conversations. Such laws or regulations should be consulted and understood *before* choosing options for the new "Apply Ready Indication Tone To Which Parties In The Call" and "Interval For Applying Periodic Alerting Tone" fields.

A feature button named `audix-rec` is used for this feature, and is added to the `Station` form. The button is available for all stations that have administrable feature buttons. When administered, the button also requires a hunt group extension number (for the AUDIX extension number).

NOTE:
Attendant consoles do not have this button.

To record a conversation when a call is in process, press the `audix-rec` button. When you push the button, the LED light for the feature button begins to flash. After about 4–6 seconds, internal users who are participating in the call will notice that the telephone display changes to `CONFERENCE`. After a few seconds, the LED light on the telephone that initiated the recording is illuminated steadily. This indicates that the AUDIX recording facility is ready and begins to record the conversation.

The internal users on the same switch with the display equipment can notice that the number of parties in the call increases by 1. At this point, depending on the administration, a ready indication tone will play to all the parties in the call, the initiator only, or none of the parties.

After enough information has been recorded, the initiator can then stop the recording by pressing the `audix-rec` button a second time when the LED light is illuminated. The LED light for the feature button on the initiator's station extinguishes. The internal users with the display equipment can again notice that the number of parties in the call decreases by 1. The call remains active.

The `Interval For Applying Periodic Alerting Tone` field is used to allow the switch administrator to choose an interval to play an alerting tone to all the parties on the call during recording. Values are 0–60, and the default is 15. This means, if the default value is used, that all parties on the call hear an alerting tone every 15 seconds that indicates the conversation is being recorded. If the value for the field is 0, then no periodic tone is played during recording.

Avaya Extension to Cellular enhancements

The following enhancements to the Avaya™ Extension to Cellular feature require Release 1.3 of Communication Manager.

Feature status button

Extension to Cellular users can activate, deactivate, and suspend Extension to Cellular service by using an administered Extension to Cellular feature status button. The Extension to Cellular button remains lit when service is enabled, off when service is disabled, and flashes at the inverted wink rate when service is suspended through the optional timer. The Extension to Cellular feature button is available on telephones which support administrable feature buttons.

An optional timer can be included with an administered Extension to Cellular feature button, allowing the user to temporarily disable Extension to Cellular service for one hour.

The timer feature can only be administered from the station form. When administered, it can be activated from the configured feature button on the desk set. When activated, this feature disables incoming calls to the supported cell/wireless device for 60 minutes. The user can enable and disable Extension to Cellular service using the feature button on the desk set.

The number of button pushes required depends upon whether the timer has been administered.

- If the timer is administered and if Extension to Cellular is currently enabled, the first button push disables the Extension to Cellular feature, the second button push changes the timer state, and the third button push enables Extension to Cellular.
- If timer is not administered and if Extension to Cellular is currently enabled, the first button push disables the Extension to Cellular feature, and the second button push enables Extension to Cellular.

Call classification

Extension to Cellular call filtering uses the same criteria for classifying a call as external or internal as the call coverage feature.

Avaya Installation Wizard (AIW) enhancements

NOTE:
The Avaya Installation Wizard (AIW) is only supported with the S8300 Media Server.

- Provide an electronic pre-installation worksheet to automate the task of importing selected pre-installation data. This capability is focused on importing IP address information.
- Provide a customizable template to allow for the selective customization of defaulted translation data.
- Support for Japan, United Kingdom, and France, including system and trunk level parameters. May be extended to Australia and other countries prior to the next release of Communication Manager.
- Support configuration of the IA770 module.
- Support the system level and trunk level parameters needed for use in the targeted G-11 countries.
- Support configuration of IP trunks.
- Support trunk diagnostics.
- Support IP address configuration of distributed G700 Media Gateways through the Gateway Installation Wizard (GIW).

Updates to AIW are available on the Web, and are not necessarily linked to any software release of Communication Manager. The latest version of AIW can be downloaded from **http://support.avaya.com/avayaiw**.

Gateway Installation Wizard (GIW)

The Gateway Installation Wizard (GIW) is a separate application that assists in installing and configuring G700 Media Gateways.

NOTE:
The Gateway Installation Wizard (GIW) is only supported with the S8300 Media Server.

- Laptop-based application for configuring G700 Media Gateways.
- Configures IP information for Media Gateway components in a stack.
- Connects to the serial console port.

Updates to GIW are available on the Web, and are not necessarily linked to any software release of Communication Manager. The latest version of GIW can be downloaded from **http://support.avaya.com/avayaiw**.

Call Detail Recording (CDR) display of physical extension

For Expert Agent Selection (EAS) agent-originated calls, if the `Record Agent ID on Outgoing?` field on the `CDR SYSTEM PARAMETERS` form is set to **y** (the default value), then the agent ID is used for outgoing calls.

If the `Record Agent ID on Outgoing?` field on the `CDR SYSTEM PARAMETERS` form is set to **n**, the physical extension is used.

Call redirection intervals

Communication Manager allows the system administrator to specify the number of times that a call rings at each call coverage point before the call proceeds to the next coverage point.

Center Stage Switch (CSS) separation

S8700 Media Servers in an Avaya MCC1 or SCC1 Media Gateway configuration, with four or more Port Networks (PN), use a Center Stage Switch (CSS) to interconnect the PNs.

The Center Stage Switch (CSS) separation feature allows for the physical separation of redundant S8700 Media Servers, and their corresponding CSS, to improve their survivability. Media Servers and the CSS can be separated up to 6.2 miles (10 km), providing backup and survivability for a communications network in one or more remote locations.

Central Office (CO) support on G700 Media Gateway — Russia

Communication Manager supports central office (CO) trunks in Russia using the G700 Media Gateway.

Increased number of G700 Media Gateways supported on an S8700 Media Server

With Communication Manager, an S8700 Media Server can support up to 250 G700 Media Gateways in an External Communication Controller (ECC) configuration.

Increased tone detection capacity on G700

The G700 Media Gateway has increased tone detection capacity from twelve (12) to fifteen (15) ports. This capability enhances call center applications.

This enhancement increases the total number of call center agents, within a stacked G700 Media Gateway configuration served by an S8300 Media Server, to 450. Within a stacked G700 Media Gateway configuration served by an S8700 Media Server, the total number of call center agents is increased to 1200.

NOTE:
The maximum number of call center agents for each G700 is thirty (30).

IP loss groups

A primary reason to accomplish a loss plan for voice communication systems is the desire to have the received speech and tone loudness at a comfortable listening level. This should be accomplished so that users can listen to each other without being concerned who or where the remote party is, or what kind of telephone equipment each may be using.

A connection with an end-to-end loss (called an Overall Loudness Rating) of 10 dB — which approximates a normal conversation between a talker and listener spaced one meter apart — provides a high degree of satisfaction for the majority of users. Therefore, voice communication standards for end-to-end loss are based on this number.

Communication Manager has now defined two additional loss groups for IP telephony. The purpose of these two loss groups is to set speech and tone loudness separately for IP connections. These loss groups use country-specific gateway loss plans.

The two IP loss groups are:

- Loss Group 18: IPtrunk — loss group for IP trunks (IP Carrier Medium)
- Loss group 19: IPphone — loss group for IP terminals (IP ports)

On an upgrade, if the default for an IP station loss plan is 2, and the IP trunk loss plan is 13, Communication Manager changes the defaults to 19 and 18 respectively.

Leave Word Calling (LWC) — QSIG/DCS

The Leave Word Calling (LWC) feature is extended to enterprise networks with QSIG as the private network protocol, as well as those with DCS.

For enterprise networks that are mixed or in transition from DCS to QSIG, interworking of the LWC feature between the protocols can be provided. LWC also works within a single non-networked switch.

NOTE:
A DCS+ signaling group is needed, but can only be used in networks with 4-digit or 5-digit dial plans.

Link Recovery

IP calls must have an H.248 link between the Avaya G700 Media Gateway and the call controller. The H.248 link between an Avaya server running Communication Manager and the Avaya Media Gateway 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 fails for any reason, the Link Recovery feature preserves any existing calls and attempts to re-establish the original link. If the gateway cannot reconnect to the original server, then Link Recovery automatically attempts to connect with alternate TN799DP (C-LAN) circuit packs within the original server's configuration, or to a Local Spare Processor (LSP).

Link Recovery does not attempt to recover or overcome any network failure that created the link outage. Link Recovery also does not diagnose or repair the network failure that caused the link outage.

Since there is no communication between the Media Gateway and call controller during a link outage, button depressions are not recognized, feature access does not work, nor does any other type of call handling. Essentially, the system is unresponsive to stimulus until the H.248 link is restored. This might be the only indication that link recovery is in process.

⚠ CAUTION:

If an administrator attempts to add a station (phone) to a gateway while that gateway is in Link Recovery, that station is not put into service when the gateway comes back. To complete the addition of the station if this happens, perform a `busyout/release` *command on that station when the gateway comes back into service.*

Local announcements on the G700 Media Gateway

G700 local announcements — also known as virtual voice announcements over LAN (or virtual VAL) — allows twenty minutes total announcement time with fifteen playback channels with Communication Manager.

Avaya voice announcement over LAN (VAL) Manager is used to manage local announcements on the G700 Media Gateway.

Local Survivable Processor (LSP) enhancements

LSP supports up to fifty G700 Media Gateways

Communication Manager allows for a single LSP to support up to fifty (50) G700 Media Gateways in an S8700 Media Server External Communication Controller (ECC) configuration.

Support for ten LSPs on S8300 Media Servers

On S8300 Media Server configurations, Communication Manager supports up to ten (10) LSPs.

Support for up to fifty LSPs

Customers (clients) can have up to fifty (50) LSPs in their system configuration.

Media Encryption

Media Encryption is the encryption of the audio (voice) portion of a Voice Over IP (VoIP) call. Media Encryption can be used to provide enhanced privacy for VoIP communications that involve exchange of sensitive information. Media Encryption is provided between Avaya media gateways and media servers.

Digitally encrypting the audio (voice) portion of a VoIP call 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. One exception is that an IP packet monitor can watch for and capture unencrypted IP packets, and can play back the conversation in real-time or store it for later playback.

Communication Manager encrypts IP packets before they traverse the IP network. An encrypted conversation sounds like white noise or static when played through an IP monitor. End users do not know that a call is encrypted because there are:

- No visual or audible indicators to indicate that the call is encrypted.
- No appreciable voice quality differences between encrypted calls and non-encrypted calls.

⚠ SECURITY ALERT:

Be sure that you understand these important media encryption limitations:

- *Any call that involves a circuit-switched (TDM) endpoint such as a DCP or analog phone is vulnerable to conventional wire-tapping techniques.*
- *Any call that involves an IP endpoint or gateway that does not support encryption can be a potential target for IP monitoring. A common example of this is are IP trunks to 3rd-party vendor switches.*
- *Any party that is not encrypting an IP conference call exposes all parties on the IP call between the unencrypted party and its supporting media processor to monitoring, even though the other IP links are encrypting.*

Media Encryption requirements

The following table lists the supported hardware, software, and firmware requirements for Media Encryption.

NOTE:
Your server must be running Communication Manager, Release 1.3 or later.

Hardware	Minimum Software or Firmware
Avaya IP phones ■ 4606 ■ 4612 ■ 4620 ■ 4624 ■ 4630	Firmware version 1.8 or later
IP Softphone	Software R4V1 with service pack 1 or later
IP SoftConsole™	Software release 1.5 or later
TN2302AP IP Media Processor circuit pack	Firmware version 47 or later

The following equipment is not supported, meaning that Media Encryption does not work with these devices:

- Avaya S8300 Media Server
- Avaya G700 Media Gateway
- Avaya 4602 IP Telephone
- Avaya R300 Remote Office
- Any gateway or IP endpoint that cannot support the Avaya encryption algorithm
- Any wired circuit-switched (TDM) telephone (digital or analog) or trunk

License file requirements

Media Encryption does not work unless the server has a valid License File with Media Encryption enabled. To determine whether Media Encryption is enabled in the current License File:

1. At the SAT type **display system-parameters customer-options** and press ENTER to display the `OPTIONAL FEATURES` form.
2. Scroll to page 3 and ensure that the `Media Encryption Over IP` field is **y** (Screen 1).

```
display system-parameters customer-options                      Page   3 of 10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                      ISDN Feature Plus? y
            Enable 'dadmin' Login? y           ISDN Network Call Redirection? y
            Enhanced Conferencing? n                       ISDN-BRI Trunks? y
                   Enhanced EC500? y                               ISDN-PRI? y
          Extended Cvg/Fwd Admin? y                   Local Spare Processor? n
     External Device Alarm Admin? y                    Malicious Call Trace? y
  Five Port Networks Max Per MCC? y                 Media Encryption Over IP? y
               Flexible Billing? y     Mode Code for Centralized Voice Mail? y
   Forced Entry of Account Codes? y
      Global Call Classification? y                  Multifrequency Signaling? y
             Hospitality (Basic)? y Multimedia Appl. Server Interface (MASI)? n
 Hospitality (G3V3 Enhancements)? y         Multimedia Call Handling (Basic)? n
                       IP Trunks? y      Multimedia Call Handling (Enhanced)? n
                                                      Multiple Locations? y
            IP Attendant Consoles? y            Personal Station Access (PSA)? y
                      IP Stations? y
```

Screen 1. Media encryption field on the Optional Features form, page 3

Message Sequence Tracer (MST) enhancements

In the past, it has been difficult to trace messages through the Message Sequence Tracer (MST) tool pertaining to a particular socket because there was no tag in each message distinguishing it from other sockets.

Now, new message formats for outgoing and incoming data includes the socket number/identifier. These new formats use new Type identifiers of 05 and 06. A pair of new formats 07 and 08 have also been created for outgoing and incoming socket control messages on the PROCR ip-interface.

By creating new format types for these new formats, the task of decoding these messages is easier.

The following enhancement was made to the Message Sequence Tracer (MST):

- Signaling messages between Communication Manager and the TN799 CLAN can now be traced for better diagnostics during network outages.
  - — Add processor TN799 CLAN socket information to the MST trace in order to help developers debug socket problems.
  - — Enhance MST to include the socket number in socket data.
  - — Add TN799 CLAN board ID to CLAN MST IP socket trace messages.

NAT with shuffling

Communication Manager allows IP endpoints to shuffle if they are behind a Network Address Translation (NAT) device in an IP network. Communication Manager supports IP direct calls (a call that has been shuffled) between two endpoints that are translated through a NAT device.

“Shuffling” means rerouting the audio channel connecting two IP endpoints. After shuffling, the audio which previously was carried in a mixed connection of IP signaling and TDM bus signaling, goes directly through the LAN or WAN between the two IP endpoints. Shuffling also can mean reversing this process if an endpoint requests a resource to support a feature, such as conferencing that requires the TDM bus.

NOTE:
Network Address Translation (NAT) is a method to address the shortage of IP V4 addresses by allowing globally registered IP addresses to be reused by native networks.

This enhancement works with static one-to-one NAT. It does not facilitate Port Address Translation (PAT), also known as Network Address Port Translation (NAPT). This enhancement does not work with many-to-one NAT.

Network Call Redirection (NCR) 2-B channel transfer

This enhancement adds support for the 2-B Channel Transfer PSTN network transfer protocols to the Network Call Redirection (NCR) feature. The protocols that are supported are:

- Telcordia TBCT (offered by local and inter-exchange PSTNs with Lucent 5Ess or Nortel DMS100 switches in US or Canada)
- 1998 ANSI Explicit Call Transfer (ECT) for future use.

Another form of network transfer is where the PBX sets up the second leg of the call, and asks the network to merge the incoming call with the outgoing call (the 2B- channels) and drops the trunks to the PBX.

No hold conference

This feature allows a user to automatically add another party to a conference call while continuing the conversation of the existing call. The new party is automatically entered into the conversation as soon as the call is answered. An optional tone can be provided prior to the party being added to the call.

NOTE:
An emergency call is not allowed to be transferred or merged into the traditional conference. The EMRG_CALL FAC was deleted from the allowable FAC list of the no hold conference feature.

After dialing is complete, if the No Hold Conference is not answered within the time specified in an administered “timeout” field, the No Hold Conference call is deactivated.

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 **login**s).

Personalized labels for the Abbreviated Dialing (AD) System list

An administrator can type personalized labels for the Abbreviated Dialing (AD) System list entries. Whenever a 2420 DCP telephone, a 4620 IP telephone, or a 4630 IP telephone, has a feature button that is administered as an entry in the AD System list, the feature button label that is downloaded to the telephone set is the personalized label.

These personalized labels can be administered in the standard supported languages (English, French, Italian, Spanish, and a user-defined language). If a personalized label has not been administered for the AD system list entry, the feature button label that is downloaded to the phone is AD*nn*, where *nn* is the abbreviated dialing number.

NOTE:
This enhancement applies only to the AD System list.

Posted messages

In most situations, after a few rings when no one answers a call, the calling party usually hears an announcement saying that the called party is not available and to please leave a message. At this point, the calling party has no clue when the called party would return the call.

The posted messages feature provides Communication Manager users with the capability of indicating the reason of their unavailability to calling parties. The system provides 30 messages for a user to choose from, such as “on vacation,” or “at lunch.” Of the 30 messages, 15 messages are fixed system messages, and the remaining 15 messages are administrable (custom messages). After a user has chosen one of the messages and thus activated the feature, the message is immediately sent to calling parties who have terminal displays.

The system provides two ways to activate/deactivate this feature: using button pushes and feature access codes. The system allows users to use the feature access codes from their own display telephone, from another station/attendant, or from a remote access trunk.

Resource Reservation Protocol (RSVP)

Resource Reservation Protocol (RSVP) is a QoS signaling protocol. RSVP provides a means of specifying the requirements of IP packet flow, and determining if the intervening network can provide the resources to protect that flow through a process called “admission control.”

RSVP protection of VoIP audio streams on WANs and other links that are susceptible to congestion can safeguard the quality of VoIP calls already in progress.

- IP phones or gateways request the network routers to reserve bandwidth.
- The routers act upon the request to allocate bandwidth according to the QoS request.
- When the bandwidth is reserved, the call is protected against other network traffic in a loaded or congested network, thereby ensuring good voice quality.

Administrators can now configure RSVP settings in Communication Manager. When the `RSVP enable` field in the `IP NETWORK REGION` form is set to ‘`y`’, the `RSVP Reservation Parameters` appear.

Separation of Bearer and Signaling (SBS)

The Separation of Bearer and Signaling (SBS) feature provides a low cost virtual private network with high voice quality for customers who cannot afford private leased lines. SBS provides a DCS+ VPN replacement for customers needing Dial Plan Expansion (DPE) functionality.

NOTE:
DCS does not work with six-digit or seven-digit dial plans. Although QSIG does work with six-digit and seven-digit dial plans, QSIG does not work over VPNs.

The SBS feature supports:

- QSIG private networking signaling over a low cost IP network.
- Voice (bearer) calls over the public switched network.
- Association between QSIG feature signaling information and each voice call.

You must always use AAR/ARS/UDP to originate an SBS call. You cannot use a Trunk Access Code / Dial Access Code to originate an SBS call.

Proper administration and configuration is required for SBS calls to work correctly. This includes:

- New fields in the `SYSTEM-PARAMETERS FEATURES` form, a new field on the `TRUNK GROUP` form, and a new `Station` type called an `SBS Extension` (an extension number without hardware assigned to it that is used to associate the separate bearer and signaling calls).
- Customers must allocate a sufficient number of SBS extensions based on expected SBS traffic volume. The same applies to SBS trunk group members.
- Each administered SBS extension must correspond to a DID/DDI number obtained from a Local Service Provider (LSP).

  NOTE:
  Obtaining a DID/DDI number for each SBS extension is not necessary if the Feature Plus Pseudo DID feature is available. The remainder of this document assumes the use of real (service-provider obtained) DID/DDI numbers for the SBS extensions.

- In remote office configurations or other remote gateway configurations with limited direct network access, these DID/DDI numbers should be obtained from a service provider that is local to the controlling gateway server, not local to the remote office/gateway. This eliminates excessive traffic through the remote office/gateway to its controlling gateway server.
- The `ISDN PUBLIC-UNKNOWN NUMBERING` form must be correctly administered to map every SBS extension to the corresponding national public network complete number (the DID/DDI number). This public form is used to develop the complete number even if the incoming SBS trunk group numbering format is administered for private numbering.

Support for the Hewlett Packard DL380G2 server

Communication Manager is now supported on Hewlett Packard (HP) DL380G2 servers in an S8700 IP-connect system configuration (an S8700 Media Server with a G600 Media Gateway).

Timed automatic disconnect for outgoing trunk calls

This feature provides the capability to automatically disconnect an outgoing trunk call after an administrable amount of time. The amount of time that can elapse before the trunk is dropped can be specified, and can vary between 2 and 999 minutes. If the timer field is blank (the default value), the feature is disabled and the trunk will not be automatically disconnected.

Timed call disconnection applies to all outgoing trunk calls initiated by a party belonging to a specified Class of Restriction (COR).

The outgoing trunk disconnect timer only affects outgoing public network trunks (CO, DIOD, FX, WATS, and ISDN public-network).

NOTE:
The outgoing trunk disconnect timer should be administered to a value large enough to provide users with adequate response time.

The outgoing trunk disconnect timer does not apply to outgoing trunk calls that are emergency or service calls. Specifically, the outgoing trunk disconnect timer does not apply to calls with ARS call types `alrt`, `emer`, `nsvc`, `op`, `svcl`, `svfl`, `svct`, or `svft`.

The outgoing trunk disconnect timer starts after the outgoing trunk call is answered. The outgoing trunk call is considered answered if:

- the network provides an answer supervision line signal
- an ISDN CONNect message is received
- the Answer Supervision Timeout timer expires
- the call classifier classifies the call as answered
- the Outgoing End of Dial Timer expires

Prior to disconnecting the trunk, warning tones are applied to all parties on the call. The first warning tone occurs when one minute remains on the call. The second warning tone occurs when 30 seconds remain on the call.

Tripwire security

Tripwire is an security program provided on S8300 and S8700 Media Servers running Communication Manager. The specific list of files that Tripwire monitors needs to be determined during design once all administration and configuration files have been identified.

When security violations are detected, Tripwire reports its findings to the security log. These events generate an alarm.

NOTE:
Tripwire normally reports security violations through e-mail. However, by reporting events to the security log, security violations can be immediately acted upon.

Release 1.2 features and enhancements

Avaya™ Communication Manager, Release 1.2, includes the following general telephony and system-wide features and enhancements.

Avaya call center features supported on the Avaya G700 Media Gateway

Full Avaya Call Center functionality is now supported on the G700 Media Gateway with Communication Manager, with either an S8300 Media Server or an Avaya™ S8700 Media Server.

The Avaya S8300 Media Server or S8700 Media Server with the Avaya G700 Media Gateway provides Avaya™ Call Center "Basic" software (included with the Communication Manager) capability and optional Computer Telephony Integration (CTI) as a lower-cost call center solution for small or branch offices. For the latest capacities of supported number of agents and media gateways, please see the capacities document available at **http://www.avaya.com/support**. See "Capacity changes" on page 53 for instructions on how to locate the capacities document.

The Avaya G700 Media Gateway with the Avaya S8300 Media Server supports more robust call center capabilities including Avaya™ Call Center "Deluxe," which supports Avaya Best Service Routing and optional Avaya Virtual Routing, and Avaya™ Call Center "Elite," which features Avaya Expert Agent Selection and services as the foundational software for the optional Avaya Business Advocate and Avaya Dynamic Advocate software.

The call center capabilities found in either optional software package (Deluxe or Elite) allow Avaya Call Center customers to enhance their customer service, help desk, travel, and other operations by providing powerful, integrated call routing via "call vectoring" and resources selection.

Avaya Communication Manager on an S8100 Media Server

The S8100 Media Server now includes the full feature set of Communication Manager. A new optional feature on the S8100 Media Server is the co-resident LAN Gateway for CTI connectivity.

Avaya Extension to Cellular enhancements

New features with Extension to Cellular (requires Release 1.2 or later of Communication Manager) include:

Call Filtering

This feature allows customers to manage cellular phone costs by limiting the calls extended to the cellular network for Extension to Cellular users. Customers can choose to deliver, on a per-user basis, only external calls (from a customer), only internal calls, all calls, or no calls. This feature allows administrators to better manage cell phone recurring expense.

Call Detail Recording (CDR)

Customers can now choose whether or not to keep unique Call Detail Records for Extension to Cellular calls.

Scheduler

This function (integrated with Avaya Unified Communication Center Release 1.0) provides rules-based Time of Day and Day of Week enable/disable of the Extension to Cellular feature accessible from PDA or PC.

Loopback Trunk Elimination

Special loopback tie trunks used to relay XMOBILE calls to a public network trunk. This required two trunks per Extension to Cellular calls. Loopback Trunk Elimination reduces the need for the DS1 and IP loopback configuration as required in earlier releases.

Now ARS and AAR routing of Extension to Cellular calls requiring only one trunk instead of the previous two. Calls to an XMOBILE station can be extended out of the PBX directly over an ISDN trunk connected to the public network. This change simplifies the administration of Extension to Cellular. The end-user functionality is unchanged. The regular ARS or AAR routing tables are used to select the trunk for the Extension to Cellular call.

Avaya IA770 INTUITY Audix Messaging Application (embedded INTUITY 5.1 messaging—Linux)

The IA770 application enhances communications and information exchange within enterprises, helping customers be more successful with call answering and messaging. The IA770 application enables customers to see messages on their PCs, add a voice mail component to an e-mail, and listen to e-mail using voice mail.

IA770 uses the Linux operating system, making it consistent with the G700 Media Gateway's operating system. The distributed architecture is designed for reliability and survivability and is centrally managed for simplicity, efficiency and quick response to help ensure business recovery.

The IA770 application consists of license file-activated software residing on the S8300 Media Server, and a small card that can be installed and upgraded in the field.

The IA770 application is available in two configurations:

- 4 ports, 100 users
- 8 ports, 300 users

The IA770 application includes the new release of INTUITY™ Message Manager Release 5.0. While the system provides text-to-speech capability in U.S. English only, there is no additional charge for initial implementation of any of the 35 available languages for prompts.

IA770 supports INTUITY digital (TCP/IP) and AMIS networking protocols. More extensive networking can be provided with the Avaya™ Interchange.

Using the Web interface, the administrator can perform a system backup and restore of all administered data — announcements, recorded names, greetings — and approximately 50 hours of messages over the local area network (LAN).

The forms are easier to understand and more intuitive, which should cut installation time and lessen the need for training and experience. The IA770 system uses smart defaults rather than requiring every field to be addressed.

For more information on the IA770 messaging application, see the *S8300 and S8700 Library CD*, 555-233-825.

Avaya Installation Wizard (AIW)

The Avaya™ Installation Wizard (AIW) is a tool with Communication Manager for use in new installations (not upgrades) to help reduce complexity, time-to-install, and the cost of installation.

CAUTION:

The Avaya Installation Wizard (AIW) and the web installation that is accessible from the Avaya maintenance Web pages should not be run at the same time. Make sure you complete one process before you start the other process.

The Avaya Installation Wizard delivers the following installation advantages:

- Intuitive user interface with on-line help
- Auto-discovery, where appropriate
- No assumption of external internet connectivity
- Ease of updating to newest software & firmware
- Ability to import customized name & number list
- Complete record of all settings
- Guided process from beginning to end

The Avaya Installation Wizard can guide installers through:

- License file and password file setup
- Media server & media gateway configuration
- Telephony, trunk, and endpoint configuration and installation
- Installation Log File Summary creation

There are several enhancements to the Avaya Installation Wizard (AIW) for Communication Manager:

- The Avaya Installation Wizard now supports a stack of up to 10 G700 Media Gateways. Previously, it supported a stack up to 5 G700 Media Gateways.
- Technicians are now able to load updated Media Module firmware versions from their laptop as part of the Avaya Installation Wizard process. Previously, the new firmware had to be preloaded on the FTP boot directory of the S8300 Media Server for the wizard to facilitate the upgrade.
- Installation of the new BRI Media Module is supported.
- The Avaya Installation Wizard now supports installation of a G700 Media Gateway with a Local Survivable Processor (LSP).
- Remote G700s without an ICC Media Module can be configured using the Avaya Installation Wizard by temporarily installing a spare ICC Media Module in the G700 until the end of the installation process.

For additional information, including a hands-on prototype of the Avaya Installation Wizard, see **http://support.avaya.com/avayaiw**.

Call forwarding of 18-digits

Communication Manager has increased the number of available input digits to forward a call from sixteen to eighteen digits.

If you need to include a feature access code (fac) or dial access code (dac), then those digits count against the total. Any special characters, like pause characters, also count against the total.

Co-resident DLG

In simplest terms, the DEFINITY Local Area Network (LAN) gateway, or DLG, is an application that enables communications between TCP/IP clients and Communication Manager call processing. In more technical terms, the DLG application is software that both routes internetwork messages from one protocol to another (ISDN to TCP/IP) and bridges all ASAI message traffic by way of a TCP/IP tunnel protocol.

In previous configurations, a DEFINITY LAN gateway (DLG) was connected externally on a separate TN801 MAPD circuit pack. The DLG application is packaged internally where it co-resides with the Communication Manager. The internally packaged DLG is referred to as the co-resident DLG.

Co-resident DLG is only available with the S8300 Media Server, while the S8100 Media Server can have either a co-resident DLG or a DLG connected externally on a separate TN801 MAPD circuit pack. The co-resident DLG is not available on the S8700 Media Server, or DEFINITY Servers R, SI, or CSI.

Co-resident DLG provides the functionality of the Adjunct/Switch Application Interface (ASAI) using an ethernet transport instead of a Basic Rate Interface (BRI) transport. In the S8300 Media Server, connectivity is provided through the processor's ethernet.

For more information on co-resident DLG and the G700 Media Gateway, see chapters "DEFINITY LAN Gateway and ASAI-Ethernet," and "Installation and Test for CallVisor ASAI," in the *Avaya MultiVantage CallVisor ASAI Technical Reference*, 555-230-220.

Also see the following documents:

- *DEFINITY Enterprise Communications Server CallVisor ASAI Applications Over MAPD*, 555-230-136
- *Installation for Adjuncts and Peripherals for Avaya™ Communication Manager*, 555-233-116.

Increased distance between multiple S8700 Media Servers

This feature allows the user to geographically separate their S8700 Redundant Media Servers by up to 6.2 miles (10 kilometers), thereby providing improved survivability. The maximum distance of 6.2 miles (10km) separation is dictated by the maximum distance that the transceivers on the DAJ-1 board will allow for the fiber extension without using any repeaters or boosters.

This feature is applicable to S8700 with an MCC1 or SCC1 Media Server (also called “Multi-Connect”), as well as IP Connect systems.

NOTE:
This feature does not provide for separation of duplicated CSS-PNC or duplicated ATM-PNC in a critical reliability system. However, separation of duplicated ATM switches (ATM-PNC) is already widely deployed in the field. Both CSS-PNC and ATM-PNC require that their respective duplicated EI boards (TN570 and TN2238) be housed in the same cabinet.

Katakana character set

Communication Manager supports the katakana character set (Japan) on 4620 IP telephones only. This nine-point character font was designed to allow the 4620 IP telephone to display katakana characters in the user interface as well as in switch-generated messages.

This feature requires 4620 firmware version 1.72 or later to work. You can obtain the latest version of 4620 firmware at no charge by going to the Avaya Web site at **http://www.avaya.com/support/**.

Local announcements on the G700 Media Gateway

G700 local announcements — also known as Virtual Voice Announcements over LAN (or Virtual VAL) — allows twenty minutes total announcement time with fifteen playback channels with Communication Manager.

NOTE:
Avaya Voice Announcement over LAN (VAL) Manager, an application in the Avaya VisAbility Management Suite, is used to manage local announcements on the G700 Media Gateway.

Local music-on-hold

The music on hold feature is now supported on the G700 Media Gateway with Communication Manager. The music source is connected to a port on the MM711 analog media module. Local music-on-hold is part of the call center functionality on the S8300 Media Server.

Local music-on-hold allows one music source. However, if you purchase the multiple music-on-hold (tenant partitioning) feature, you can have up to 100 music sources.

For more information, see the *Installation for Adjuncts and Peripherals for Avaya™ Communication Manager*, 555-233-116. Also see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

Migrations of DEFINITY Server SI and DEFINITY Server R Direct Connect to S8700 Media Server with G600 Media Gateway

You can now migrate your system from a DEFINITY Server SI or a DEFINITY Server R Direct Connect to an S8700 Media Server with G600 Media Gateway (also called "IP Connect").

This is not an upgrade to your existing system, but rather a migration to an S8700 Media Server with G600 Media Gateway. Cabinets cannot be reused in this migration, but most circuit packs can be reused. For more information, contact your local Avaya representative.

Multiple music sources

On an MCC1, SCC1, CMC1, or G600 Media Gateway, this feature allows the customer to provide multiple distinct music sources for use with the call vectoring features, calls placed on hole, calls awaiting pickup, and so on. By purchasing the multiple music-on-hold (also called tenant partitioning) feature, you can have up to 100 music sources.

MultiTech gateway support

Communication Manager supports a voice over IP (VoIP) gateway from MultiTech, a third-party vendor. Any system running Communication Manager can connect and run a MultiTech gateway.

With a 2-port, 4-port, or 8-port MultiTech gateway, Communication Manager offers a cost-effective and survivable VoIP gateway solution within the Communication Manager environment for a client's branch or remote office with fewer then ten stations.

Connecting a MultiTech gateway to your system is just like connecting any H.323 generic IP endpoint -- similar to connecting an IP telephone. Once connected, a customer could then connect analog stations and trunks to the MultiTech gateway.

Punctuation on station displays

On digital telephone displays, Communication Manager can display punctuation to make reading a 6-digit or 7-digit extension easier. The number of digits plus punctuation that can be displayed cannot exceed eight characters.

Punctuation marks that are allowed include:

- hyphen (for example, `xxx-xxxx`)
- period (for example, `xxx.xxxx`)
- space (for example, `xx xx xx`)

Formats for displaying numbers with punctuation are on the dial plan parameter form.

- the default 6-digit extension display format is `xx.xx.xx`
- the default 7-digit extension display format is `xxx-xxxx`

For more information on the dial plan parameter form, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

S8100 Media Server embedded INTUITY AUDIX

This application provides voice, fax, and text messaging, along with text-to-speech and message manager functionality in a single processor mezzanine board on the S8100 Media Server.

Included are Avaya Directory Enabled Management (DEM) and Fax Extended Dialing (FED).

- ADEM provides real time directory-based access to Communication Manager and INTUITY AUDIX.
- FED allows the customer to specify restrictions on the destination numbers, as well as eliminate the need to administer fax number ranges as remote AMIS networking machines. Additionally, FED addresses the entry of international destination numbers by allowing up to 23 digits for fax endpoints.

The INTUITY AUDIX mezzanine card also provides the necessary DSP resources for messaging. This hardware eliminates the need for the INTUITY Map 5P adjunct, usually required for this functionality.

Selective conference mute

Selective conference mute allows a conference call participant, who has a display station, to mute a noisy trunk line. Selective conference mute is also known as far end mute.

Examples of noisy trunk lines that might need to be muted during a conference call are:

- cell phones
- phones that utilize the Music-On-Hold feature
- phones with no mute capabilities

Selective conference mute only applies to trunk lines on the conference call, and not to stations. Only one trunk line on the conference call can be selectively muted at a time. This enhanced conferencing feature can be activated from any display station with a “conf-dsp” button and an “fe-mute” button.

The selective conference mute feature works with any conference established through Communication Manager, either a traditional 3 or 6 party conference, or a Meet-Me conference.

NOTE:
This feature requires that the enhanced conferencing feature be set to **Y** on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form. For more information on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, see the *Administrator’s Guide for Avaya™ Communication Manager*, 555-233-506.

Activating selective conference mute

While on a conference call, a display station user repeatedly presses the “conf-dsp” button to display the noisy trunk line. While the trunk line information is displayed, the display station user then presses the “fe-mute” button to mute the noisy trunk. The conference display is updated to indicate the endpoint is muted.

NOTE:
The person who has the noisy trunk line does not know that the line has been muted.

There are two ways to unmute the line:

- The person whose trunk line was muted can press the # key on the telephone keypad.
- The display station user can repeatedly press the “conf-dsp” button to display the muted party, then press the “fe-mute” button to unmute the line.

SNMP agents on S8700 and S8300 Media Servers

The S8300 and S8700 Media Servers each include a Simple Network Management Protocol (SNMP) agent. The SNMP agents work with a standards-based customer network management system (NMS) to deliver traps (alarms) and set/retrieve variables defined in the Communication Manager Management Information Base (MIB).

NOTE:
Alarms continue to be separately forwarded to the Initialization and Administration System (INADS) for analysis by automated tools and Avaya services personnel. An alarm sub-agent exists to package switch alarms as SNMP traps and forward them to INADS.

VDN override for ASAI messages

This feature provides a VDN option to override the called number in certain ASAI messages for ISDN calls. This applies to CTI applications that require the active VDN extension instead of the called number. This is a new field on page 2 of the VDN form - “VDN Override for ISDN Trunk ASAI Messages (default is **n**).

For calls to VDNs with the option set to y(es), the called number provided will correspond to the active VDN for call instead of the original called number provided in the incoming ISDN SETUP message. This applies to the ASAI call-offered, alerting, queued and connect event messages and the adjunct route-request message.

Release 1.1.2 features and enhancements

Avaya™ Communication Manager, Release 1.1.2, includes the following general telephony and system-wide features and enhancements.

Analog busy automatic callback without flash

This is a feature with Communication Manager for analog stations supporting automatic callback without the user flashing the hook. It will be applied only when the called station is busy and no other coverage path (or call forwarding) has been specified for it. The caller can then enable the automatic callback without flashing the hook or entering the feature access code.

With Analog Busy Automatic Callback without Flash, when callers place calls through an analog station, and the called station is busy and has no coverage path or forwarding, callers hear announcements presenting them with a set of options. Depending on the callers' selection from the announced options list, their calls are then queued to Automatic Callback, routed to an extension, or dropped.

Avaya Extension to Cellular

The Avaya™ Extension to Cellular feature offers users the freedom to work anywhere, anytime, using any type of cellular or wireless phone. With Extension to Cellular, calls to an office number are extended to a cell phone, allowing users to receive work-related calls wherever they are and whenever they need to. Additionally, the cell phone can be administered so that when a user calls into the office, the user's name and office telephone number appear in the caller ID display of the phone being called. When the Extension to Cellular cell phone is administered to send office caller ID, the user also has the option of picking up an ongoing Extension to Cellular cell phone call on the office phone.

The Extension to Cellular cell phone user receives the same features and capabilities for incoming calls as a caller ID-enabled telephone connected directly to any Avaya media server or DEFINITY Server. Extension to Cellular provides this capability, regardless of the cell phone's Cellular Service Provider or the cellular standard in use.

CLAN QoS and CIDR support

The TN799 Control-LAN (CLAN) circuit pack support for both Classless Inter-Domain Routing (CIDR) and Variable Length Subnet Mask (VLSM) provides enhanced flexibility in IP addressing and routing, as well as ensuring customer-network compatibility.

CLAN support for multiple network regions

A CLAN circuit pack now supports multiple IP network regions, providing a lower-cost implementation of VoIP to customers with several IP networks. IP telephones may be registered to any of the network regions the CLAN supports.

Conference/transfer enhancements

Feature enhancements to telephones for the conference, transfer, and hold are easier to use. The enhancements include:

- Meet-me Conference allows up to six parties to be conferenced together with other parties. The Meet-me Conference feature is a special VDN given to end users and is secured through an access code. This provides users their own personal bridge for use at any time, and does not incur conferencing expenses to service providers.
- New display prompts based on users' class of restriction (COR). These display prompts vary depending on the activation of certain conferencing features. Many phrases were added to the Language Translations form in Release 10 of Avaya DEFINITY software (and in later releases) to reflect the enhancements to Conference/Transfer/Hold.
- The ability for a caller to toggle or swap connections to multiple conference parties (alternately placing each called party on soft hold) with the new Toggle/Swap button. The caller can still press the Conference button to conference with all the called parties, or can press Transfer to drop his/her own connection, thereby conferencing only the others (called parties).
- Selective conference party display and drop (or forced release on the attendant console). Repeated presses of the Conference Display button cycle through the display of the names and numbers (if available) of all parties on the call. The caller may drop each party from the conference.

For more information, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

Dial Plan Expansion (DPE)

Communication Manager allows you to expand your dial plan to 6 or 7 digits (from 4-digit or 5-digit dial plans). This affects all extensions, including stations, data modules, agent login IDs, vectors, and so on.

This change increases the total number of extensions that can exist in any dial plan. It also allows Avaya servers to participate in networks that already use 6-digit or 7-digit dial plans — for example, a network of switches made by other vendors.

Customers upgrading to Communication Manager can choose to migrate to the 6-digit or 7-digit dial plan or not. Customers who choose not to migrate now may convert their dial plans at a later date.

NOTE:
Distributed Communications System (DCS) is limited to a dial plan of 4 or 5 digits, so if your dial plan requires 6 or 7 digits, QSIG — which is the generic name for a family of signaling protocols— is required.

Enhanced abbreviated dialing list increase

The S8700 Media Server now supports 20,000 entries within the enhanced abbreviated dialing system list. This second enhanced abbreviated dialing list doubles the capacity to from 10,000 entries to 20,000 entries.

Future increases to the enhanced abbreviated dialing list can be performed easily by increasing the number of lists. Increasing the number of lists increases the overall capacity by multiples of 10,000 entries.

France 25% trunk alarming

The France 25% Trunk Alarming feature changes what is generally a major alarm to a warning alarm. When this feature is activated and 25% or more of the system trunks are out of service, a warning alarm is generated instead of a major alarm.

H.248 Media Gateway control

New support with Communication Manager for the H.248 standard of call-control signaling enables a true client/server architecture between Avaya media servers and Avaya G700 Media Gateways. Among the supported data are signals, events, statistics and properties.

IP serviceability enhancements

CLAN supports new maintenance commands with Communication Manager, which enhance IP serviceability and extend IP administration capabilities. Administrators can now diagnose possible problems with duplicate IP addresses, as well as restore an original IP routing table on a CLAN circuit pack without any service disruption. See Chapter 4, ‘‘New and changed commands’’ for more information.

Location by region

Location by region provides a way to administer location by IP network region with Communication Manager. This allows for the correct date and time information and trunk routing based on the IP network region. Correctly interpreting this regional information is crucial to correctly handling and routing users' calls.

Location by region offers the capability to have an IP phone registered anywhere, and have that IP phone display the correct time and date worldwide. The IP phone can be registered in one network region, but then the IP phone's calls can route over trunks local to the phone. It allows IP telephone users the ability to move from location to location and always have correct display information. Remote users are identified in a network region and location that routes them to correct services or notifies them via announcements, with information appropriate to this jurisdiction remote to that of the Avaya server to which they are registered.

For example, location by region tries to overcome a limitation in the emergency response system. 911 call handling for some IP telephones has had a limitation because there has been no way to pop up screens on the IP phones to let users know why their 911 calls were blocked and advise them on what to do.

Now, you can choose to dedicate one location to handle such "roaming" IP telephones. That special location could have corresponding ARS routing tables that route all 911 calls to a repeating announcement, saying something like "You are too far away from the switch for the [name of the home location]'s public safety office to be able to help you. Please call 911 from a local circuit-switched phone."

Support for Russia DATS/ISDN network

This feature with Communication Manager supports ISDN/DATS trunk networks when the tone generator field is set to 15 (Russia) on the `SYSTEM-PARAMETERS COUNTRY-OPTIONS` form. When the feature is activated, the overlap sending delay and ISDN T302 and T304 timers are modified to support the Russian trunk network.

Time of day clock synchronization

The time of day clock synchronization feature provides users with the capability of synchronizing their Avaya DEFINITY Server or S8000-series Media Server clock(s) with Internet servers that provide Coordinated Universal Time (UTC).

Time of day clock synchronization enables an Avaya server to synchronize its internal clock to UTC time provided by Internet time servers. The Linux or Windows 2000 platforms, running NTP or SNTP software, poll the time servers for the UTC time. UTC time is then converted to the local time of the switch. The platform system clock then provides the synchronized time to the Avaya server.

For more information, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506, plus the documentation or online help for the software products comprising Avaya VisAbility™ Management Suite.

Voice mail retrieval button

Avaya™ Communication Manager supports the voice mail retrieval feature as a fixed feature button on the 2420 DCP and the 4602 IP telephone.

A field, "voice-mail Number: _____" appears on page 3 of the `STATION` form for stations of type 2420 and 4602. The allowed values for this field are identical to the values allowed for an auto dial feature button number. The field is a fixed field allowing entry of up to 16 digits that are auto-dialed to access the user's voice mail system.

- If the number field is blank, the voice mail retrieval button is treated like the "Transfer to Voice Mail" button. If the number field is not blank, the voice mail retrieval button is treated like an auto dial button.

Capacity changes

System capacities have been expanded for these products:

- S8700 Media Servers with G700 Media Gateways in standard or Local Spare Processor (LSP) Configuration
- S8300 Media Server with G700 Media Gateways in standard or Local Spare Processor Configuration
- Number of supported users and trunks on G700

NOTE:
If the link between the remote media gateway and the main server (S8700 or S8300) is broken, the LSP will activate and take over for those endpoints and media gateways that are connected to the main processing server. A software license for each LSP is required to activate the LSP feature.

Please see the capacities document 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 at http://www.avaya.com/support.

To view the system capacity limits:

1. Type the word `capacity` in the Keywords text box, and then click **Search**.
2. Locate the latest version of the system capacities table document, and then click the title to download the information.

Hardware

2

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 1.3 hardware additions

Avaya™ Communication Manager, Release 1.3, includes the following general hardware additions.

3606 wireless VoIP telephone

The 3606 wireless VoIP telephone solution is an IEEE 802.11b standards-based, 2.4 GHz wireless LAN telephone system. Using voice over IP (VoIP) technology, the 3606 wireless VoIP telephone solution provides high quality mobile voice communications throughout the workplace.

The 3606 wireless VoIP telephone solution requires one DCP port per handset and emulates an 8410D desk telephone.

NOTE:
When setting up this telephone, it should be aliased as a 4606 IP telephone.

The 3606 wireless VoIP telephone solution requires four components:

- 3606 wireless telephones
- a SpectraLink Voice Priority (SVP) server
- telephony gateways

an 802.11b wireless LAN with SVP-enabled access points (such as the Avaya™ AP-1, AP-2, or AP-3 SVP-enabled access point).

4620SW and 4630SW telephones

Similar to the 4620 and 4630 IP telephones, respectively, both of these two model revisions include the following:

- Support IEEE powering
- Include an internal ethernet switch (indicated by the “SW” at the end of the model number)
- CSIPR Class B

All features associated with the 4620 IP telephone also apply to the 4620SW IP telephone. All features associated with the 4630 IP telephone also apply to the 4630SW IP telephone.

Avaya IP softphone for pocket PC

Avaya™ IP softphone for pocket PC extends the level of Communication Manager services. This feature turns a hand-held personal digital assistant (PDA) into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PDAs.

Release 1.2 hardware additions

Avaya™ Communication Manager, Release 1.2, includes the following general hardware additions.

3410 wireless telephone

The 3410 wireless telephone solution consists of the 3410 handsets, line-powered four-channel wireless base stations, and a Master Control Unit (MCU). The 3410 wireless telephone solution requires one DCP port per handset, has up to six line appearances and twelve feature buttons, and emulates an 8410D desk telephone.

Then MCU has two scalable configurations:

	Link 3000	Link 150
Wireless phones	3200	64
Base stations	1000	16
Simultaneous calls	1600	32
Coverage (in million sq. ft.)	100	1.5

The 3410 wireless telephone solution uses 902-928 MHz spread spectrum frequency hopping radio technology with a high level of integration with enterprise telephone switching to provide an advanced wireless telephone system.

International analog media module

Media modules allow for traditional interfacing of service provider network access solutions such as T1/E1, International ISDN BRI, Loop Start/Ground Start Trunks, as well as connections to TDM-based endpoints such as DCP digital phones, analog phones, and tip/ring devices.

Media modules are optional components that can be mixed in any combination inside G700 Media Gateway devices. All media modules plug into the front of a G700 Media Gateway.

The MM711 analog media module is used only with the G700 Media Gateway. The MM711 analog media module supports eight analog interfaces allowing the connectivity of Loop Start, Ground Start, Analog DID trunks, and 2-wire analog Outgoing CAMA E911 trunks for connectivity to the PSTN. As well, the analog combo media module allows connectivity of analog tip/ring devices such as single line telephones, modems, or group 3 fax machines. Each port may be configured as either a trunk interface or a station interface.

Also included is support for caller ID signaling, ring voltage generation for a variety of international frequencies and cadences, telephone presence detection, and administrable line termination styles.

BRI media module

The MM720 BRI media module is used only with the G700 Media Gateway. The MM720 international ISDN BRI media module contains 8 BRI ports that interface to the central office at the ISDN T reference point. Information is communicated through two 64 Kb/s channels known as B1 and B2 and over a 16Kb/s channel called the D channel. The D channel is used for signaling. Channel B1 and B2 can be circuit switched simultaneously. The D channels are circuit switched from the PPE to the NCE through concentration highway B, and then routed to the TDM bus occupying one timeslot for all 8 D channels. The circuit switched connections have an m-law or A-law option for voice operation, and operate as 64Kb/s clear channels when in the data mode.

Although it may be desirable to switch both B channels together as a 128 Kb/s wide-band channel, the S/T interface transceiver does not support this. Both B channels may be circuit switched at the same time, but they are separate channels and no byte-ordering relationship between them is guaranteed. This media module does not support wide-band channels.

NOTE:
The BRI media module does not support BRI stations, only BRI trunks.

Five EPN maximum in MCC1 Media Gateways

NOTE:
This feature is for MCC1 Media Gateways when used with an S8700 Media Server or DEFINITY® Server R configurations only.

This optional software feature allows customers that require high calling traffic capacities to have from two to five expansion port networks (EPN) in a single MCC1 Media Gateway. Until now, only two port networks (PN) were available unless a specialized cable was purchased from Avaya and workarounds were performed in software administration to make additional carriers function as EPNs.

When this feature is activated, Communication Manager enables administration of up to five carriers as EPNs and no custom cables are necessary. This means that the full bandwidth of the TDM bus is available to each carrier while still enabling the customer to have the footprint of an MCC1 Media Gateway. This is especially appealing to call centers without IPSI/PNC duplication, where systems can be quite large and heavily utilized.

The hardware limitation of the MCC1 Media Gateway is five port carriers. All five can be expansion port carriers, although traffic considerations may dictate some number less than that which is optimum. For example, a customer may choose to have three EPN carriers and two standard port carriers.

There is only one maintenance board, which is placed in carrier A. This is the only maintenance board in the cabinet.

NOTE:
Only two PNs are physically supported in S8700 Media Server IPSI-enabled systems when high/critical reliability options are desired. Only two PNs are physically supported in DEFINITY Server R systems when critical/ATM Network Duplication reliability is desired.

The following table shows the number of port networks allowed in an MCC1 Media Gateway.

Number of Port Networks allowed in an MCC1 Media Gateway

	DEFINITY Server R				S8700 Media Server (all IPSI-enabled PNs)		
	Std	High	Critical	ATM Net Dup	Duplex	High	Critical
1 PN	Yes	Yes	Yes	Yes	Yes	Yes	Yes
2 PN	Yes	Yes	Yes	Yes	Yes	Yes	Yes
3 PN	Yes	Yes	No	No	Yes	No	No
4 PN	Yes	Yes	No	No	Yes	No	No
5 PN	Yes	Yes	No	No	Yes	No	No

For more information on this feature, see your local Avaya representative.

Release 1.1.2 hardware additions

Avaya™ Communication Manager, Release 1.1.2, includes the following general hardware additions.

Avaya Media Servers and Media Gateways

The following hardware products are new components of Communication Manager:

- Avaya™ S8300 Media Server with an Avaya™ G700 Media Gateway
- Avaya™ S8700 Media Server for IP Connect Configurations
- Avaya™ S8700 Media Server for Multi-Connect Configurations
- Avaya™ S8700 Media Server controlling a remote G700 media gateway (with or without an Avaya™ S8300 Media Server configured as an LSP).

Avaya S8300 Media Server and Avaya G700 Media Gateway

The following hardware products are new components with Communication Manager:

- Avaya™ S8300 Media Server with an Avaya™ G700 Media Gateway, or
- An Avaya™ G700 Media Gateway (sold separately).

Avaya S8300 Media Server

The Avaya S8300 Media Server and G700 Media Gateway combination seamlessly delivers a business's voice, fax, and messaging capabilities over an IP network. This unique combination converges the power of the Communication Manager feature set with the power of distributed switching from the Avaya Cajun™ P330 line of network switches.

Several elements comprise an S8300 Media Server and G700 Media Gateway combination:

- A G700 Media Gateway is always required. It can host an S8300 Media Server or various other media modules depending on the telephony needs at a particular location. Key components include the Cajun stack processor, Media Gateway Processor (MGP), and Voice over IP (VoIP) engine on the MGP board.

- The S8300 Media Server is a special type of media module. It supports the Communication Manager that provides call-processing capabilities for the system. The S8300 Media Server can be configured as the primary call controller, or as a Local Survivable Processor (LSP) standby server for either an S8700 Media Server or for another S8300 Media Server in the configuration.
- Communication Manager provides the call processing and telephony features. It resides on the S8300 Media Server, or on a remote S8700 Media Server if the G700 Media Gateway does not contain an S8300 Media Server.

Each of these components must be correctly configured in order to bring a new system into service. The different components also need ongoing administration and maintenance in order to upgrade or to expand the system, or to diagnose problems if they arise.

Avaya S8700 Media Server configurations

The following hardware products are new components of Communication Manager:

- Avaya™ S8700 Media Server for IP Connect Configurations comprises an Avaya™ S8700 Media Server with an Avaya™ G600 Media Gateway.
- Avaya™ S8700 Media Server for Multi-Connect Configurations comprises an Avaya S8700 Media Server with an MCC1 or SCC1 Media Gateway. The single-carrier cabinets and multi-carrier cabinets are existing Avaya products enhanced for use in these configurations for the new media servers.
- Avaya™ S8700 Media Server also may be configured to control a remote Avaya™ G700 Media Gateway. This configuration also typically features an Avaya™ S8300 Media Server in the G700 Media Gateway, with the S8300 serving as Local Survivable Processor (LSP), rather than primary call controller.

S8700 Media Server with G600 Media Gateway

The S8700 Media Server with G600 Media Gateway (called “IP Connect”) configurations is an all-IP, 19 inch data rack component. The S8700 IP Connect is always comprised of two duplicated S8700 Media Servers running the Linux operating system, at least one Ethernet switch within the customer’s own local area network (LAN) or one provided by Avaya for the customer’s LAN, and up to 64 Port Networks (PN) using G600 Media Gateways.

Each server is backed-up by duplicated Uninterruptible Power Supplies (UPS). It is strongly recommended that the Ethernet switch is also backed up by a UPS.

This duplex reliability scheme is the only supported configuration. Also note that mixing of G600 Media Gateways with traditional Expansion Port Network cabinets, CMC1, SCC1 and MCC1, is not supported.

The S8700 IP Connect provides the advantage of IP connectivity between PNs. Utilizing customer's existing IP infrastructure, the S8700 Media Server for IP Connect configuration saves customers the cost of building a separate telephony network. As an all-IP configuration, traditional forms of bearer network direct connect, Center Stage Switch (CSS) connect, and ATM PN connectivity are not supported. Also, traditional survivability options are not supported such as the Survivable Remote Processor or the ATM WAN Spare Processor.

S8700 IP Connect supports as many as 12,000 IP endpoints and 4,000 traditional endpoints such as DCP, Analog and ISDN. However, DMI Mode 2, Data Modules, and Mode 3 data or BX.25 links are not supported.

The two S8700 Media Servers, commercial servers with Intel Pentium III processors, can be located anywhere in the network and can be physically separated by up to 100 meters of cable distance.

The IP Connect control network is comprised of the customer LAN, and the IP Server interface connectivity via an IP Switch Interface (IPSI) board. The IPSI (TN2312) provides control network connectivity and Tone Clock/Global Call Classifier functionality.

Highlights of the S8700 IP Connect are:

- An S8700 Media Server (always duplicated)
- A G600 Media Gateway
  - — As many as four G600 Media Gateways per PN
  - — A maximum of 64 PNs
- Scalable to as many as 12,000 IP endpoints
- Scalable to as many as 4,000 traditional stations and trunks
- 2 UPSs (one per Server)
- Avaya™ Communication Manager
- Utilization of any customer's IP network
- Leveraging of existing assets such as circuit packs and endpoints.

For more information about the high-level capabilities of S8700 IP Connect, refer to the *Avaya MultiVantage™ Solutions Hardware Guide*.

S8700 Media Server with MCC1 or SCC1 Media Gateway

The Avaya™ S8700 Media Server with MCC1 or SCC1 Media Gateway (called "Multi-Connect") configurations uses a standard microprocessor engine with an Intel processor on a commercial server. It provides the building block for a flexible, highly reliable configuration that meets a variety of customer telephony needs.

The S8700 Multi-Connect converges voice, data, and video and routes it using high-speed connections between analog and digital trunks, data lines connected to host computers, data-entry terminals, personal computers (PCs), and internet addresses. The servers are duplicated in a S8700 Multi-Connect configuration.

The S8700 Multi-Connect uses a Linux platform on an Intel server. It is derived from the current Avaya DEFINITY® processor, has fewer physical components, and provides most of the same features and functionality with increased capacity. The S8700 Multi-Connect separates call control from the bearer network and uses a dedicated local area network (LAN) for transport of the control data.

NOTE:
The call control network MUST be on a dedicated network.

For more information about the high-level capabilities of S8700 Multi-Connect, refer to the *Avaya MultiVantage™ Solutions Hardware Guide.*

Avaya telephones

The following telephones are new components of Communication Manager:

- Avaya™ 2420 DCP telephone
- Avaya™ 4602 IP telephone
- Avaya™ 4620 IP telephone

2420 DCP telephone

The 2420 is a new digital phone with an optional feature expansion module and downloadable information for its call appearance/feature buttons, eliminating the need for paper labels. The button information appears on a screen on the phone. The firmware for the 2420 can be changed via the digital connection to the server running Communication Manager. The 2420 uses icons to indicate the status of call appearances, bridge call appearances and features. The phone maintains a call log with calling party information. The 2420 has a button for headset on/off. The button label information for the 2420 is automatically downloaded to the phone when a link is established between the switch and the phone.

The speakerphone options are 2-way and group listen. The 2420 has a Drop button, a redial button, and a voice mail retrieval button. Eurofont and Katakana are the available fonts for this phone. Labels for the 2420 may be downloaded in English, French, Italian, Spanish and user-defined sets. The 2420 does not support soft keys or dedicated buttons for Next, Previous, or Menu. The 2420 has 24 administrable call appearance/feature buttons, a 7-line by 24-character display, and a headset jack.

For more information, see the Phone Feature description section of the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506, plus the 2420 DCP telephone-related documents on the Avaya Web site at **http://www.avaya.com/support/**.

For more information, see administration documents' phone feature descriptions.

4602 IP telephone

The 4602 is a new IP telephone with two call appearance buttons, a Drop button, a listen-only Speaker button, a redial button, and a button for retrieving voice mail.

The 4602 IP telephone has separate LEDs to indicate the on/off status of the speaker and mute buttons. The 4602 has a 2-line by 24-character display. The 4602 has no administrable feature buttons, 2 fixed call appearance buttons, a one-way speaker or no speaker option, a fixed Drop button, and a fixed voice mail retrieval button.

For more information, see the Phone Feature description section of the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506, plus the 4602 IP telephone-related documents on the Avaya Web site at **http://www.avaya.com/support/**.

4620 IP telephone

The 4620 is a new IP telephone with an optional feature expansion module, downloadable call appearance/feature button information, and built-in features such as speed dial, call log, and Web browsing using the Wireless Markup Language (WML). The 4620 IP phone does not need paper labels. The button information appears on a screen on the phone.

The 4620 ‘uses icons to indicate the status of call appearances, bridge call appearances and features. The phone maintains a call log with calling party and called party information. The 4620 has a local button for headset on/off. The button label information for the 4620 is automatically downloaded to the phone when a link is established between the switch and the phone. There are three speakerphone options on the 4620: none, 1-way and 2-way. Labels on the 4620 can be downloaded in English, French, Italian, Spanish, and user-defined languages.

For more information, see the Phone Feature description section of the *Administrator’s Guide for Avaya™ Communication Manager*, 555-233-506, plus the 4620 IP telephone-related documents on the Avaya Web site at **http://www.avaya.com/support/**.

New and changed administration forms

3

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

New forms

2420 DCP telephone

There are two new forms for the 2420 DCP telephone:

- TFTP-Server
- Display-Messages Button-Labels (language translations)

TFTP Server Configuration

This form allows specification of the TFTP server the Avaya Call Processing engine uses to get download files.

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

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

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

Form 2. TFTP server configuration form

Local Node Name

The local node name must be a valid entry from the `IP NODE NAMES` form. The node must be assigned to a CLAN ip-interface or **`procr`** (processor CLAN).

Valid entries	Usage
1-15 characters	Node name of the CLAN circuit pack.
`procr`	Processor module for the S8300 or S8700 Media Servers.

TFTP Server Node Name

The TFTP server node name must be a valid entry from the `IP NODE NAMES` form.

Valid entries	Usage
1-15 characters	Node name of the TFTP server.

TFTP Server Port

Valid entries	Usage
`1-64,500`	Type a number for the remote TCP port.

File to Retrieve

Valid entries	Usage
	Type the name of the file you are going to retrieve using up to 32 alpha-numeric, case sensitive, characters for identification.

File Status

A display-only field showing Download In Progress, Download Failed, File Not Found, or Download Completed.

File Size

A display-only field showing the number of bytes transferred.

Filename in Memory

A display-only field showing the name of the file currently in ACP memory.

display-messages button-labels (language translations)

This multi-page form allows you to define language translations for both the 4620 IP telephone and the 2420 DCP telephone feature buttons. For more information, see the "Screen Reference" chapter in the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

NOTE:
The items on this multi-page form are listed in alphabetical order (your form may be different from the form shown in the *Administrator's Guide*).

4620 IP telephone

There is one new form for the 4620 IP telephone:

display-messages button-labels (language translations)

This multi-page form allows you to define language translations for both the 4620 IP telephone and the 2420 DCP telephone feature buttons. For more information, see the "Screen Reference" chapter in the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

NOTE:
The items on this multi-page form are listed in alphabetical order (your form may be different from the form shown in the *Administrator's Guide*).

Administering and Enabling IPSI

For IP server interface (IPSI) administration, two new forms were added.

CAUTION:

The following procedure must be executed on all configurations. This includes configurations that are translated using a bulk provisioning tool such as Avaya Site Administration or ProVision. When Communication Manager is first initialized, ***it does not know*** *the primary and secondary control subnet addresses. Submitting the following form causes Communication Manager to read this information from the media server and update itself correctly.*

1. Type **change system-parameters ipserver-interface** and press ENTER.

```
change system parameters ipserver-interface                          Page 1 of 1

                   IP SERVER INTERFACE (IPSI) SYSTEM PARAMETERS

SERVER INFORMATION

                 IPSI Host Name Prefix: vodka
        Primary Control Subnet Address: 198.152.254.  0 *
      Secondary Control Subnet Address: 198.152.255.  0 *

OPTIONS

                     Switch Identifier: A
         IPSI Control of Port Networks: disabled
```

2. Verify that the `Primary Control Subnet Address` and (if equipped) `Secondary Control Subnet Address` fields are correct. The subnet in these fields should match the most significant 3 octets of the ***Server IP address on control network*** entry from the Pre-Installation Network Worksheet. If there is an asterisk (*****) to the right of the `Subnet Address` fields, it means that the call processing software does not contain the subnet information displayed. After verifying the displayed information, submitting this form causes the Communication Manager to be updated with the displayed subnet information.

   NOTE:

   If the information displayed in the `Primary Control Subnet Address` and `Secondary Control Subnet Address` fields is not correct, it must be changed on the servers. Use the **Configure Server** entry on the S8700 Media Server Web interface to change the server configuration. Then return here to perform this step.

3. Verify that the Switch Identifier is set correctly for this installation.

CAUTION:
It is critical that the correct Switch Identifier is entered here before TN2312 IPSI circuit packs are administered in the next procedure.

4. Press ENTER to submit the form.

Add IPSI translations to Communication Manager

NOTE:
Adding IPSI translations is only required if bulk translations, including the IPSI translations, were not entered earlier. However, it is recommended that connectivity to the IPSIs be tested no matter how the translations were entered. See "Test connectivity to IPSI circuit packs" on page 72.

1. Type **add ipserver-interface <Port Network>**.

```
add ipserver-interface 8                                        Page 1 of 1

               IP SERVER INTERFACE (IPSI) ADMINISTRATION - PORT NETWORK 8


PRIMARY IPSI                                          QoS Parameters
------------                                          --------------
 Location:  8AXX                                        Call Control 802.1p: 6
     Host: ipsi-A08a                                  Call Control DiffServ: 46
  DHCP ID: ipsi-A08a


SECONDARY IPSI
--------------
 Location:  8AXX
     Host: ipsi-A08a
  DHCP ID: ipsi-A08a
```

2. Verify that the fields associated with the `Primary IPSI` and `Secondary IPSI` (if equipped) are populated with default data. The `Host` and `DHCP ID` fields are set by the DHCP server.
3. Press ENTER to submit the form.
4. Repeat the **`add ipserver-interface <Port Network>`** for each IPSI controlled Port Network.

Test connectivity to IPSI circuit packs

The following procedure is performed from both ASA and the S8700 Media Server Web Interface while connected to the **active** media server.

1. From ASA type **`list ipserver-interface`** and press ENTER
   - Verify that all ISPI circuit packs are translated.
2. From the S8700 Media Server Web Interface under **`Diagnostics`** click **`Execute Pingall.`** Select **`Other Server(s)`**, **`All IPSIs`**, **`Ethernet switches`** and click **`Execute Pingall.`**
   - Verify that all endpoints respond correctly.

Verify IPSI circuit pack version

1. From the S8700 Media Server Web Interface under **Installation and Upgrades** click `View IPSI Version`. Select **`Query All`** and click the **`View IPSI Version`** button.
   - Verify the firmware release for each TN2312AP IPSI. If upgrade is required, refer to the upgrade procedures in documentation.

Enable control of IPSI

NOTE:
The next procedure will enable the IPSI circuit packs and allow them to control the port networks.

1. Type **`change system-parameters ipserver-interface`** and press ENTER.

```
change system-parameters ipserver-interface                    Page   1 of   1

                IP SERVER INTERFACE (IPSI) SYSTEM PARAMETERS

SERVER INFORMATION

               IPSI Host Name Prefix: vodka
      Primary Control Subnet Address: 198.152.254.  0
    Secondary Control Subnet Address: 198.152.255.  0

OPTIONS

                   Switch Identifier: A
   IPSI Control of Port Networks: enabled
```

2. Set the `IPSI Control of Port Networks` field to **`enabled`**
3. Press ENTER to effect the change.

Co-resident DLG

cti-link

The **cti-link** commands are available only if, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, either the ASAI Link Core Capabilities and/or the Computer Telephony Adjunct Links field is **y**.

Field descriptions for page 1

```
add cti-link next                                              Page 1 of 2
                                CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI
     Port: 1C0501                                                 COR: 1
     Name: ASAI CTI Link 1









BRI OPTIONS
               XID? y     Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

Form 3. CTI Link form when Type field is ASAI or ADJLK

```
add cti-link next                                              Page 1 of 2
                                CTI LINK
 CTI Link: 1
Extension: 40001
     Type: ASAI-IP
                                                                 COR: 1
     Name: ASAI CTI Link 1
```

Form 4. CTI Link form when Type field is ASAI-IP or ADJ-IP

CTI Link

A display-only field indicating the CTI link number.

Valid entries	Usage
1-8	S8100 Media Server with a CMC1 Avaya™ Communication Manager on a DEFINITY CSI DEFINITY G3i
1-16	S8300 Media Server Avaya™ Communication Manager on a DEFINITY R S8700 Multi-Connect

Extension

This field displays the extension for this link.

Type

For each link that you want to add to your system, you must specify the CTI link type.

Valid entries	Usage
ADJLK	For ASAI links not used by co-resident DLG.
ADJ-IP	For ASAI adjunct links used by co-Resident DLG.
ASAI	For ASAI adjunct links not used by co-resident DLG.
ASAI-IP	For ASAI links used by co-Resident DLG.

Port

Appears when the Type field is **ASAI** or **ADJLK**. Either type 7 characters to specify a port, or type an **x**.

Valid entries	Usage
01 through **44** (DEFINITY R)	First and second numbers are the cabinet number
01 through **03** (DEFINITY SI)	
A through **E**	Third character is the carrier

Valid entries	Usage
01 through **20**	Fourth and fifth characters are the slot number
01 through **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH.

Name

Type a name associated with this CTI link.

COR

Type a Class of Restriction (COR) number to select the desired restriction.

BRI options

XID

Appears when the Type field is **ASAI** or **ADJLK**. Used to identify Layer 2 XID testing capability.

MIM Support

Management Information Message Support. A display-only field that appears when the Type field is **ASAI** or **ADJLK**.

Fixed TEI

Appears when the Type field is **ASAI** or **ADJLK**. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the TEI field. For ASAI, type **y**.

CRV Length

Appears when the Type field is **ASAI** or **ADJLK**. Type **1** or **2** to indicate the length of CRV for each interface.

Field descriptions for page 2

```
add cti-link next                                          Page 2 of 2
                              CTI LINK
FEATURE OPTIONS

    Event Minimization? n      Special Character for Restricted Number? n
```

Form 5. CTI-Link form when Type field is ASAI-IP or ADJ-IP

Event Minimization

May be used when event reports normally would be sent on multiple associations, but the adjunct does not need to see more than one. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link.

When enabled, this option allows only a single such event to be sent. The selection of the association on which the event will be sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled). Use the `STATION` form to change this option. The new option settings take effect the next time the ASAI link is activated.

Valid entries	Usage
y/n	Type **y** to control the behavior for that particular link.

Special Character for Restricted Number

Enables an ASAI CTI link to indicate the calling number restricted presentation within an event report. For further information, see *DEFINITY® Enterprise Communications Server CallVisor® ASAI Technical Reference*.

Valid entries	Usage
y/n	When set to **y** and a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then an “*” is appended to the calling party number in the ASAI message.

DLG administration

This form appears if, on the `STATION` form, the `Service Type` field is **`DLG`**. This form includes all clients that are allowed to connect to the virtual DEFINITY LAN Gateway (DLG).

Field descriptions for page 1

The following form appears if the `Service Type` field is **`DLG`** and is administered. This form includes all clients that are allowed to connect to the co-resident DEFINITY LAN Gateway (DLG). This page is always the last page on the `IP SERVICES` form.

```
                          DLG Administration                    Page X of Y

CTI_-Link    Enabled    Client Name         Client Link    Client Status

    1          y        client1                 1            in use
    3          y        client2                 1            idle
    4          y        client1                 2            idle
```

Form 6. DLG Administration

Client Link

The client name on the CTI switch side identifies the logical connection on the client side (IP address and host name) for multiple links and for the corresponding client (CTI link number/client name) that are connected to you.

Valid entries	Usage
`1-X` or blank	Type the number between 1 and the number of CTI links allowed on the system (for instance, 8 or 16).

Client Name

Type the node name of the client adjunct requesting service for the specified CTI link.

Client Status

A display-only field indicating **`idle`** (currently there is no active connection) or **`in use`** (currently there is an active connection for this client link).

CTI Link

A display-only field indicating the link number.

Enabled

Controls whether the co-resident DLG will allow connections for the corresponding client.

Valid entries	Usage
y	Type **y** to access DLG.
n	Will refuse connections for that Client Name/Client Link pair.

Dial plan expansion (DPE)

This development changes significantly the way you administer the Dial Plan and the Uniform Dial Plan. In previous releases, you used the DIAL PLAN RECORD, the SECOND DIGIT TABLE, and the UNIFORM DIAL PLAN form. In Communication Manager and future releases, you use the DIAL PLAN ANALYSIS TABLE, the DIAL PLAN PARAMETERS form, and the UNIFORM DIAL PLAN TABLE for these tasks.

Dial Plan Analysis Table

The DIAL PLAN ANALYSIS TABLE is a new form that replaces both a DIAL PLAN RECORD form and the SECOND DIGIT TABLE. This form allows you to determine the beginning digits and total length for each type of call that your switch needs to interpret.

```
change dialplan analysis                                Page   1 of   3   SPE A
                          DIAL PLAN ANALYSIS TABLE
                                                          Percent Full:    9


      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       3    dac
        20      5    ext
        21      2    fac
        3       6    ext

        4       4    ext
        4       7    ext
        5       7    ext
        6       5    ext
        8       1    fac
        9       5    ext
        *       3    fac
        #       3    fac
```

Form 7. Dial Plan Analysis Table

Percent Full

Displays the percentage (0 to 100) of the system's memory resources that have been allocated for the dial plan that are currently being used.

Dialed String

The dialed string contains the digits that the switch will analyze to determine how to process the call.

Valid entries	Usage
0–9, *****, & **#**	Type up to 2 characters for each call type. ***** and **#** can only be the first digit in a string. **NOTE:** For call type attd, if the total length is 2, the Dialed String must be 2 digits long.

Total Length

Valid entries	Usage
1–2 for attd **1–4 for dac** **1–4 for fac** **1–7 for ext** **2–6 for pext**	Type the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

Call Type

Valid entries	Usage
attd	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.

Valid entries	Usage
dac	Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. Dial access codes can start with any number from 0–9, * or # and can contain up to 4 digits. The system requires that a DAC have the longest total length for a given Dialed String. You can use the DAC to activate or deactivate a switch feature or to seize a trunk from a trunk group, or both. In the first case, the DAC functions as a FAC, in the second as a TAC. For example, you can define the group 300–399 for dial access codes, and allow both FAC and TAC in that range. You can use 4-digit DACs for ordinary trunk access, but they do not work for attendant control of trunk groups, trunk-ID buttons, or DCS, and only the last 3 digits of the codes can be recorded in CDR records. A DAC must be the last item entered in a row when mixed station numbering is used.
ext	Primary extension — Defines extension ranges that can be used on your system. Extension can have a first digit of 0 through 9 and can be 1 – 7 digits in length. Extension cannot have the same first digit as the ARS or AAR feature access code (FAC).
fac	Feature access code only — A FAC can be any number from 1–9 and contain up to 4 digits. You can use * or #, but only as a first digit. It is recommended that a FAC have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems may occur when, for example, 3-digit FACs and 4-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it *must* have the longest total length in the dial plan.

Valid entries	Usage
pext	Prefixed extension — Is made up of a prefix (first digit) that can be a **0**–**9** (* and # not allowed) and an extension number of up to 5 digits in length. The maximum length of a prefix and extension combination is 6 digits. You cannot administer a dial access code with the same first digit as a prefixed extension. The purpose of the prefix is to identify the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits (extension number) are then processed. A prefixed extension allows the use of extensions numbers with any dialed string (the extension length must be specified on the table). The "prefixed extension" cannot have the same dialed string as the ARS or AAR facility access code (FAC). When a dial plan has mixed station numbering, extensions of various lengths (all with the same first digit) are mapped on the Dial Plan Analysis table. The system then employs an inter-digit time-out to ensure that all dialed digits are collected. The inter-digit time-out may add several seconds to the dial time. An alternative to the delay required in the time-out mechanism at the expense of dialing an extra digit is to use prefixed extensions in the dial plan.

Dial plan parameters

The DIAL PLAN PARAMETERS form works with the DIAL PLAN ANALYSIS TABLE to define your system's dial plan.

```
change dialplan parameters                                    Page 1 of 1
                            DIAL PLAN PARAMETERS

                                                  Local Node Number:  2
                                                    ETA Node Number:
                                                ETA Routing Pattern:
          UDP Extension Search Order: local-extensions-first
```

Form 8. Dial Plan Parameters form

Local Node Number

Type a number to identify a specific node in a switch network. This entry must match the DCS switch node number and the CDR node number if they are specified.

Valid entries	Usage
1–63	Type the number of a specific node in a network.
blank	The field may be left blank if automatic restoration, DCS, and CDR are not used.

ETA Node Number

Type the number of the destination switch for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Valid entries	Usage
1 – 999	Type the number of a destination switch.

ETA Routing Pattern

Type the number of the routing pattern to reach the destination switch.

Valid entries	Usage
1 – 254	Type the number of the ETA routing pattern

UDP Extension Search Order

Specifies the first table to search to match a dialed extension.

Valid entries	Usage
local-extensions-first	Search the local Dial Plan first to match a dialed extension.
udp-table-first	Search the UDP tables for an off-switch (UDP) conversion.

Uniform Dial Plan Table

The `Uniform Dialing Plan` field must be **y** on the `SYSTEM-PARAMETERS CUSTOMER-OPTIONS` form before you can administer this table.

The UDP provides a common 3-digit to 7-digit dial plan length — or a combination of extension lengths — that can be shared among a group of switches. Additionally, UDP can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

```
change uniform-dialplan 0                                              Page 1 of 2
                              UNIFORM DIAL PLAN TABLE
                                                       Percent Full:   2

Matching             Insert          Node      Matching             Insert          Node
Pattern   Len Del    Digits Net Conv Num       Pattern   Len Del Digits Net Conv Num
2          4   0     817    aar  n
4          5   1     334    aar  n
43659      5   1     928    aar  y
623        3   3     5380   aar  n
73012      5   1            enp  n    31
74100      5   0     81     ars  y
8          5   0            ext  n
911        3   0            ars  n
```

Form 9. Uniform Dial Plan Table

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used.

Matching Pattern

Valid entries	Usage
0–9 (1 to 7 digits)	Type the number you want the switch to match to dialed numbers.

Len

Valid entries	Usage
3–7	Type the number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the number entered in the `Matching Pattern` field.

Del

Valid entries	Usage
0–3	Type the number of digits to delete before routing the call. This number must be less than or equal to the number entered in the Len field.

Insert Digits

Valid entries	Usage
0-9 (1 to 4 digits)	Type the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.

Net

Type the switch network used to analyze the converted number.

Valid entries	Usage
aar, ars, enp, ext	The converted digit-string will be routed either as an extension number or via its converted AAR address, its converted ARS address, or its ENP node number. If you type **enp**, you must type the ENP node number in the Node Num field. The Insert Digits field must be blank, and Conv must be **n**.

Conv

Valid entries	Usage
y/n	Type **y** to allow additional digit conversion

Node Num

This is the ENP (Extension Number Portability) Node Number.

Valid entries	Usage
1–999	Type the ENP node number.

Posted Messages

display-messages posted-messages

The user-defined language translations for the system messages, as well as the English, Italian, French, Spanish, and user-defined language translations for the custom messages, are accessed though the `display-messages posted-messages` form.

The next six forms show the detailed layout for the posted messages.

- Page 1 shows the 15 system messages.
- Page 2 shows room for 15 additional custom posted messages in English.
- Pages 3-6 show room for 15 additional custom posted messages for Italian, French, Spanish, and user-defined language translations.

NOTE:
The entered English translations for the custom messages are automatically populated on the left side of pages 3-6. This makes it easy to keep message administration consistent.

```
change display-messages posted-messages                          Page   1 of   6
                            SYSTEM POSTED MESSAGES
                       USER DEFINED LANGUAGE TRANSLATIONS

Message  English                          Message  Translation
Number                                    Number
 1.      In Meeting                        1.      ****************************
 2.      Out To Lunch                      2.      ****************************
 3.      Away From Desk                    3.      ****************************
 4.      Do Not Disturb                    4.      ****************************
 5.      Out All Day                       5.      ****************************
 6.      On Vacation                       6.      ****************************
 7.      Gone For The Day                  7.      ****************************
 8.      Out Sick                          8.      ****************************
 9.      On Business Trip                  9.      ****************************
10.      With Client                      10.      ****************************
11.      Working From Home                11.      ****************************
12.      On Leave                         12.      ****************************
13.      Back In 5 Minutes                13.      ****************************
14.      Back In 30 Minutes               14.      ****************************
15.      Back In 1 Hour                   15.      ****************************
```

Form 10. SYSTEM POSTED MESSAGES form, page 1

```
change display-messages posted-messages                          Page   2 of   6
                            CUSTOM POSTED MESSAGES
                             ENGLISH TRANSLATIONS

                                       Message  Translation
                                       Number
                                       16.      ****************************
                                       17.      ****************************
                                       18.      ****************************
                                       19.      ****************************
                                       20.      ****************************
Send Custom Messages Through QSIG? n   21.      ****************************
                                       22.      ****************************
                                       23.      ****************************
                                       24.      ****************************
                                       25.      ****************************
                                       26.      ****************************
                                       27.      ****************************
                                       28.      ****************************
                                       29.      ****************************
                                       30.      ****************************
```

Form 11. SYSTEM POSTED MESSAGES form, page 2

```
change display-messages posted-messages                          Page   3 of   6
                            CUSTOM POSTED MESSAGES
                             ITALIAN TRANSLATIONS

Message  English                       Message  Translation
Number                                 Number
16.                                    16.      ****************************
17.                                    17.      ****************************
18.                                    18.      ****************************
19.                                    19.      ****************************
20.                                    20.      ****************************
21.                                    21.      ****************************
22.                                    22.      ****************************
23.                                    23.      ****************************
24.                                    24.      ****************************
25.                                    25.      ****************************
26.                                    26.      ****************************
27.                                    27.      ****************************
28.                                    28.      ****************************
29.                                    29.      ****************************
30.                                    30.      ****************************
```

Form 12. SYSTEM POSTED MESSAGES form, page 3

```
change display-messages posted-messages                          Page   4 of   6
                            CUSTOM POSTED MESSAGES
                              FRENCH TRANSLATIONS

Message  English                          Message  Translation
Number                                    Number
16.                                       16.      ****************************
17.                                       17.      ****************************
18.                                       18.      ****************************
19.                                       19.      ****************************
20.                                       20.      ****************************
21.                                       21.      ****************************
22.                                       22.      ****************************
23.                                       23.      ****************************
24.                                       24.      ****************************
25.                                       25.      ****************************
26.                                       26.      ****************************
27.                                       27.      ****************************
28.                                       28.      ****************************
29.                                       29.      ****************************
30.                                       30.      ****************************
```

Form 13. SYSTEM POSTED MESSAGES form, page 4

```
change display-messages posted-messages                          Page   5 of   6
                            CUSTOM POSTED MESSAGES
                             SPANISH TRANSLATIONS

Message  English                          Message  Translation
Number                                    Number
16.                                       16.      ****************************
17.                                       17.      ****************************
18.                                       18.      ****************************
19.                                       19.      ****************************
20.                                       20.      ****************************
21.                                       21.      ****************************
22.                                       22.      ****************************
23.                                       23.      ****************************
24.                                       24.      ****************************
25.                                       25.      ****************************
26.                                       26.      ****************************
27.                                       27.      ****************************
28.                                       28.      ****************************
29.                                       29.      ****************************
30.                                       30.      ****************************
```

Form 14. SYSTEM POSTED MESSAGES form, page 5

```
change display-messages posted-messages                          Page   6 of   6
                            CUSTOM POSTED MESSAGES
                        USER DEFINED LANGUAGE TRANSLATIONS

Message  English                          Message  Translation
Number                                    Number
16.                                       16.      ****************************
17.                                       17.      ****************************
18.                                       18.      ****************************
19.                                       19.      ****************************
20.                                       20.      ****************************
21.                                       21.      ****************************
22.                                       22.      ****************************
23.                                       23.      ****************************
24.                                       24.      ****************************
25.                                       25.      ****************************
26.                                       26.      ****************************
27.                                       27.      ****************************
28.                                       28.      ****************************
29.                                       29.      ****************************
30.                                       30.      ****************************
```

Form 15. SYSTEM POSTED MESSAGES form, page 6

Time of Day clock synchronization

For time of day (TOD) clock synchronization of new and enhanced Communication Manager, the new/changed forms are within the Avaya VisAbility™ Management Suite. Please see the documentation and online help for the suite's software products for more information about forms.

Changed forms

NOTE:
Only the forms that have changed appear in this chapter. Only the fields that have changed or have been added on these forms are described. For more complete information on specific form changes, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

2420 DCP telephone

Several forms changed to accommodate the 2420 phone and firmware download:

- Abbreviated Dialing System list form
- Feature Access Code form
- Feature-Related System Parameters form
- Station form
- Terminal Parameters form
- list usage node-name
- list usage ip-address

Abbreviated Dialing System list

This form implements a system abbreviated-dialing list. Only one system list can be assigned and is administered by the System Administrator. The list can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

```
add abbreviated-dialing system                                  Page  1 of 7
                        ABBREVIATED DIALING LIST

                                  SYSTEM LIST
Size (multiple of 5): 100      Privileged? n      Label Language:english
DIAL CODE                                  LABELS FOR 2420/4620 STATIONS
  11:                                    11:*************
  12:                                    12:*************
  13:                                    13:*************
  14:                                    14:*************
  15:                                    15:*************
  16:                                    16:*************
  17:                                    17:*************
  18:                                    18:*************
  19:                                    19:*************
  20:                                    20:*************
  21:                                    21:*************
  22:                                    22:*************
  23:                                    23:*************
  24:                                    24:*************
  25:                                    25:*************
```

Form 16. Abbreviated Dialing System List screen

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only 1 through 5 display initially. If you enter a number greater than 5 in the Size field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Use this only if network response time is more than 30 seconds.

Label Language

This field provides administration of personalized labels on the 2420/4620/4630 telephone sets. If this field is changed to another language, all administered labels in the original language are saved and the labels for the new language are read in and displayed.

Valid entries	Usage
English, **Italian**, **French**, **Spanish**, **user-defined**	

Privileged

Valid entries	Usage
y	Enter **y** if the originating party's class of restriction (COR) is never checked and any number in the list can be dialed.
n	Enter **n** if the COR is to be checked to determine if the number can be dialed.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Form 17 shows the last page of the `ABBREVIATED DIALING SYSTEM` screen when, on the `SYSTEM-PARAMETERS CUSTOMER-OPTIONS` screen, the `A/D Grp/Sys List Dialing Start at 01` field is **n**.

```
add abbreviated-dialing system                              Page  7 of 7
                        ABBREVIATED DIALING LIST

                                SYSTEM LIST
                                       Label Language:english
DIAL CODE                                LABELS FOR 2420/4620 STATIONS
  01:                                 01:*************
  02:                                 02:*************
  03:                                 03:*************
  04:                                 04:*************
  05:                                 05:*************
  06:                                 06:*************
  07:                                 07:*************
  08:                                 08:*************
  09:                                 09:*************
  10:                                 10:*************
```

Form 17. Abbreviated Dialing System List screen

Form 18 shows the last page of the ABBREVIATED DIALING SYSTEM screen when, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS screen, the A/D Grp/Sys List Dialing Start at 01 field is **y**.

```
add abbreviated-dialing system                                  Page  7 of 7
                        ABBREVIATED DIALING LIST

                                  SYSTEM LIST
                                        Label Language:english
DIAL CODE                                 LABELS FOR 2420/4620 STATIONS
  91:                                  91:*************
  92:                                  92:*************
  93:                                  93:*************
  94:                                  94:*************
  95:                                  95:*************
  96:                                  96:*************
  97:                                  97:*************
  98:                                  98:*************
  99:                                  99:*************
  00:                                  00:*************
```

Form 18. Abbreviated Dialing System List screen

Feature Access Code

There is a new Station Firmware Download Access Code field on the FEATURE ACCESS CODE form.

```
 change feature-access-code                                        Page x of x
                        FEATURE ACCESS CODE (FAC)
             Priority Calling Access Code:
                      Program Access Code:
   Refresh Terminal Parameters Access Code:
          Remote Send All Calls Activation:
           Self Station Display Activation:
     Station Firmware Download Access Code:
  Station Security Code Change Access Code:
          Station User Admin of FBI Assign:
```

Form 19. Feature Access Code form

Station Firmware Download Access Code

This field specifies the feature access code used for 2420 DCP station downloads.

Valid entries	Usage
1-4 digit number; * and # can be used for the first digit.	Type the code you want to use for station firmware downloads.

Feature-related system parameters

There are two new fields on this form:

- `Date Format` on 607/2420/4600/6400 terminals
- `On-hook Dial` on 607/2420/4600/6400/8400 terminals

```
change system-parameters features                              Page   9 of 10
                     FEATURE-RELATED SYSTEM PARAMETERS

                 Pull Transfer: n            Update Transferred Ring Pattern? n
         Outpulse Without Tone? y              Wait Answer Supervision Timer? n
         Misoperation Alerting? n               Repetitive Call Waiting Tone? n
    Allow Conference via Flash? y
 Vector Disconnect Timer (min):       Network Feedback During Tone Detection? y
   Hear Zip Tone Following VOA? y    System Updates Time On Station Displays? n
      Intercept Treatment On Failed Trunk Transfers? n
                   Station Tone Forward Disconnect: silence
                           Level Of Tone Detection: precise
          Charge Display Update Frequency (seconds): 30
        Date Format on 607/2420/4600/6400 Terminals: mm/dd/yy
  On-hook Dial on 607/2420/4600/6400/8400 Terminals? n
RECALL TIMING
        Flashhook Interval? y                   Upper Bound (msec): 1000
                                                Lower Bound (msec): 200
                                  Forward Disconnect Timer (msec): 600
ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? n
```

Date Format on 607/2420/4600/6400 Terminals

Valid entries	Usage
mm/dd/yy dd/mm/yy yy/mm/dd	Defines how the date is formatted on the display for 607, 2420, 4600-series, and 6400-series phones.

On-hook Dial on 607/2420/4600/6400/8400 Terminals

Valid entries	Usage
y/n	Type **y** to allow on-hook dialing for 607, 2420, 4600-series, 6400-series, and 8400-series phones.

Station

There is a new entry (2420) for the `Type` field on the `STATION` form.

```
add station 4005                                          Page 1 of 4
                                    STATION

Extension: 4005                        Lock Messages? n       BCC: 0
     Type: 2420                        Security Code: ______   TN: 1
     Port: _____                     Coverage Path 1: ___      COR: 1
     Name: ________________________  Coverage Path 2: ___      COS: 1
                                    Hunt-to-Station: ____

STATION OPTIONS
             Loss Group: _                Personalized Ringing Pattern: 3
            Data Option? n                          Message Lamp Ext: 1014
          Speakerphone: 2-way                 Mute button enabled? y
       Display Language? English                Expansion Module? n
                                              Media Complex Ext:
                                                   IP Softphone? n
                                             Remote Office Phone: n
```

Form 20. Station form

Page 2 of the `STATION` form is unchanged in Communication Manager.

On the 2420DCP telephone, page 3 of the `STATION` form has an additional field: `voice-mail Number`. For more information on this field, see "Voice mail retrieval button" on page 52.

Examples of the third, fourth and fifth pages (if applicable) follow.

```
add station 4005                                          Page 3 of 4
                                    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
 2: call-appr

    voice-mail Number: _________
```

```
add station 4005                                          Page 4 of 4
                                      STATION

FEATURE BUTTON ASSIGNMENTS

 9:
10:
11:
12:
13:
14:
15:
16:
17:
18:
19:
20:
21:
22:
23:
24:
```

If the `Expansion Module` field is **y** on the first page, then a fifth page appears.

```
add station 4005                                          Page 5 of 5
                                      STATION
EXPANSION MODULE BUTTON ASSIGNMENTS

1:                                    13:
2:                                    14:
3:                                    15:
4:                                    16:
5:                                    17:
6:                                    18:
7:                                    19:
8:                                    20:
9:                                    21:
10:                                   22:
11:                                   23:
12:                                   24:
```

Type

For each station that you want to add to your system, you must specify the type of telephone in the `Type` field. This is how you distinguish between the many different types of telephones.

The following table lists the telephones, virtual phones, and personal computers that you can administer on Avaya Media Servers or DEFINITY Servers. To administer telephones that are not in the table, use the `ALIAS STATION` form.

NOTE:
You cannot change an analog phone administered with hardware to a virtual extension if TTI is **y** on the `FEATURE-RELATED SYSTEM PARAMETERS CUSTOMER OPTIONS` form. Contact your Avaya representative for more information.

Table 1. telephones

Telephone type	Model	Administer as
Multiappearance digital	2420	2420
	6402	6402
	6408	6408

Terminal Parameters

The command for accessing this form has changed to include 2420 phones. The full command is now **change terminal parameters 6400/607A1/4600/2420**. The title on the form now also includes the 2420. The fields on the form did not change.

usage-node-name

Usage of the node name tftp-local is displayed as Local Node if it is administered as the Local Node Name on the TFTP-Server form.

```
list usage node-name tftp-local

                                LIST USAGE REPORT
Used By
Processor Channel   Channel Number 4   Destination Node
TFTP Server                           Local Node
```

The node name `tftpserv` is displayed as Server Node Name if it is administered as the TFTP Server Node Name on the `TFTP-SERVER` form.

```
list usage node-name tftpserv

                                LIST USAGE REPORT
Used By
TFTP Server                           Server Node Name
```

usage-ip-address

The list usage ip-address command may also be used. Output is similar to forms displayed by the list usage node-name command.

4602 IP telephone

Station

There is a new entry (4602) in the `Type` field on the `STATION` form.

```
add station next                                              Page 1 of 3
STATION

Extension: 4005                        Lock Messages? n       BCC: 0
     Type: 4602                        Security Code: ______   TN: 1
     Port: IP                     Coverage Path 1: ___      COR: 1
     Name: Bldg D, Rm H11_____    Coverage Path 2: ___      COS: 1
                                  Hunt-to-Station: ____

STATION OPTIONS
        Loss Group: _             Personalized Ringing Pattern:
                                              Message Lamp Ext:
      Speakerphone: 1-way                  Mute button enabled?
  Display Language? English                  Media Complex Ext:
                                                  IP Softphone? y
```

```
add station next                                              Page 2 of 3
                                      STATION
FEATURE OPTIONS
          LWC Reception: msa-spe         Auto Select Any Idle Appearance? n
         LWC Activation? y                         Coverage Msg Retrieval? y
 LWC Log External Calls? n                                 Auto Ansnwer: none
            CDR Privacy? n                               Data Restriction? n
  Redirect Notification? n                   Idle Appearance Preference? n
Per Button Ring Control? n
  Bridged Call Alerting? n                     Restrict Last Appearance? n
 Active Station Ringing: single

       H.320 Conversion? n         Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed
        Multimedia Mode:                  basic Audible Message Waiting? n
   MWI Served User Type:                     Display Client Redirection? n
           AUDIX Name:                     Select Last Used Appearance? n
  Messaging Server Name:                     Coverage After Forwarding? n
       Automatic Moves: n                      Multimedia Early Answer? n
    IP Emergency Calls:                  Direct IP-IP Audio Connections? y
 Emergency Location Ext:                           IP Audio Hairpinning? y
```

NOTE:
Notice the `Restrict Last Appearance` field defaults to **n**.

On the 4602 IP telephone, page 3 of the `STATION` form has an additional field: `voice-mail Number`. For more information on this field, see "Voice mail retrieval button" on page 52.

```
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
 2: call-appr

    voice-mail Number: _________
```

Type

For each station that you want to add to your system, you must specify the type of telephone in the `Type` field. This is how you distinguish between the many different types of telephones.

The following table lists telephones, virtual phones, and personal computers that you can administer on an Avaya DEFINITY Server or S8000-series Media Server. To administer telephones that are not in the table, use the `ALIAS STATION` form.

NOTE:
You cannot change an analog phone administered with hardware to a virtual extension if TTI is **y** on the **FEATURE-RELATED SYSTEM PARAMETERS** form. Contact your Avaya representative for more information.

Table 2. telephones

Telephone type	Model	Administer as
IP Telephone	4602	4602
	4602SW	4602
	4606	4606
	4612	4612
	4620	4620
	4624	4624
	4630	4630
	4630SW	4630
Wireless Telephone	3606	4606

4620 IP telephone

Abbreviated Dialing System list

This is the same as the Abbreviated Dialing System list for the 2420 DCP telephone. See "Abbreviated Dialing System list" on page 89.

Station

There is a new entry in the `Type` field on the `STATION` form. Page 2 of the `STATION` form is the same. However, pages 3 through 5 look different for 4620 telephone.

```
add station 1014                                          Page 3 of X
                                  STATION

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


ABBREVIATED DIALING
     List1: ________          List2: _________         List3: _________

BUTTON ASSIGNMENTS - SCREEN 1
 1: call-appr                          5: ________
 2: call-appr                          6: ________
 3: call-appr                          7: ________
 4: ________                           8: ________
```

```
add station 4005                                          Page 4 of 4
                                  STATION

 FEATURE BUTTON ASSIGNMENTS

  9:
 10:
 11:
 12:
 13:
 14:
 15:
 16:
 17:
 18:
 19:
 20:
 21:
 22:
 23:
 24:
```

If the Expansion Module field is **y**, a fifth page appears.

```
add station 4005                                              Page 5 of 5
                                      STATION
EXPANSION MODULE BUTTON ASSIGNMENTS

1:                                    13:
2:                                    14:
3:                                    15:
4:                                    16:
5:                                    17:
6:                                    18:
7:                                    19:
8:                                    20:
9:                                    21:
10:                                   22:
11:                                   23:
12:                                   24:
```

Type

For each station that you want to add to your system, you must specify the type of telephone in the `Type` field. This is how you distinguish between the many different types of telephones.

The following table lists telephones, virtual phones, and personal computers that you can administer on an Avaya DEFINITY Server or S8000-series Media Server. To administer telephones that are not in the table, use the `ALIAS STATION` form.

Table 3. telephones

Telephone type	Model	Administer as
IP Telephone	4602	4602
	4602SW	4602
	4606	4606
	4612	4612
	4620	4620
	4624	4624
	4630	4630
	4630SW	4630
Wireless Telephone	3606	4606

Analog busy automatic callback without flash

This section shows changes to existing forms and new options for existing fields associated with this development item. The introduction explains why the administrator uses the form, and the table describes the use of each new field or option on the form.

Appears only if, on the Feature-Related System Parameters form, the `Without Flash` field is **y**. This field then defaults to **y** for all analog phones that allow Analog Automatic Callback.

Valid entries	Usage
y/n	Type **y** to provide automatic callback for a calling analog station without flashing the hook.

Feature-related system parameters

Field descriptions for page 6

```
change system-parameters features (page 6)
                        FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer? n              No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n   Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                 Unhold? n
   No Hold Conference Timeout: 60



ANALOG BUSY AUTO CALLBACK
               Without Flash?                       Announcement:
                                      Voice Mail Hunt Group Ext:
```

Form 21. Feature-Related System Parameters form

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Valid entries	Usage
y/n	Type **y** to provide automatic callback for a calling analog station without flashing the hook.

Announcement

Appears only if the Without Flash field is **y**.

Valid entries	Usage
	Type a valid announcement extension. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the Without Flash field is **y**.

Valid entries	Usage
y/n	Type a voice mail hunt group extension.

Station

```
change station 75001                                          Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? msa-spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
            CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single

       H.320 Conversion? n            Per Station CPN - Send Calling Number? _
      Service Link Mode: as-needed         Busy Auto Callback Without Flash? n
        Multimedia Mode: basic
    MWI Served User Type: ______                 Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                    Select Last Used Appearance? n
   Messaging Server Name: ______                Coverage After Forwarding? _
     Recall Rotary Digit? n                      Multimedia Early Answer? n
      IP Emergency Calls: extension       Direct IP-IP Audio Connections? n
  Emergency Location Ext: 75001                        IP Audio Hairpinning? n
```

Form 22. Station form

AUDIX one-step recording

The changes in the following forms allow for the AUDIX one-step recording feature to be administered.

Station

```
change station 001                                          Page 3 of 4
                                  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. audix-rec  Ext: 4000                  10:
```

Form 23. STATION form, page 3

Button assignments

For the `audix-rec` button assignment, type a valid hunt group extension in the `Ext` field.

display-messages view-buttons

```
change display-messages view-buttons                          Page 9 of 9
                            LANGUAGE TRANSLATIONS

 English                              Translation

  1.  Station Lock                      1. **************************
  2.  License Error                     2. **************************
  3.  Conference Display                3. **************************
  4.  Conf/Trans Toggle-Swap            4. **************************
  5.  Far End Mute                      5. **************************
  6.  Audix Recording                   6. **************************
```

Form 24. LANGUAGE TRANSLATIONS form, page 9

system-parameters features

```
change system-parameters features                                Page 6 of 12
                         FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer? n                    No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n           Select Line Appearance Conferencing? n
Abort Conference Upon Hang-up? n                                      Unhold? n

ANALOG BUSY AUTO CALLBACK
                Without Flash? n


AUDIX ONE-STEP RECORDING
Apply Ready Indication Tone To Which Parties In The Call? all
  Interval For Applying Periodic Alerting Tone (seconds): 15
```

Form 25. FEATURE-RELATED SYSTEM PARAMETERS form, page 6 (default settings)

Apply Ready Indication Tone To Which Parties In The Call?

Valid entries	Usage
all	Default value. All members on the call hear an alert tone to indicate that the call is being recorded. See the "Note" below for a further explanation.
initiator	Only the initiator of the call hears an alert tone.
none	No one on the call hears an alert tone.

NOTE:
When the default answer **all** is chosen for the Apply Ready Indication Tone To Which Parties In The Call? field, the Interval For Applying Periodic Alerting Tone (seconds) field becomes visible.

If **initiator** or **none** is chosen for the Apply Ready Indication Tone To Which Parties In The Call? field, the Interval For Applying Periodic Alerting Tone (seconds) field does not appear.

Interval For Applying Periodic Alerting Tone (seconds)

Valid entries	Usage
0-60	Type a number from 0 to 60, indicating the interval that an alert tone will sound to all members on the call. For example, by typing 30, an alert tone will be sounded every thirty seconds to indicate to the participants the call is being recorded. The default value is 15. A value of 0 disables the tone.

display-messages button-labels

```
change display-messages button-labels                          Page 2 of 10
                            LANGUAGE TRANSLATIONS

English                              Translation

 1.  Alternate FRL                    1. ***************
 2.  ANI Request                      2. ***************
 3.  Assist                           3. ************
 4.  ASVN Halt                        4. ***************
 5.  AttQueueCall                     5. ***************
 6.  AttQueueTime                     6. ***************
 7.  Audix Record                     7. ***************
 8.  Auto Callback                    8. ***************
 9.  Auto Ckt Halt                    9. ***************
10.  AutoIC                          10. ********
11.  Auto In                         11. *************
12.  AutoWakeAlarm                   12. ***************
13.  Auto Wakeup                     13. ***************
14.  AuxWork                         14. ********
15.  Busy                            15. *****
```

Form 26. LANGUAGE TRANSLATIONS form, page 2

The button label for 2420 DCP telephones and for 4620 IP telephones for the new feature button can be user defined. Entries on this page are listed in alphabetical order (your `LANGUAGE TRANSLATIONS` form may look different from this example).

Call Detail Recording (CDR) display of physical extension

system-parameters cdr

```
display system-parameters cdr                                  Page 1 of 2
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                        CDR Date Format: month/day
      Primary Output Format:
    Secondary Output Format:
           Use ISDN Layouts? n
       Use Enhanced Formats? n       Condition Code 'T' For Redirected Calls? n
Modified Circuit ID Display? n                   Remove # From Called Number? n
                 Record Outgoing Calls Only? n             Intra-switch CDR? n
  Suppress CDR for Ineffective Call Attempts? y      Outg Trk Call Splitting? y
      Disconnect Information in Place of FRL? n        Outg Attd Call Record? y
                                                    Interworking Feat-flag? n
 Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                  Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? n       Record Agent ID on Outgoing? n
     Inc Trk Call Splitting? n              Inc Attd Call Record? n
  Record Non-Call-Assoc TSC? n
      Record Call-Assoc TSC? n    Digits to Record for Outgoing Calls: dialed
Privacy - Digits to Hide: 0                  CDR Account Code Length: 15
```

Form 27. CDR SYSTEM PARAMETERS form, page 1

Record Agent ID on Incoming? field

Valid entries	Usage
y	Type **y** to display the agent ID on the CDR input.
n	Type **n** to display the physical extension on the CDR input.

Record Agent ID on Outgoing? field

Valid entries	Usage
y	Type **y** to display the agent ID on the CDR output.
n	Type **n** to display the physical extension on the CDR output.

CLAN QoS and CIDR support

This section shows changes to existing IP administration forms, including new fields/values and new options for existing fields associated with these IP enhancements.

It is now possible to administer some features through other administration interfaces than Avaya call-processing forms. In particular, the S8300 Media Server with a G700 Media Gateway can be administered through its Linux operating system.

Windows Operating System

Whether the TN799 Control-LAN circuit pack tags frames with VLAN and user-priority values is administered on the IP Network Region form.

Linux Operating System

Whether the TN799 Control-LAN circuit pack tags frames with VLAN and user-priority values is administered on the IP Network Region form. The IP Network Region form and the bash command under the Linux operating system both set the same entries in a configuration file. There is no difference between setting them through one user interface versus the other.

Tagging is administered through the bash command line as follows:

vlanconfig -c -d device -v vlan_id [-i ip_address] [-g gw_address] [-m netmask] [-e on|off] [-f]

vlanconfig -r -d device -v vlan_id [-nf]

vlanconfig -q [-d device] [-v vlan_id]

-c	will create/change an interface
-r	will remove an interface
-q	will query an interface
-d device	create/change, remove, or query this device (valid with the -c, -r, and -q options)
-e on|off	enable or disable the interface (valid only with the -c option)
-f	force the command to execute (valid only with the -c and -r options)

IP-routing

To support Classless Inter-Domain Routing (CIDR) and the Variable Length Subnet Mask (VLSM), two new inter-related fields appear on the IP Routing form:

- Network Bits
- Subnet Mask

Field descriptions

```
change ip-route 1                                              Page 1 of 1

                                IP ROUTING

    Route Number:
Destination Node:
    Network Bits:      Subnet Mask:
         Gateway:
           Board:
          Metric:
```

Form 28. IP Routing form

Network Bits/Subnet Mask

There is one-to-one mapping between the Network Bits and the Subnet Mask fields; entering a value in one field uniquely determines the other field. Refer to more detailed information contained in networking documentation for Avaya products.

NOTE:
For the Network Bits and Subnet Mask fields, if you put a value into either field and then press ENTER or TAB to move the cursor to another field, the other field gets populated automatically with a value corresponding to the one you just entered.

Network Bits

This field is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
0-32	To set the size of the network portion of the subnet mask. Default is 24.

Subnet Mask

The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
First 4 octets: 255 254 252 248 240 224 192 128 0	Identifies the subnet mask associated with the IP address for this IP interface. Default is 255.255.255.0.

Board

Valid entries	Usage
1 to **44**	cabinet
A to **E**	carrier
0 to **20**	slot
1 to **44**	gateway
V1 to **V9**	module

IP-network-region

Field descriptions for page 1

```
change ip-network-region 3                                      Page 1 of 2

                                IP Network Region
      Region: 3        Name: North
    Location: __

                      Intra-region IP-IP Direct Audio? translated(NAT)
AUDIO PARAMETERS      Inter-region IP-IP Direct Audio? native (NAT)
  Codec Set: 2                                        IP Audio Hairpinning? y
   Location: 1
 UPD Port Range                                     RTCP Reporting Enabled? y
          Min: 2048_                             RTCP MONITOR SERVER PARAMETERS
          Max: 3028                               Use Default Server Parameters? n
                                                   Server IP Address: 1 .2 .3 .4
DIFFSERV/TOS PARAMETERS                                        Server Port: 5005
 Call Control PHB Value: 34_                      RTCP Report Period(secs): 5
    Audio PHB Value: 46
          BBE PHB Value: 43                   Resource Reservation Parameters
                                                                 RSVP Enabled? y
                                                        RSVP Refresh Rate(secs): 15
 Call Control 802.1p Priority: 7               Retry upon RSVP Failure Enabled? y
      Audio 802.1p Priority: 6                 RSVP Profile: guaranteed-service
```

Form 29. IP Network Region form

Region

A display-only field indicating the number of the region being administered.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

AUDIO PARAMETERS

Codec Set

Specifies the codec assigned to the region.

Valid entries	Usage
1-7	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Location

Specifies the location by IP network region allowing correct date and time information and trunk routing based on IP network region.

Valid entries	Usage
1-44	(For DEFINITY R, CSI, SI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1-64	(For Avaya S8300 Media Server, Avaya S8700 Multi-Connect, and Avaya S8700 IP-Connect only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN that the endpoint registered through or the media gateway containing the Internal Call Controller or Local Spare Processor on an Avaya S8300 Media Server that the endpoint registered through. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP reporting. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.

UDP Port Range

UPD Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
3-65535	Enter the highest UDP port number to be used for audio transport.

UPD Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
2-65534	Enter the lowest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the Use Default Server Parameters field is set to **n** and the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
5-30	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the Use Default Server Parameters field is set to **n** and the and the RTCP Enabled field is set to **y**

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the Use Default Server Parameters field is set to n and the and the RTCP Enabled field is set to **y**.

Valid entries	Usage
1-65535	Enter the port for the RTCP Monitor server.

Use Default Server Parameters

This field only appears when the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If set to **y**, you must complete the Default Server IP Address field on the IP Options System Parameters screen.
n	If you enter **n**, you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

DIFFSERVE/TOS PARAMETERS

BBE PHB Value

This field contains the Better than Best Effort (BBE) PHB value.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ BBE PHB value.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the 802.1p priority value.

Call Control Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the DiffServ codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control PHB value.

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the Audio 802.1p priority value.

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the Audio PHB codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ Audio PHB value.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the RSVP Enabled field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	
controlled-load	

RSVP Refresh Rate (secs)

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
1-99	Enter the RSVP refresh rate in seconds.

Co-Resident DLG

Call vector

These forms program a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN). See the *Avaya™ Communication Manager Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide* for additional information.

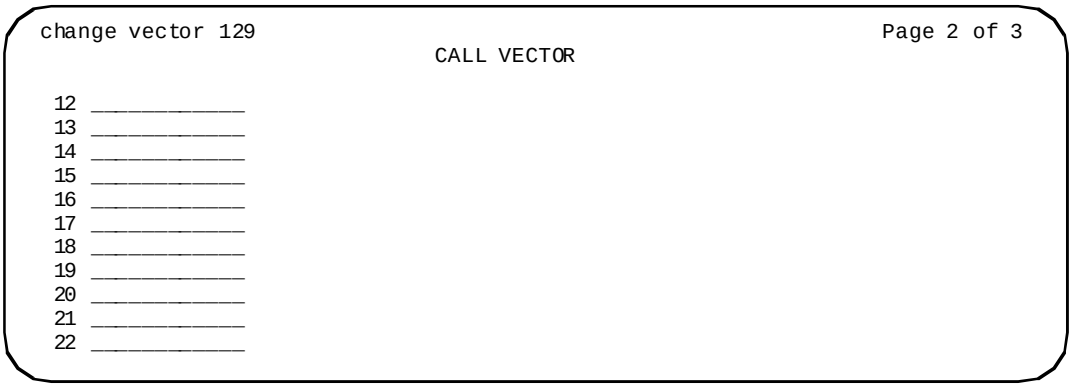
Field descriptions for page 1

```
 change vector 129                                              Page 1 of 3
                                CALL VECTOR
   Number: 129                Name: _______________
Multimedia? n      Attendant Vectoring? n     Meet-me Conf? y          Lock? n
     Basic? y  EAS? n   G3V4 Enhanced? y    ANI/II-Digits? y   ASAI Routing? n
 Prompting? y  LAI? n  G3V4 Adv Route? y    CINFO? y   BSR? n       Holidays? n

  01 _____________
  02 _____________
  03 _____________
  04 _____________
  05 _____________
  06 _____________
  07 _____________
  08 _____________
  09 _____________
  10 _____________
  11 _____________
```

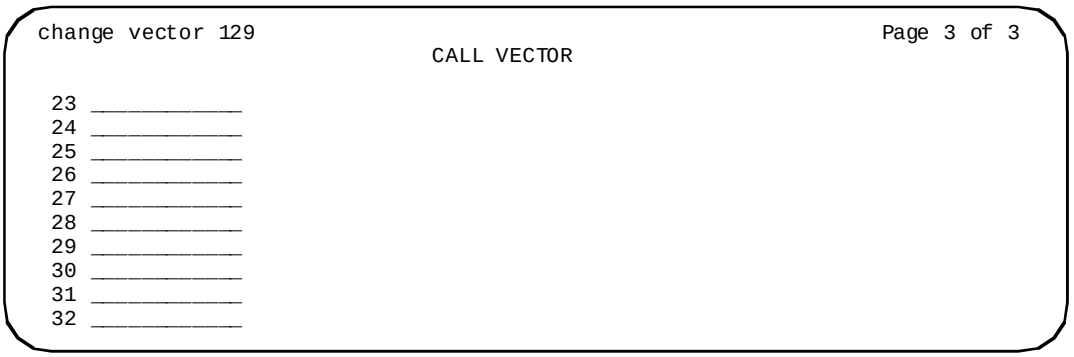
Form 30. Call Vector form 1 of 3

Field descriptions for page 2



Form 31. Call Vector form 2 of 3

Field descriptions for page 3



Form 32. Call Vector form 3 of 3

01 through XX

Type vector commands as required (up to the maximum allowed in your configuration). For more information, see the *Avaya™ Communication Manager Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide*.

Valid entries	Usage
adjunct	Causes a message to be sent to an adjunct requesting routing instructions based on the CTI link number.
announcement	Provides the caller with a recorded announcement.
busy	Gives the caller a busy signal and causes termination of vector processing.

Valid entries	Usage
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).
collect	Allows the user to type up to 16 digits from a touch-tone phone, or allows the vector to retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the switch. Also allows the optional assignment of an announcement that will play immediately before the disconnect.
goto	Allows conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	Allows the caller to leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it “returns” best data for its location to the primary vector on the origin switch.
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number).
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command’s syntax. Also provides feedback (in the form of silence, ringback, or music) to the caller while the call advances in queue.

Hunt group

Allows calls to be answered by users (agents) at a predefined group of telephones or devices.

This form creates a hunt group that is identified by a hunt group number. Users assigned to a hunt group are identified by their extension number.

This form can be used to implement a hunt group and its associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing. Look at the various hunt group forms and choose the forms that can be used to implement your hunt group requirements.

The total number of pages vary depending on your System configuration. See the *Avaya™ MultiVantage Solutions Hardware Guide* for the maximum number of hunt groups supported by each configuration.

The System checks for the busy or idle status of extension numbers in the hunt group when answering calls. A Uniform Call Distribution (UCD) type hunt group selects the “most idle” extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call.

Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the “most idle” agent or the “least occupied” agent with the highest skill level for the call’s skill.

Vector controlled splits/skills can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that will terminate the call to a vector controlled split/skill); however, the calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

Field description for page 2

Page 2 of the `HUNT GROUP` form appears only when the ACD field on page 1 is **y**. If the ACD field is **n**, page 3 becomes page 2 and all subsequent page numbers are decreased by one.

The `Timed ACW Interval` field appears only if, on the `SYSTEM-PARAMETERS CUSTOMER-OPTION` form, the `Timed ACW` field on page 3 is **y**.

```
change hunt-group 4                                          Page 2 of X
                                  HUNT GROUP
                  Skill? _            Expected Call Handling Time (sec): ___
                    AAS? _
                Measured: ____
   Supervisor Extension: ____
  Priority on Intraflow? _

    Controlling Adjunct: ____



Timed ACW Interval (sec): ___
  Multiple Call Handling: _____________
                                          Redirect on No Answer (rings): ___
                                                    Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Form 33. Hunt Group form when ACD is y and Queue and Vector are n

```
change hunt-group x                                          Page 2 of X
                                  HUNT GROUP
                    Skill? _          Expected Call Handling Time (sec): ___
                      AAS? _             Acceptable Service Level (sec):
                  Measured: internal
    Supervisor Extension: ____



     Controlling Adjunct: ____



       VuStats Objective: ____
Timed ACW Interval (sec): ___
  Multiple Call Handling: _____________
                                          Redirect on No Answer (rings): ___
                                                    Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Form 34. Hunt Group form when Queue and Vector are y

Controlling Adjunct

Appears only if the ACD field is **y**. If the controlling adjunct is a CONVERSANT voice system (requires an ASAI link), then type **asai** in this field. (On the SYSTEM-PARAMETERS CUSTOMER-OPTION form, the ASAI Link Core Capabilities and Computer Telephony Adjunct Links fields must be **y** for CallVisor ASAI capability and for an entry other than none.)

Valid entries	Usage
none	Indicates that members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can only use their data terminal keyboards to perform phone functions (for example, change work state).
adjlk	Computer Telephony Adjunct Links
asai-ip	Indicates ASAI links administered without hardware and used by the Co-Resident DLG application. You cannot use this selection unless, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, the Co-Res DEFINITY LAN Gateway field is **y**.
adj-ip	Indicates ASAI adjunct links administered without hardware and used by the Co-Resident DLG application. You cannot use this selection unless, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, the Co-Res DEFINITY LAN Gateway field is **y**.

IP services

```
change ip-services                                              Page 1 of X

                              IP SERVICES
Service      Enabled    Local      Local         Remote       Remote
  Type                   Node       Port          Node         Port
DLG_________    y      clan10       5678   ____________________ _____
DLG_________    y      clan2        5678   ____________________ _____
DLG_________    y      clan3        5678   ____________________ _____
DLG_________    y      clan4        5678   ____________________ _____
DLG_________    y      clan5        5678   ____________________ _____
DLG_________    y      clan6        5678   ____________________ _____
DLG_________    y      clan7        5678   ____________________ _____
DLG_________    y      clan1        5678   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
____________    _      _________    ____   ____________________ _____
```

Form 35. IP Services form

Enabled

When the Service Type field is **DLG**, this field controls whether the co-resident DLG application listens to the interface to which it is bound.

Valid entries	Usage
y	Type **y** to enable this IP service.
n	Does not listen over the interface.

Local Node

Specify the node name for the port. When the Service Type field is **DLG**, there can be only one entry on this form with a Service Type field of **DLG** and the same Local Node.

Valid entries	Usage
Node names as defined on the NODE NAMES form.	If the link is administered for services over the C-LAN circuit pack, type a node name defined on the NODE NAME form. See *Administration for Network Connectivity for Avaya™ Communication Manager* for information on how to administer node names.
processor	Processor is only available for S8100 Media Server and S8300 Media Server.

Local Port

Specify the originating port number. When the Service Type field is **DLG**, this field becomes **5678** and is read-only.

Valid entries	Usage
5000 to **9999**	5111-5117 for SAT applications 5678 for ASAI
0	For client applications, this defaults to zero.

Remote Node

Specify the switch at the far end of the link for SAT. The remote node should not be defined as a link on the IP INTERFACE or DATA MODULE forms. When the Service Type field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
Node name as defined on the NODE NAMES form	For SAT, use a node name to provide added security.
any	Use any available node.

Remote Port

Specify the port number of the destination. When the Service Type field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
5000 to **64500**	Use if this service is a client application, such as CDR or PMS. This must match the port administered on the adjunct, PC or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

Defines the service provided.

Valid entries	Usage
ALARM1, ALARM2	Available only on DEFINITY R. Use this to connect send alarms over a TCP/IP link.
cbc	Type **cbc** to reserve the trunk for outgoing use only to enhance Network Call Redirection.
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.
DAPI	Available only on DEFINITY R, CSI, SI, S8300 Media Server, S8700 IP-Connect, S8700 Multi-Connect.

Valid entries	Usage
DLG	For S8300 Media Server and S8100 Media Server only. Can be entered only if, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, the Co-Res DEFINITY LAN Gateway field is **y**. Use this service type for co-resident DEFINITY LAN Gateway. For more information, see the *Avaya™ Communication Manager CallVisor® ASAI Technical Reference*. **NOTE:** If DLG is selected, another form displays. See "DLG administration" on page 77.
PMS	
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.
SAT	System administration terminal. Not available on S8100 Media Server with a CMC1.
SYS_PRINT	Use this to connect the system printer over a TCP/IP link.

IP Services form (Session Layer Timers page)

Use this form to enable reliable protocol for TCP/IP links, and to establish other session-layer parameters. This form only appears if you type **CDR1**, **CDR2**, **PMS_JOURNAL**, or **PMS_LOG** in the Service Type field on page 1 or 2.

```
change ip-services                                        Page   3 of   3

                              SESSION LAYER TIMERS
Service    Reliable    Packet Resp     Session Connect     SPDU     Connectivity
 Type      Protocol       Timer          Message Cntr      Cntr        Timer
CDR1          y           3                   1             1           1
```

Form 36. IP Services form (Session Layer Timer page)

Connectivity Timer

Valid entries	Usage
1–255	Indicates the amount of time (in seconds) that the link can be idle before the switch sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid entries	Usage
1–255	Determines the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Indicates whether you want to use reliable protocol over this link.

Valid entries	Usage
y/n	Use reliable protocol if the adjunct on the far end of the link supports it.

Service Type

A display-only field that identifies the service type for which you are establishing parameters.

Valid entries	Usage
CDR1, CDR2	Used to connect either the primary or secondary CDR device over a TCP/IP link.
PMS_JOURNAL	Used to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Used to connect the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid entries	Usage
1–5	The Session Connect Message counter indicates the number of times the switch tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid entries	Usage
1–5	The Session Protocol Data Unit counter indicates the number of times the switch transmits a unit of protocol data before generating an error.

Conference/transfer enhancements

Changes were made to administration forms and field values for the conference enhancements. This section describes those changes.

Changes to existing forms and new options for existing fields are shown if they are associated with this development item. The introduction explains why the administrator uses the form, and the table describes the use of each new field or option on the form.

New feature buttons for stations and consoles

There following buttons are new for the conference enhancements:

- togle-swap

  This button allows a user to toggle between two called parties before completing a conference or a transfer. This button can be assigned to stations (**add/change station XX**) but not to an attendant console. The attendant console already has this function using the Split Swap button.

  To use this new button, on the System-Parameters Customer-Options form, the G3 Version field must be set to **V11** or higher (see “System-parameters customer-options”).

- conf-dsp

  This button allows a user to display information about each party of a conference call. This button can be assigned to both stations (**add/change station XX**) and attendant consoles (**add/change attendant XX**).

  To use this new button, on the System-Parameters Customer-Options form, the G3 Version field must be set to **V11** or higher and the Enhanced Conferencing option must be **y** (see “System-parameters customer-options”).

System-parameters customer-options

To enable the Conference Enhancements, the `G3 Version` field must be set to **`V11`** on the `SYSTEM-PARAMETERS CUSTOMER-OPTIONS` form. The field should be set according to the installed license file.

```
change system-parameters customer-options                Page   1 of   9   SPE A
                              OPTIONAL FEATURES
                                                              Used
     G3 Version: V11                      Maximum Ports: 2800  1437
       Location: 1                Maximum XMOBILE Stations: 0     0
       Platform: 2

IP PORT CAPACITIES
                        Maximum Administered IP Trunks: 100   83
            Maximum Concurrently Registered IP Stations: 100   2
              Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
               Maximum Concurrently Registered IP eCons: 0     0



   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                                    Maximum VAL Boards: 1     0


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

A new field, `Enhanced Conferencing`, has been added to the `SYSTEM-PARAMETERS CUSTOMER-OPTIONS` form. The field should be set according to the installed license file.

```
change system-parameters customer-options                Page   3 of   9   SPE A
                              OPTIONAL FEATURES

   Emergency Access to Attendant? y                          ISDN-BRI Trunks? y
            Enhanced Conferencing? y                                 ISDN-PRI? y
                   Enhanced EC500? n                     Local Spare Processor? n
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y         Multimedia Call Handling (Basic)? y
 Hospitality (G3V3 Enhancements)? y      Multimedia Call Handling (Enhanced)? y
                       IP Trunks? y                       Multiple Locations? n
            IP Attendent Consoles? n            Personal Station Access (PSA)? y
                     IP Stations? y

               ISDN Feature Plus? n
   ISDN Network Call Redirection? y


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

Enhanced conferencing

Enhanced conferencing allows the customer to use the Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and No Hold Conference features. This field can be enabled only if, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, the G3 Version field is **V11** or higher.

Valid entries	Usage
y/n	Type **y** to enable access to the Enhanced Conferencing features.

Meet-me Conference VDN

A new field, Meet-me Conference, has been added to the VDN form.

```
add vdn 36090                                          Page   1 of   2   SPE A
                          VECTOR DIRECTORY NUMBER

                           Extension: 36090
                                Name: Enhanced Conferencing VDN
                      Vector Number: 90

                  Meet-me Conference? y

                                COR: 1
                                 TN: 1
```

Meet-me Conference

This field appears only if, on the SYSTEM-PARAMETERS CUSTOMER-OPTIONS form, the Enhanced Conferencing field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

NOTE:
If the VDN extension is part of the customer's DID block, external users will be able to access the conference VDN. If the VDN extension is not part of the customer's DID block, only internal callers on the customer's network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Type **y** to enable Meet-me Conference for this VDN. If Meet-me Conference is enabled, several fields do not display. Both Attendant Vectoring and Meet-me Conference cannot be enabled at the same time. If Enhanced Conferencing is enabled, but no other vectoring customer options are enabled, only Meet-me Conference vectors can be assigned.

If Meet-me Conference is enabled, the following new fields display on Page 2.

```
change vdn 36090                                        Page   2 of   2   SPE A
                          VECTOR DIRECTORY NUMBER

                       MEET-ME CONFERENCE PARAMETERS
       Conference Access Code: 937821
        Conference Controller: 80378
```

Conference Access Code

This field appears only if, on the VECTOR DIRECTORY NUMBER form, the Meet-me Conference field is **y**. This field allows you to assign an access code to the Meet-me Conference VDN.

SECURITY ALERT:
You should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
blank or 6-digit number	Type a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave the field blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the “init” superuser login.

Conference Controller

This field appears only if, on the `VECTOR DIRECTORY NUMBER` form, the `Meet-me Conference` field is **y**. This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code.

NOTE:
A user can change the access code only after it has been first assigned by the system administrator, and only system administrators can remove an access code.

Valid entries	Usage
extension number or blank	If an extension number is entered, a user at that extension can change the access code for the Meet-me Conference VDN using a feature access code. If this field is blank, only a station user that is assigned with console permissions can change the access code for the Meet-me Conference VDN using a feature access code. In addition, remote access users can change a Meet-me Conference access code using the feature access code.

Meet-me Conference Call Vector

The Call Vector form has a new field that designates the vector as a Meet-me Conference vector. The **collect**, **goto step**, and **route-to** vector steps have new options or conditions for the Meet-me Conference feature.

The following forms shows an example of a Meet-me Conference vector.

```
change vector 90                                        Page   1 of   3   SPE A
                                CALL VECTOR

    Number: 90               Name: Enh Conf Vec
                  Attendant Vectoring? n    Meet-me Conf? y           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n    BSR? n   Holidays? n

01 collect      6    digits after announcement 12340
02 goto         step   6   if digits           =      meet-me-access
03 collect      6    digits after announcement 12341
04 goto         step   6   if digits           =      meet-me-access
05 disconnect   after announcement 12342
06 goto         step   11  if meet-me-idle
07 goto         step   14  if meet-me-full
08 announcement 12343
09 route-to     meetme
10 stop
11 announcement 12344
```

```
change vector 90                                    Page   2 of   3   SPE A
                                CALL VECTOR

12 route-to     meetme
13 stop
14 disconnect   after announcement 12345
15 stop
16
17
18
19
20
21
22
```

Meet-me Conf

This field appears only if, on the System-Parameters Customer-Options form, the Enhanced Conferencing field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Type **y** to enable Meet-me Conference for this vector. If `Meet-me Conference` field is **y**, the `Lock` field also must be **y**. When the `Lock` field is **y**, the vector cannot be changed by adjunct vectoring programs such as Visual Vectors. Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

New options for vector steps

collect step. When the `Meet-me Conf` field is enabled, the **collect** vector step has been modified to collect the next six digits and use those digits as the access code for a Meet-me Conference call. See vector steps 1 and 3 in the example above.

goto step. The **goto step** vector step has two new conditions:

- meet-me-idle
- meet-me-full

The **meet-me-idle** condition routes the first caller accessing a Meet-me Conference to the conference call. An announcement step saying they are the first party to access the call can be given to the caller. See vector steps 6 and 11 in the example above.

The **meet-me-full** condition is used when the Meet-me Conference already has the maximum of six parties on the call. See vector steps 7 and 14 in the example above.

The goto step vector also has a new option, **meet-me access**, for the digits condition to verify that the access code is valid. If the access code entered by the caller equals the access code administered for the VDN, vector processing continues. See vector steps #2 and #4 in the example above.

route-to step. The **route-to** vector step has one new condition: **meetme**. When successful, this condition adds the caller to the Meet-me Conference call, and all parties on the call hear an "entry" tone to signify that another caller has joined the conference. This condition is valid when the caller has entered the correct access code and there are not already six parties on the call. See vector steps 9 and 12 in the example above.

If the **route to meetme** step ever fails, vector processing stops and the caller hears busy tone.

Meet-me Conference vector scenario

Joe Davis has a sales review scheduled with four associates located in different cities. He has reserved Meet-me Conference telephone number 865-253-6090. In switch administration, this number has been assigned to vector 90. See the following form.

```
change vdn 36090                                      Page   1 of   2   SPE A
                          VECTOR DIRECTORY NUMBER

                            Extension: 36090
                                 Name: Enhanced Conferencing VDN
                         Vector Number: 90

                    Meet-me Conference? y

                                  COR: 1
                                   TN: 1
```

VDN 36090 is administered with an access code of 835944. See the following form.

```
change vdn 36090                                      Page   2 of   2   SPE A
                          VECTOR DIRECTORY NUMBER

                        MEET-ME CONFERENCE PARAMETERS
       Conference Access Code: 835944
        Conference Controller: _______
```

When each associate calls the Meet-me Conference telephone number, the following vector processing occurs:

```
change vector 90                                        Page   1 of   3   SPE A
                                CALL VECTOR

    Number: 90               Name: Enh Conf Vec
               Attendant Vectoring? n    Meet-me Conf? y           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n

01 collect      6    digits after announcement 12340
02 goto         step   6   if digits           =      meet-me-access
03 collect      6    digits after announcement 12341
04 goto         step   6   if digits           =      meet-me-access
05 disconnect   after announcement 12342
06 goto         step   11  if meet-me-idle
07 goto         step   14  if meet-me-full
08 announcement 12343
09 route-to     meetme
10 stop
11 announcement 12344
```

```
change vector 90                                        Page   2 of   3   SPE A
                                CALL VECTOR

12 route-to     meetme
13 stop
14 disconnect   after announcement 12345
15 stop
16
17
18
19
20
21
22
```

Each caller hears announcement 12340, which says something similar to “Welcome to the Meet-me Conferencing service. Type your conference access code.” Each caller enters the access code 835944.

The **collect** vector step 1 collects the access code digits. If the access code is valid, the vector processing continues with vector step 6. If the access code is invalid, the vector processing continues with vector step 3, which plays announcement 12341. Announcement 12341 says something similar to “This access code is invalid. Please enter the access code again.” If the caller enters the wrong access code again, the vector processing continues with vector step 5, which plays announcement 12342. Announcement 12342 says something similar to “This access code is invalid. Please contact the conference call coordinator to make sure you have the correct conference telephone number and access code. Good bye.”

Vector step 6 is only valid for the first caller into the Meet-me Conference. The **meet-me-idle** condition routes the first caller to announcement 12344 (vector step 11). The recorded announcement says something similar to “You are the first party to join the call.” The caller is then routed to the Meet-me Conference call by vector step 12 and vector processing stops.

Vector step 7 is used when the Meet-me Conference already has the maximum of six parties on the call. The **meet-me-full** condition disconnects the caller after playing announcement 12345 (vector step 14). The recorded announcement says something similar to “This Meet-me Conference is filled to capacity. Please contact the conference call coordinator for assistance. Good bye”

If a caller enters the correct access code, is not the first caller, and the conference call is not full, vector processing continues with vector step 8, which plays announcement 12343. The announcement says something similar to “Your conference call is already in progress.” The caller is then routed to the Meet-me Conference call by vector step 9 and vector processing stops. As each caller enters the conference call, all parties on the call will hear an “entry” tone.

When the conference call is over and callers drop out of the conference call, any remaining parties on the call will hear an “exit” tone.

Interactions for Meet-me Conference vectors

A non Meet-me Conference vector cannot be assigned to a Meet-me Conference VDN and a Meet-me Conference vector cannot be assigned to a non Meet-me Conference VDN.

There will be no restrictions in vector chaining between Meet-me Conference and non Meet-me Conference vectors (for example, using the **goto vector** or **route-to number** commands). When calls interflow from one type of vector processing to another, they will be removed from any queue (if applicable) and treated as new calls to vectoring, not a continuation of vectoring.

Feature access codes

A new feature access code is added to allow the controlling user of a Meet-me Conference VDN to change the access code.

```
change feature-access-codes                                    Page   2 of   6
                              FEATURE ACCESS CODE (FAC)
     Emergency Access to Attendant Access Code:
                    Enhanced EC500 Activation: 652    Deactivation: 653
  Extended Call Fwd Activate Busy D/A      All:       Deactivation:
       Extended Group Call Pickup Access Code:
              Facility Test Calls Access Code:
                           Flash Access Code:
            Group Control Restrict Activation:       Deactivation:
                  Hunt Group Busy Activation: *14    Deactivation: *15
                           ISDN Access Code:
               Last Number Dialed Access Code: *59
    Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
            Leave Word Calling Send A Message: *49
          Leave Word Calling Cancel A Message: *41
              Malicious Call Trace Activation: *35    Deactivation: *34
        Meet-me Conference Access Code Change: #777
 PASTE (Display PBX data on Phone) Access Code: *50
  Personal Station Access (PSA) Associate Code: *77    Dissociate Code: #772
       Per Call CPN Blocking Code Access Code:
     Per Call CPN Unblocking Code Access Code:
```

Display messages

The display messages for conference and transfer have changed.

```
change display-messages transfer-conference                    Page   3 of   4
                              LANGUAGE TRANSLATIONS

12.     English: Select line ~ to add party.
    Translation: ************************************

13.     English: Select alerting line to answer call.
    Translation: ************************************

14.     English: Transfer canceled.
    Translation: ************************************

15.     English: Connecting to ~.
    Translation: *************************

16.     English: Called party ~ is busy.
    Translation: *************************
```

```
change display-messages transfer-conference                    Page   4 of   4
                              LANGUAGE TRANSLATIONS

17.     English: Invalid number dialed ~.
    Translation: *************************

18.     English: Party ~ is not avaiable.
    Translation: *************************

19.     English: Mute
    Translation: ****
```

Language translations - self-administration

Users on 6400-series telephones that support softkey labels can self-administer a new softkey for the new Conference/Transfer Toggle/Swap and Selective Conference Party Display features. for more information, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

Language translations - softkey labels

```
change display-messages self-administration                    Page   2 of   3
                              LANGUAGE TRANSLATIONS

 English   Translation  English                 Translation

 1. Acct    *****       CDR Acount Code         ************************
 2. AutoD   *****       Automatic Dialing       ************************
 3. CFrwd   *****       Call Forwarding         ************************
 4. CPark   *****       Call Park               ************************
 5. CPkUp   *****       Call Pickup             ************************
 6. DPkUp   *****       Directed Call Pickup    ************************
 7. GrpPg   *****       Group Paging            ************************
 8. SAC     *****       Send All Calls          ************************
 9. Swap    *****       Conf/Trans Toggle-Swap  ************************
10. WspPg   *****       Activate whisper Page   ************************
11. WspAn   *****       Answerback for Whisper  ************************
12. WsOff   *****       Whisper Page Off        ************************
13. Blank   *****       Blank Button            ************************
```

For telephones that support softkey labels, administrators can add a new softkey for the Selective Conference Party Display, Selective Conference Party Mute, No Hold Conference, and Toggle/Swap features. See the following example.

```
change display-messages softkey-labels                         Page   1 of   1
                              LANGUAGE TRANSLATIONS

 English    Translation English    Translation English    Translation

  1. Acct    1. *****   17. Drop   17. *****   33. RngOf  33. *****
  2.  AD     2. *****   18. Excl   18. *****   34.  SAC   34. *****
  3. AdBsy   3. *****   19. FMute  19. *****   35. SFunc  35. *****
  4. Admin   4. *****   20. GrpPg  20. *****   36. Spres  36. *****
  5. AutCB   5. *****   21. HFAns  21. *****   37. Stats  37. *****
  6. BtnVu   6. *****   22. IAuto  22. *****   38. Stop   38. *****
  7. CFrwd   7. *****   23. IDial  23. *****   39. Swap   39. *****
  8. CnfDs   8. *****   24. Inspt  24. *****   40. Timer  40. *****
  9. CnLWC   9. *****   25. Last   25. *****   41. TmDay  41. *****
 10. Cnslt  10. *****   26.  LWC   26. *****   42. View   42. *****
 11. Count  11. *****   27. Mark   27. *****   43. Wait   43. *****
 12. CPark  12. *****   28. NHCnf  28. *****   44. WspAn  44. *****
 13. CPkUp  13. *****   29. Pause  29. *****   45. WspPg  45. *****
 14. CTime  14. *****   30. PCall  30. *****
 15.  Dir   15. *****   31. Prog   31. *****
 16. DPkUp  16. *****   32. RmBsy  32. *****
```

Language translations - view buttons

Administrators can set the user-defined option for the Conference Display, Toggle/Swap, No Hold Conference, and Far End Mute features. See the following example.

```
change display-messages view-buttons                          Page   9 of   9
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1.  Station Lock                    1. ************************
  2.  License Error                   2. ************************
  3.  Conference Display              3. ************************
  4.  Conf/Trans Toggle-Swap          4. ************************
  5.  No Hold Conference              5. ************************
  6.  Far End Mute                    6. ************************
```

Feature-related system parameters

A new field has been added to the Conference/Transfer features to control the timeout of No Hold Conference call setup.

```
change system-parameters features                             Page   6 of  12
                        FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer? n                 No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n        Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                    Unhold? n
   No Hold Conference Timeout: 60
```

Valid entries	Usage
20-120 seconds	This field controls when an attempted No Hold Conference will drop the call attempt and deny the conference call. The default is 60 seconds. The Answer Supervision timeout of trunks using No Hold Conference must also be set at the lowest possible value.

Dial Plan Expansion (DPE)

Because of the Dial Plan Expansion (DPE), numerous forms have changed to accommodate the wider fields needed for the longer extension numbers. Sometimes this also changed other form formatting.

For more detailed information, see the *Administrator's Guide for Avaya™ Communication Manager*, 555-233-506.

Feature-related system parameters

This form implements system parameters associated with various System features.

This form used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a new form, which you can access with the command **`change system-parameters coverage-forwarding`**.

Field descriptions for page 7

```
change system-parameters features                                    page 7
                         FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Names as Connected Name?
 Display Connected Name/Number for ISDN DCS Calls?
       Send ISDN Trunk Group Name on Tandem calls?

                               QSIG TSC Extension:
MWI - Number of Digits Per Voice Mail Subscriber:

                              National CPN Prefix:
                         International CPN Prefix:
                        Pass Prefixed CPN to ASAI:
   Unknown Numbers Considered Internal for AUDIX?
             UNSI Calling Name for Outgoing Calls?
               Path Replacement with Measurements?
                  QSIG Path Replacement Extension:
            Path Replace While in Queue/Vectoring?
```

Form 37. Feature-Related System Parameters form

ISDN parameters

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the `Basic Supplementary Services` field or the `ISDN Feature Plus` field on the **SYSTEM-PARAMETERS CUSTOMER-OPTIONS** form is **y**. This field provides an indication of the number of digits per AUDIX subscriber.

NOTE:
For QSIG-MWI only. These routing digits and inserted digits must form a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message center switch to a served user switch) then every lamp update message places the `Inserted Digits` field (from the **MESSAGE WAITING INDICATION SUBSCRIBER NUMBER PREFIXES** form) in front of the AUDIX subscriber number to form a complete QSIG network number for the served user.

NOTE:
For Feature Plus MWI only. The routing digits and inserted digits must form a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (Served User) switch. The Inserted Digits field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Type a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the `Extension Length` field on the `SWITCH INTERFACE ADMINISTRATION` form of AUDIX.

Five EPN maximum in MCC1 Media Gateways

system-parameters customer-options

This form shows you which optional features are enabled for your system as determined by the installed license file. If you have any questions about disabling or enabling one of these features contact your Avaya representative.

Field descriptions for page 3

```
    display system-parameters customer-options                    (page 3)
                                OPTIONAL FEATURES

  Emergency Access to Attendant?                      ISDN Feature Plus?
          Enable 'dadmin' login?           ISDN Network Call Redirection?
          Enhanced Conferencing?                        ISDN-BRI Trunks?
                  Enhanced EC500?                               ISDN-PRI?
        Extended Cvg/Fwd Admin?                    Local Spare Processor?
    External Device Alarm Admin?                    Malicious Call Trace?
 Five Port Networks Max Per MCC?     Mode Code for Centralized Voice Mail?
               Flexible Billing?
  Forced Entry of Account Codes?                 Multifrequency Signaling?
     Global Call Classification?  Multimedia Appl.Server Interface(MASI)?
             Hospitality (Basic)?        Multimedia Call Handling (Basic)?
Hospitality (G3V3 Enhancements)?      Multimedia Call Handling(Enhanced)?
                   H.323 Trunks?                     Multiple Locations?
                                        Personal Station Access (PSA?)?
          IP Attendant Consoles?
                    IP Stations?
```

Form 38. System Parameters Customer-Options

Five Port Networks Max Per MCC

Available only for DEFINITY R and S8700 Multi-Connect. Allows system administration to create five port networks in a multi-carrier cabinet. If there are any cabinets with more than 2 PNs assigned, this field cannot be set to **n**.

IP loss groups

With the addition of two new IP loss groups (see "IP loss groups" on page 25), the `system-parameters country-options` form has added two new rows and columns.

system-parameters country-options

Field descriptions for page 3

```
display system-parameters country-otpions                      Page 3 of 24
                                2 PARTY LOSS PLAN

    TO:   1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17  18  19
     1:   0   0   0   0   0   0   0   0   0   3   0   0   0   0   0   3   3   0   0
     2:   0   0   0   0   0   0   0   0   0   3   0   0   0   0   0   3   3   0   0
     3:   3  -1   0   0   0   0   0   0   0   3   0   0   0   0   0   3   3   6   6
     4:   0   0  -3   0   0   3   3   3   2   3   0   0   0   0   2   3   3   0   0
     5:   0   0  -3   0   0   3   3   3   2   3   0   0   0   0   0   3   3   0   0
     6:   0   0  -3   3   3   6   8   6   5   5   5   3   3   3   5   3   3   0   0
F    7:   0   0  -3   3   3   8   8   6   5   5   5   3   3   3   5   3   3   0   0
R    8:   0   0  -3   3   3   6   6   6   3   5   3   3   0   0   3   3   3   0   0
O    9:   0   0  -3   2   2   5   5   3   0   0   2  -3  -3  -3   0   3   3   0   0
M   10:   3   3   0   3   3   5   5   5   0   0   3  -3  -3  -3   3   3   3   3   3
    11:   0   0  -3   0   0   5   5   3   2   3   0   0   0  -3   0   3   3   0   0
    12:   6   6   3   6   6   9   9   9   3   3   6   0   0   0   6   3   3   6   6
    13:   6   6   0   6   6   9   9   6   3   3   6   0   0   0   6   3   3   6   6
    14:   6   6   0   6   6   9   9   6   3   3   3   0   0   0   6   3   3   6   6
    15:   0   0  -3   2   0   5   5   3   0   3   0   0   0   0   0   3   3   0   0
    16:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
    17:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
    18:   0   0   0   0   0   0   0   0   0   3   0   0   0   0   0   3   3   0   0
    19:   0   0   0   0   0   0   0   0   0   3   0   0   0   0   0   3   3   0   0
```

Form 39. system-parameters country-options form, page 3

FROM / TO

Display-only fields that identify the variable digital loss values.

Valid entries	Usage
-3 through **15**	An unsigned number is a loss, while a number preceded with a minus sign is a gain.

Field descriptions for page 4

```
display system-parameters country-otpions                          Page 4of 24

                        TONE & CONFERENCE LOSS PLANS

                                        TO
            1  2  3  4  5  6  7  8  9 10 11 12 13 14 15 16 17 18 19
     Dial:  0  3  3  0  0  6  6  6  5  0  6  5  5  5  5  0  0  0  0
  Confirm:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
  Reorder:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
     Busy:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
  Ringing:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Spec Ring:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Intercept:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
  Waiting:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
   Verify:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
  Intrude:  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
      Zip:  3  3  3 -3 -3 -3 -3 -3 -3 -3 -3 -3 -3 -3  0  0  0 -3
    Music:  0  3  3  0  0  6  6  6  3  0  6  3  3  3  3  0  0  0  0

End-to-End total loss (dB) in a n-party conference:
3: 15      4: 21      5: 26      6:29
```

Form 40. system-parameters country-options form, page 4

FROM / TO

Display-only fields that identify the variable digital tone values.

Valid entries	Usage
-3 through **15**	An unsigned number is a loss, while a number preceded with a minus sign is a gain.

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Valid entries	Usage
0 through **99**	The higher the number listed for a call with a fixed display number of parties, the more loss the switch adds into a conference call with that number of parties; therefore, the conference call is quieter.

Location by region

This section shows changes to existing forms and new options for existing fields associated with this development item. The introduction explains why the administrator uses the form, and the table describes the use of each new field or option on the form.

IP-network-region

Field descriptions for page 1

```
change ip-network-region 3                                    Page 1 of 2

                                IP Network Region
      Region: 3        Name: North
    Location: __

                      Intra-region IP-IP Direct Audio? translated(NAT)
AUDIO PARAMETERS      Inter-region IP-IP Direct Audio? native (NAT)
  Codec Set: 2                                        IP Audio Hairpinning? y
   Location: 1
 UPD Port Range                                     RTCP Reporting Enabled? y
          Min: 2048_                             RTCP MONITOR SERVER PARAMETERS
          Max: 3028                               Use Default Server Parameters? n
                                                   Server IP Address: 1 .2 .3 .4
DIFFSERV/TOS PARAMETERS                                          Server Port: 5005
 Call Control PHB Value: 34_                      RTCP Report Period(secs): 5
    Audio PHB Value: 46
          BBE PHB Value: 43                   Resource Reservation Parameters
                                                                 RSVP Enabled? y
                                                        RSVP Refresh Rate(secs): 15
 Call Control 802.1p Priority: 7               Retry upon RSVP Failure Enabled? y
      Audio 802.1p Priority: 6                 RSVP Profile: guaranteed-service
```

Form 41. IP Network Region form

Region

A display-only field indicating the number of the region being administered.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

AUDIO PARAMETERS

Codec Set

Specifies the codec assigned to the region.

Valid entries	Usage
1-7	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Location

Specifies the location by IP network region allowing correct date and time information and trunk routing based on IP network region.

Valid entries	Usage
1-44	(For DEFINITY R, CSI, SI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1-64	(For Avaya S8300 Media Server, Avaya S8700 Multi-Connect, and Avaya S8700 IP-Connect only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN that the endpoint registered through or the media gateway containing the Internal Call Controller or Local Spare Processor on an Avaya S8300 Media Server that the endpoint registered through. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP reporting. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.

UDP Port Range

UPD Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
3-65535	Enter the highest UDP port number to be used for audio transport.

UPD Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
2-65534	Enter the lowest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the Use Default Server Parameters field is set to **n** and the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
5-30	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the Use Default Server Parameters field is set to **n** and the and the RTCP Enabled field is set to **y**

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the Use Default Server Parameters field is set to n and the and the RTCP Enabled field is set to **y**.

Valid entries	Usage
1-65535	Enter the port for the RTCP Monitor server.

Use Default Server Parameters

This field only appears when the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If set to **y**, you must complete the Default Server IP Address field on the IP Options System Parameters screen.
n	If you enter **n**, you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

DIFFSERVE/TOS PARAMETERS

BBE PHB Value

This field contains the Better than Best Effort (BBE) PHB value.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ BBE PHB value.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the 802.1p priority value.

Call Control Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the DiffServ codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control PHB value.

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the Audio 802.1p priority value.

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the Audio PHB codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ Audio PHB value.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the RSVP Enabled field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	
controlled-load	

RSVP Refresh Rate (secs)

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
1-99	Enter the RSVP refresh rate in seconds.

Maximum G700 VAL sources

For S8300 and S8700 Media Servers, field names have been changed on the `system-parameters customer-options` form and the `capacity` form.

system-parameters customer-options

Field descriptions for page 1

```
display system-parameters customer-options                          Page 1 of 9
                              OPTIONAL FEATURES

                                                          Used
      G3 Version: V11                    Maximum Ports: 20000 1166
        Location: 2           Maximum XMOBILE Stations: 0     0
        Platform: 8

IP PORT CABACITIES
                        Maximum Administered IP Trunks: 1000  103
            Maximum Concurrently Registered IP Stations: 5000  505
              Maximum Administered Remote Office Trunks: 1000  0
Maximum Concurrently Registered Remote Office Stations: 1000  0
               Maximum Concurrently Registered IP eCons: 1     0
                     Maximum Administered IP SoftPhones: 2800  48

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                              Maximum TN2501 VAL Boards: 5     1
                               Maximum G700 VAL Sources: 3     2


          (Note: you must logoff & login to effect the permission changes.)
```

Form 42. system-parameters customer-options form, page 1

Maximum TN2501 VAL Boards

Valid entries	Usage
0–10 (DEFINITY R and S8700 Media Servers) **0-5** (DEFINITY SI and CSI, S8100 Media Servers, and S8300 Media Server)	This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system. ■ For values greater than 1, the VAL Full 1-Hour Capacity field on page 4 of the `system-parameters customer-options` form must be set to **y**. ■ This field updates the `System Limit` field on the System Capacity report.

Maximum G700 VAL Sources

Valid entries	Usage
0–250	

capacity

Field descriptions for page 6

```
display capacity                                                      Page 6 of 10
                                  SYSTEM CAPACITY

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

            Recorded Announcement Analog Queue Slots:      2      998    1000
                                  TN2501 VAL Boards:       0        8*     10

         TEMPORARY SIGNALING CONNECTIONS(TSC)
                                  Administered TSCs:       3      125     128
                                      NCA-TSC Calls:       3      253     256
                         TRUNKS
                                  DS1 Circuit Packs:      20      146     166
                         DS1 With Echo Cancellation:       0      166     166
                               ICHT For ISDN Trunks:       0      576     576
                 ISDN CBC Service Selection Trunks:        0      200     200
                                       Trunk Groups:      71      595     666
                                        Trunk Ports:     639     3361    4000

  '*'  Available VAL Boards limited by other inserted integ type annc boards
```

Form 43. capacity form, page 6

No Hold Conference

system-parameters customer-options

NOTE:
On the first OPTIONAL FEATURES form, the G3 Version field must be set to V11 or higher to be able to activate the enhanced conferencing feature options.

Page 3 of the OPTIONAL FEATURES form shows a sample of what the form looks like with the Enhanced Conferencing field included. Your version of the form might be slightly different.

When activated. a user has access to the Conf/Trans Toggle-Swap, Selective Conference Display and Drop, and Meet-me Conference features.

```
change system-parameters customer-options                       Page   3 of   8
                              OPTIONAL FEATURES

   Emergency Access to Attendant? y                          ISDN-BRI Trunks? n
            Enhanced Conferencing? y                                 ISDN-PRI? n
                   Enhanced EC500? n                    Local Spare Processor? n
          Extended Cvg/Fwd Admin? n                      Malicious Call Trace? n
     External Device Alarm Admin? n     Mode Code for Centralized Voice Mail? n
                 Flexible Billing? n
   Forced Entry of Account Codes? n                  Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
              Hospitality (Basic)? y         Multimedia Call Handling (Basic)? n
 Hospitality (G3V3 Enhancements)? n      Multimedia Call Handling (Enhanced)? n
                       IP Trunks? n                         Multiple Locations? n
                                                Personal Station Access (PSA)? n
                    IP Stations? n
               ISDN Feature Plus? n
   ISDN Network Call Redirection? n


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

Form 44. OPTIONAL FEATURES form, page 3

Enhanced Conferencing?

Valid entries	Usage
y/n	Type **y** to turn on the enhanced conferencing feature. Type **n** to turn off the enhanced conferencing feature.

system-parameters features

The `No Hold Conference Timeout` field has been added to page 6 of the `FEATURE-RELATED SYSTEM PARAMETERS` form. This field designates the amount of time (in seconds) before the No Hold Conference call is deactivated if there is no answer.

```
change system-parameters features                              Page   6 of  11
                     FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

               Abort Transfer? n                 No Dial Tone Conferencing? n
       Transfer Upon Hang-Up? n       Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                   Unhold? n
   No Hold Conference Timeout: 60
CPN/ANI/ICLID PARAMETERS

   CPN/ANI/ICLID Replacement for Restricted Calls:
  CPN/ANI/ICLID Replacement for Unavailable Calls:
```

Form 45. FEATURE-RELATED SYSTEM PARAMETERS form, page 6

No Hold Conference Timeout

Valid entries	Usage
20-120	Type the number of seconds before the No Hold Conference call should be deactivated upon no answer. The default value is **60**.

station

The following is a sample of a STATION form with the no-hld-cnf button assigned.

```
change station 4001                                          Page   4 of   5
                                    STATION

FEATURE BUTTON ASSIGNMENTS

 9: conf-dsp
10:
11:
12:
13:
14:
15:
16: no-hld-cnf
```

Form 46. STATION form, page 4

display-messages transfer-conference

The command change display-messages transfer-conference has additional displays for No Hold Conference on pages 3 and 4 of the LANGUAGE TRANSLATION form.

NOTE:
The help messages for this command must be updated accordingly.

```
change display-messages transfer-conference                  Page   3 of   4
                              LANGUAGE TRANSLATIONS

12.     English: Select line ^ to add party.
    Translation: ************************************

13.     English: Select alerting line to answer call.
    Translation: ************************************

14.     English: Transfer canceled.
    Translation: ************************************

15.     English: Connecting to ^.
    Translation: *************************

16.     English: Called party ^ is busy.
    Translation: *************************
```

Form 47. LANGUAGE TRANSLATIONS form, page 3

```
change display-messages transfer-conference                    Page   4 of   4
                              LANGUAGE TRANSLATIONS

17.     English: Invalid number dialed ^.
    Translation: **************************

18.     English: Party ^ is not available.
    Translation: **************************
```

Form 48. LANGUAGE TRANSLATIONS form, page 4

The “^” in messages 15 through 18 will be replaced by a string of digits with a maximum length of 15 digits. Therefore, the number of positions for characters is limited to 26 so that if the “^” is included, there will be room for a 15-digit number.

display-messages softkey-labels

The softkey labels may also be user defined. Your actual display may very slightly based on other feature additions.

```
change display-messages softkey-labels                         Page   1 of   1
                              LANGUAGE TRANSLATIONS

 English    Translation English    Translation English    Translation

  1. Acct    1. *****   17. Drop   17. *****   33. RngOf  33. *****
  2.  AD     2. *****   18. Excl   18. *****   34.  SAC   34. *****
  3. AdBsy   3. *****   19. FMute  19. *****   35. SFunc  35. *****
  4. Admin   4. *****   20. GrpPg  20. *****   36. Spres  36. *****
  5. AutCB   5. *****   21. HFAns  21. *****   37. Stats  37. *****
  6. BtnVu   6. *****   22. IAuto  22. *****   38. Stop   38. *****
  7. CFrwd   7. *****   23. IDial  23. *****   39. Swap   39. *****
  8. CnfDs   8. *****   24. Inspt  24. *****   40. Timer  40. *****
  9. CnLWC   9. *****   25. Last   25. *****   41. TmDay  41. *****
 10. Cnslt  10. *****   26.  LWC   26. *****   42. View   42. *****
 11. Count  11. *****   27. Mark   27. *****   43. Wait   43. *****
 12. CPark  12. *****   28. NHCnf  28. *****   44. WspAn  44. *****
 13. CPkUp  13. *****   29. Pause  29. *****   45. WspPg  45. *****
 14. CTime  14. *****   30. PCall  30. *****
 15.  Dir   15. *****   31. Prog   31. *****
 16. DPkUp  16. *****   32. RmBsy  32. *****
```

Form 49. LANGUAGE TRANSLATIONS form, page 1

display-messages view-buttons

New entries are required on page 9 of the `display-messages view-buttons` form to add the user-defined display option for the No Hold Conference feature. Your actual display may very slightly based on other feature additions.

```
change display-messages view-buttons                          Page   9 of   9
                              LANGUAGE TRANSLATIONS

English                              Translation

 1.  Conf/Trans Toggle-Swap           1. ************************
 2.  Conference Display               2. ************************
 3.  No Hold Conference               3. ************************
 4.  Far End Mute                     4. ************************
```

Form 50. LANGUAGE TRANSLATIONS form, page 9

display-messages button-labels

This shows the layout of page 7 of the `display-messages button-labels` form. Your actual display may very slightly based on other feature additions.

```
display display-messages button-labels                        Page   7 of 10
                              LANGUAGE TRANSLATIONS

English                              Translation

 1. MM Basic                          1. *************
 2. MM Call                           2. *************
 3. MM Call Fwd                       3. *************
 4. MM Data Cnf                       4. *************
 5. MM Mult Nbr                       5. *************
 6. MM PC Audio                       6. *************
 7. Msg                               7. *****
 8. Msg Retrieve                      8. *************
 9. MsgW                              9. *****
10. MsgWaitAct                       10. *************
11. MsgWaitDeact                     11. *************
12. MST Debug                        12. *************
13. Next                             13. *************
14. Night Service                    14. *************
15. No Hold Conf                     15. *************
```

Form 51. LANGUAGE TRANSLATIONS form, page 7

Posted Messages

system-parameters customer-options

NOTE:
On the first OPTIONAL FEATURES form, the G3 Version field must be set to V11 or higher to be able to activate the Posted Messages feature.

Page 3 of the OPTIONAL FEATURES form shows a sample of what the form should look like. Your the form might be slightly different. The important point is that the feature names are sorted in alphabetical order.

```
change system-parameters customer-options                       Page   3 of   9
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                         ISDN-BRI Trunks? n
                Enhanced EC500? n                                 ISDN-PRI? n
         Extended Cvg/Fwd Admin? n                      Malicious Call Trace? n
     External Device Alarm Admin? n     Mode Code for Centralized Voice Mail? n
              Flexible Billing? n
   Forced Entry of Account Codes? n                 Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y         Multimedia Call Handling (Basic)? n
 Hospitality (G3V3 Enhancements)? n      Multimedia Call Handling (Enhanced)? n
                   H.323 Trunks? n                       Multiple Locations? n
                                              Personal Station Access (PSA)? n
           IP Attendant Consoles? n                        Posted Messages? y
                     IP Stations? n
               ISDN Feature Plus? n
   ISDN Network Call Redirection? n


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

Form 52. OPTIONAL FEATURES form, page 3

Posted Messages

Valid entries	Usage
y/n	Type **y** to turn on the Posted Messages feature. Type **n** to turn off the Posted Messages feature.

display-messages view-buttons

The Posted Messages entry is visible on page 9 of the LANGUAGE TRANSLATIONS form. This entry is administrable even if the Posted Messages option is not activated on the OPTIONAL FEATURES form. Your actual display may very slightly based on other feature additions.

```
change display-messages view-buttons                          Page   9 of   9
                            LANGUAGE TRANSLATIONS

English                             Translation

 1.  Station Lock                     1. ************************
 2.  License Error                    2. ************************
 3.  Conference Display               3. ************************
 4.  Conf/Trans Toggle-Swap           4. ************************
 5.  Posted Messages                  5. ************************
```

Form 53. LANGUAGE TRANSLATIONS form, page 9

display-messages softkey-labels

The PoMsg entry is visible on page 1 of the LANGUAGE TRANSLATIONS form. This entry is administrable even if the Posted Messages option is not activated on the OPTIONAL FEATURES form. Your actual display may very slightly based on other feature additions.

```
change display-messages softkey-labels                          Page   1 of   1
                            LANGUAGE TRANSLATIONS

English    Translation English    Translation English    Translation

 1. Acct    1. *****   17. Drop   17. *****   33.  SAC   33. *****
 2.  AD     2. *****   18. Excl   18. *****   34. SFunc  34. *****
 3. AdBsy   3. *****   19. GrpPg  19. *****   35. Spres  35. *****
 4. Admin   4. *****   20. HFAns  20. *****   36. Stats  36. *****
 5. AutCB   5. *****   21. IAuto  21. *****   37. Stop   37. *****
 6. BtnVu   6. *****   22. IDial  22. *****   38. Swap   38. *****
 7. CFrwd   7. *****   23. Inspt  23. *****   39. Timer  39. *****
 8. CnfDs   8. *****   24. Last   24. *****   40. TmDay  40. *****
 9. CnLWC   9. *****   25.  LWC   25. *****   41. View   41. *****
10. Cnslt  10. *****   26. Mark   26. *****   42. Wait   42. *****
11. Count  11. *****   27. Pause  27. *****   43. WspAn  43. *****
12. CPark  12. *****   28. PCall  28. *****   44. WspPg  44. *****
13. CPkUp  13. *****   29. PoMsg  29. *****
14. CTime  14. *****   30. Prog   30. *****
15.  Dir   15. *****   31. RmBsy  31. *****
16. DPkUp  16. *****   32. RngOf  32. *****
```

Form 54. LANGUAGE TRANSLATIONS form, page 1

display-messages button-labels

The Posted MSGs entry is visible on page 7 of the LANGUAGE TRANSLATIONS form. Your version of the form might be slightly different. The important point is that the entries are sorted in alphabetical order.

```
change display-messages button-labels                          Page 7 of 9
                                  LANGUAGE TRANSLATIONS

English                           Translation
 1. NoAnsAlrt                      1. **********
 2. OffBoardAlarm                  2. **********
 3. PMS Failure                    3. **********
 4. PMS Ptr Alarm                  4. **********
 5. Posted MSGs                    5. **********
 6. Priority Call                  6. **********
 7. QueueCall                      7. **********
 8. QueueTime                      8. **********
 9. Release                        9. **********
10. RemBusyInd                    10. **********
11. ResetAlert                    11. **********
12. Ringer Off                    12. **********
13. Ring Stat                     13. **********
14. RSVN Halt                     14. **********
15. SD                            15. **
16. SendAllCalls                  16. **********
```

Form 55. LANGUAGE TRANSLATIONS form, page 7

feature-access-codes

The activation and deactivation feature access codes for the Posted Messages feature can be administered on the `FEATURE ACCESS CODE (FAC)` form on page 3. The new entries are not shown if the `Posted Messages` option is **n** on the `system-parameters customer-options` form.

```
change feature-access-codes                                  Page   3 of   4
                           FEATURE ACCESS CODE (FAC)
                    Posted Messages Activation:        Deactivation:
                  Priority Calling Access Code:
                           Program Access Code:
       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation:
                     Send All Calls Activation:        Deactivation:
                       Station Lock Activation:        Deactivation:
     Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                  Remove:
  Station User Button Ring Control Access Code:
             Terminal Dial-Up Test Access Code:
Terminal Translation Initialization Merge Code:          Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:        Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code:
```

Form 56. FEATURE ACCESS CODE (FAC) form, page 3

status station

On page 1 of the GENERAL STATUS form, the Posted Messages Activated? field has a status of either **y** or **n**. The Message No. field shows the message number of the activated posted message.

```
status station xxxx                                          Page   1 of   2
                            GENERAL STATUS
     Administered Type:                      Service State:
        Connected Type:                    Download Status:
             Extension:                      SAC Activated?
                  Port:                   User Cntrl Restr:
           Call Parked?                  Group Cntrl Restr:
      Ring Cut Off Act?                 CF Destination Ext:
Active Coverage Option:          Posted Messages Activated?    Message No.:


       Message Waiting:
   Connected Ports:


              ACD STATUS
Grp/Mod Grp/Mod Grp/Mod Grp/Mod Grp/Mod
   /       /       /       /       /           HOSPITALITY STATUS
   /       /       /       /       /         Awaken at:
   /       /       /       /       /          User DND:
   /       /       /       /       /         Group DND:
  On ACD Call?      Occupancy:              Room Status:
```

Form 57. GENERAL STATUS form, page 1

Posted Messages Activated?

Valid entries	Usage
y/n	Type **y** to activate the Posted Messages feature on this station. Type **n** to deactivate the Posted Messages feature on this station.

Message No.

NOTE:
If the Posted Messages feature is not activated for the station, the Message No. field is not shown.

Valid entries	Usage
1-30	Type the number of the message (see page 85).

Resource Reservation Protocol (RSVP)

IP network region

Use this screen to pool devices with LAN-like connectivity. The first page is used to modify the audio and QoS settings. The `Codec Set` field on this page reflects the CODEC set that must be used for connections between phones within this region or between phones and MedPro/Prowler boards within this region.

```
change ip-network-region 1                                    Page 1 of 3
                              IP NETWORK REGION

      Region: 1                Name: region 1
    Location: 1
                               Intra-region IP-IP Direct Audio: y
AUDIO PARAMETERS               Inter-region IP-IP Direct Audio: y
   Codec Set: 1                           IP Audio Hairpinning? n
  UDP Port Range
         Min: 2048                      RTCP Reporting Enabled? y
         Max: 3028               RTCP MONITOR SERVER PARAMETERS
                                  Use Default Server Parameters? n
                                              Server IP Address: 123-45-67-89
DIFFSERV/TOS PARAMETERS                             Server Port: 5005
 Call Control PHB Value: 46            RTCP Report Period(secs): 5
        Audio PHB Value: 46
                              AUDIO RESOURCE RESERVATION PARAMETERS
                                                  RSVP Enabled? y
          802.1P/Q PARAMETERS                RSVP Refresh Rate(secs): 5
 Call Control 802.1p Priority: 7     Retry upon RSVP Failure Enabled? n
        Audio 802.1p Priority: 6                   RSVP Profile: guaranteed-service
                                   RSVP unreserved (BBE) PHB Value: 46
```

Form 58. IP network region form, page 1

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

Region

A display-only field indicating the number of the region being administered.

Audio Parameters

Codec Set

Specifies the codec assigned to the region.

Valid entries	Usage
1-7	Type the number for the codec set for the region.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Type **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Type **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Location

Specifies the location by IP network region allowing correct date and time information and trunk routing based on IP network region.

Valid entries	Usage
1-44	(For DEFINITY R, CSI, SI only.) Type the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1-64	(For Avaya S8300 Media Server, Avaya S8700 Multi-Connect, and Avaya S8700 IP-Connect only.) Type the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN that the endpoint registered through or the media gateway containing the Internal Call Controller or Local Spare Processor on an Avaya S8300 Media Server that the endpoint registered through. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.

RTCP Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.

UDP Port Range

UPD Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
3-65535	Type the highest UDP port number to be used for audio transport.

UPD Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
2-65534	Type the lowest UDP port number to be used for audio transport.

RTCP Monitor Server Parameters

RTCP Report Period (secs)

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
5-30	Type the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
0-255 (in `nnn.nnn.nnn.nnn` format)	Type the IP address for the RTCP Monitor server.

Server Port

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
1-65535	Type the port for the RTCP Monitor server.

Use Default Server Parameters

This field only appears when the RTCP Enabled field is set to **y**.

Valid entries	Usage
y	Type **y** to use the default RTCP Monitor server parameters as defined on the IP OPTIONS SYSTEM PARAMETERS screen. If set to **y**, you must complete the Default Server IP Address field on the IP OPTIONS SYSTEM PARAMETERS screen.
n	If you type **n**, you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

DiffServ/TOS Parameters

BBE PHB Value

This field contains the Better than Best Effort (BBE) PHB value.

Valid entries	Usage
0-63	Type the decimal equivalent of the DiffServ BBE PHB value.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the 802.1p priority value.

Call Control Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the DiffServ codepoint.

Valid entries	Usage
0-63	Type the decimal equivalent of the Call Control PHB value.

VoIP Media 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the Audio 802.1p priority value.

VoIP Media PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the Audio PHB codepoint.

Valid entries	Usage
0-63	Type the decimal equivalent of the DiffServ Audio PHB value.

Resource Reservation Parameters

Retry upon RSVP Failure Enabled

This field only appears if the `RSVP Enabled` field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the `RSVP Enabled` field is set to **y**. You set this field to what you have configured on your network.

Valid entries
guaranteed-service
controlled-load

RSVP Refresh Rate (secs)

This field only appears if the `RSVP Enabled` field is set to **y**.

Valid entries	Usage
1-99	Type the RSVP refresh rate in seconds.

RTCP Monitor Server Parameters

Use Default Server Parameters

This field only appears when the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
y/n	Type **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters form. If you type **n**, then you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

Server IP Address

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	Type the IP address for the RTCP Monitor server.

Server Port

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
1-65535	Type the port for the RTCP Monitor server.

RTCP Report Period (secs)

This field only appears when the `Use Default Server Parameters` field is set to **n** and the and the `RTCP Enabled` field is set to **y**.

Valid entries	Usage
5-30	Type the report period for the RTCP Monitor server in seconds.

Resource Reservation Parameters

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Refresh Rate (secs)

This field only appears if the `RSVP Enabled` field is set to **y**.

Valid entries	Usage
1-99	Type the RSVP refresh rate in seconds.

Retry upon RSVP Failure Enabled

This field only appears if the `RSVP Enabled` field is set to **y**.

Valid entries	Usage
y/n	Specifies whether you to enable retries when RSVP fails.

RSVP Profile

This field only appears if the `RSVP Enabled` field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	
controlled-load	

S8300 Media Server

This section shows changes to existing forms and new options for existing fields associated with this development item. The introduction explains why the administrator uses the form, and the table describes the use of each new field or option on the form.

IP-network-region

Field descriptions for page 1

```
change ip-network-region 3                                      Page 1 of 2

                                IP Network Region
      Region: 3        Name: North
    Location: __

                      Intra-region IP-IP Direct Audio? translated(NAT)
AUDIO PARAMETERS      Inter-region IP-IP Direct Audio? native (NAT)
  Codec Set: 2                                        IP Audio Hairpinning? y
   Location: 1
 UPD Port Range                                      RTCP Reporting Enabled? y
          Min: 2048_                             RTCP MONITOR SERVER PARAMETERS
          Max: 3028                               Use Default Server Parameters? n
                                                   Server IP Address: 1 .2 .3 .4
DIFFSERV/TOS PARAMETERS                                        Server Port: 5005
 Call Control PHB Value: 34_                     RTCP Report Period(secs): 5
    Audio PHB Value: 46
          BBE PHB Value: 43                   Resource Reservation Parameters
                                                              RSVP Enabled? y
                                                     RSVP Refresh Rate(secs): 15
 Call Control 802.1p Priority: 7                Retry upon RSVP Failure Enabled? y
      Audio 802.1p Priority: 6                  RSVP Profile: guaranteed-service
```

Form 59. IP Network Region form

Region

A display-only field indicating the number of the region being administered.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

AUDIO PARAMETERS

Codec Set

Specifies the codec assigned to the region.

Valid entries	Usage
1-7	Enter the number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a region.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.
native(NAT)	Enter **native(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT).
translated(NAT)	Enter **translated(NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Location

Specifies the location by IP network region allowing correct date and time information and trunk routing based on IP network region.

Valid entries	Usage
1-44	(For DEFINITY R, CSI, SI only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1-64	(For Avaya S8300 Media Server, Avaya S8700 Multi-Connect, and Avaya S8700 IP-Connect only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN that the endpoint registered through or the media gateway containing the Internal Call Controller or Local Spare Processor on an Avaya S8300 Media Server that the endpoint registered through. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.

RTCP Reporting Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP reporting. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.

UDP Port Range

UPD Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
3-65535	Enter the highest UDP port number to be used for audio transport.

UPD Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
2-65534	Enter the lowest UDP port number to be used for audio transport.

RTCP MONITOR SERVER PARAMETERS

RTCP Report Period (secs)

This field only appears when the Use Default Server Parameters field is set to **n** and the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
5-30	Enter the report period for the RTCP Monitor server in seconds.

Server IP Address

This field only appears when the Use Default Server Parameters field is set to **n** and the and the RTCP Enabled field is set to **y**

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the Use Default Server Parameters field is set to n and the and the RTCP Enabled field is set to **y**.

Valid entries	Usage
1-65535	Enter the port for the RTCP Monitor server.

Use Default Server Parameters

This field only appears when the RTCP Reporting Enabled field is set to **y**.

Valid entries	Usage
y	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If set to **y**, you must complete the Default Server IP Address field on the IP Options System Parameters screen.
n	If you enter **n**, you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

DIFFSERVE/TOS PARAMETERS

BBE PHB Value

This field contains the Better than Best Effort (BBE) PHB value.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ BBE PHB value.

Call Control 802.1p Priority

Provides Layer 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the 802.1p priority value.

Call Control Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the DiffServ codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control PHB value.

Audio 802.1p Priority

Provides Lay 2 priority for Layer 2 switches.

Valid entries	Usage
0-7	Specifies the Audio 802.1p priority value.

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the Audio PHB codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ Audio PHB value.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
y/n	Specifies whether to enable retries when RSVP fails.

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Profile

This field only appears if the RSVP Enabled field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	
controlled-load	

RSVP Refresh Rate (secs)

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
1-99	Enter the RSVP refresh rate in seconds.

Timed automatic disconnect for outgoing trunk calls

Class of Restriction

```
change cor 1                                                  Page 2 of 4
                            CLASS OF RESTRICTION

                  MF Incoming Call Trace? n
              Brazil Collect Call Blocking? n
                    Block Transfer Display? n



                          Station Lock COR: 1
Outgoing Trunk Disconnect Timer (minutes):
```

Form 60. Class of Restriction (COR) form, page 2

Outgoing Trunk Disconnect Timer (minutes)

Valid entries	Usage
blank	Default value. The feature is disabled and the trunk will not be automatically disconnected.
2 – 999	The number of minutes before an outgoing trunk is automatically disconnected (dropped). The outgoing trunk disconnect timer should be administered to a value large enough to provide users with adequate response time.

NOTE:
This feature applies to all outgoing trunk calls initiated by a party belonging to a specified Class of Restriction (COR). Prior to disconnecting the trunk, warning tones are applied to all parties on the call. The first warning tone occurs when one minute remains on the call. The second warning tone occurs when 30 seconds remain on the call.

VLAN administration

To avoid confusion when administering virtual local area networks (VLAN), VLAN administration has been split into two separate areas:

- VLAN administration for boards
- VLAN administration for phones

The VLAN administration — specifically, the `802.1Q VLAN` field — has been removed from the `ip-network-region` form. In its place, VLAN administration has been moved to two separate forms:

- `ip-interfaces` form (for VLAN board administration)
- `ip-network-map` form (for VLAN phone administration)

IP-interfaces

```
change ip-interfaces                                           Page 1 of 4

                               IP INTERFACES

Inter-region IP connectivity allowed?
                                                                  Net
ON Type   Slot  Code Sfx Node Name       Subnet Mask   Gateway Address Rgn VLAN
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
n  ______ _____          _______________ 255.255.255.0 ___.___.___.___ ___ ____
```

Form 61. IP INTERFACES form

VLAN

Valid entries	Usage
0-4095	Specifies the 802.1Q virtual LAN value. The default value is zero (0).

The VLAN field has the same allowed values, default, help, and error messages as it had on the ip-network-region form.

The use of the VLAN field, however, is different on the ip-interfaces form than it was on the ip-network region form. The VLAN field on the ip-interfaces form sends VLAN instructions to the CLAN, processor CLAN function, or Media Processor boards. It does not send VLAN instructions to IP endpoints, such as IP telephones and softphones.

IP-network-map

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

                          IP PARAMETERS BY IP ADDRESS

                                      Subnet          802.1Q
FROM IP Address   (TO IP Address  or Mask)  Region   VLAN
  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
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
___.___.___.___   ___.___.___.___    __      ___      ____
```

Form 62. IP NETWORK MAP form

802.1Q VLAN

Valid entries	Usage
0-4095	Specifies the 802.1Q virtual LAN value. The default value is zero (0).

The `802.1Q VLAN` field specifies the 802.1Q virtual LAN value, and appears only if 802.1p/Q is enabled. The `802.1Q VLAN` field has the same allowed values, default, help, and error messages as it had on the `ip-network-region` form.

The use of the `802.1Q VLAN` field, however, is different on the `ip-network-map` form than it was on the `ip-network region` form. The `802.1Q VLAN` field on the `ip-interfaces` form sends VLAN instructions to IP endpoints such as IP telephones and softphones. It does not send VLAN instructions to the CLAN, processor CLAN function, or Media Processor boards.

New and changed commands

4

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

New commands

2420 DCP telephones

display-messages button-labels (language translations)

action	object	qualifier
change	display-message view-buttons	
display	display-message view-buttons	

tftp-server

action	object	qualifier
change	tftp-server	
display	tftp-server	

4620 IP telephones

display-messages button-labels (language translations)

action	object	qualifier
change	display-message view-buttons	
display	display-message view-buttons	

CLAN QoS and CIDR support

The output of two commands now contain new columns of Subnet Mask data in Communication Manager. The syntax for the two commands are unchanged.

netstat ip-route

The **netstat ip-route** lists IP routes from all circuits.

netstat ip-route board *<board location>*

The **netstat ip-route board *<board location>*** lists routes from a specified circuit pack.

The routes shown in netstat ip-route command output are obtained directly from the circuit packs using SNMP queries. All administered routes in the switch can be seen using the **list ip-route** command and **status-link** together.

```
netstat ip-route board 01C07                                    page 1 of 1

                                    IP ROUTING

Bd/Pt     Destination     Gateway          Subnet Mask         Interface

01C0717   0.0.0.0         135.9.77.254     255.255.255.255       cpm0
01C0717   135.9.77.0      135.9.77.88      255.255.255.0         cpm0
01C0717   135.9.193.254   135.9.77.88      255.255.254.0         cpm0
01C0711   192.255.255.2   192.255.255.1    255.255.255.255       ppp10
01C0711   192.255.255.17  192.255.255.2    255.255.255.255       ppp10
01C0718   127.0.0.1       127.0.0.1 255.   255.255.255           lo0
```

Form 63. netstat ip-route command form

Co-resident DLG

crm-features

Action	Object	Qualifier
list	crm-features	

cti-link

Action	Object	Qualifier
add	cti-link	<link number> (or ‘next’) (DEFINITY R, CSI, SI, S8700 IP-Connect)
busyout	cti-link	<link number> (DEFINITY R, CSI, SI)
change	cti-link	<link number> (DEFINITY R, CSI, SI, S8700 IP-Connect)
display	cti-link	<link number> [‘count’ n] [‘print’ or ‘schedule’] (DEFINITY R)
display	cti-link	<link number> [‘print’ or ‘schedule’] (DEFINITY CSI, SI)
list	cti-link	[cti-link number 1-MAX], [‘count’ n], [print or schedule] (DEFINITY R, CSI, SI)
list	cti-link	[cti-link number 1-MAX], [‘count n], [‘schedule’] (S8700 IP-Connect, S8700 Multi-Connect)
list	usage cti-link	[cti-link number 1-MAX], [‘print’ or ‘schedule’] (DEFINITY CSI, SI)
list	usage cti-link	1-MAX (not for DEFINITY R, S8700 IP-Connect)
remove	cti-link	<link number> (DEFINITY R, CSI, SI, S8700 IP-Connect)
test	cti-link	<link number> [‘long’ or ‘short’] [‘repeat’ (1-100) or ‘clear’] (DEFINITY CSI, SI)

dlg cti-link

Action	Object	Qualifier
status	dlg cti-link	[‘print’]

dlg interface

Action	Object	Qualifier
status	dlg interface	[‘print’]

usage extension

Action	Object	Qualifier
list	usage extension	<extension number> [‘vector’], [‘print’ or ‘schedule’] (DEFINITY CSI, SI)

Conference/transfer enhancements

list meet-me-vdn

You can list Meet-me Conference VDNs as shown in the following example:

```
list meet-me-vdn

                          MEET-ME VECTOR DIRECTORY NUMBERS


                                    Access          Vec  Control
Name                          Ext    Code    COR TN  Num    Ext

Secure Meet-me Conference     4000     *       1   1   1
Nonsecure Meet-me Conference  4006             1   1   2    84590
```

If the Access Code field shows an asterisk (*), an access code is assigned. If the Access Code field is blank, no access code is assigned. The access code is displayed for administrators with super-user permissions (such as *init*).

status meet-me-vdn

The status of a Meet-me Conference VDN can be displayed as shown in the following example. In this example, there are three parties connected to the Meet-me Conference call.

```
status meet-me-vdn 4003                                    Page   1 of   1
                            GENERAL STATUS
                                        Service State: active

             Extension: 4003









   Connected Ports:  01A10002  03B08013  05D18009
```

In this example, the Meet-me Conference VDN is administered, but there are no parties active on a call.

```
status meet-me-vdn 4003                                    Page   1 of   1
                            GENERAL STATUS
                                        Service State: idle

             Extension: 4003









   Connected Ports:
```

reset meet-me-vdn

A Meet-me Conference VDN can be reset using the **rest** command. When reset, any conference callers are dropped from the conference call, and the VDN returns to the idle state. This can be done, for example, if the administrator suspects that an unauthorized user is using the Meet-me Conference feature.

The syntax for the command is as follows (***xxxxx*** is the VDN):

- **reset meet-me-vdn *xxxxx***

Dial plan expansion (DPE)

dialplan analysis

Action	Object	Qualifier
change	dialplan analysis	—
display	dialplan analysis	[‘print’ or ‘schedule’]

dialplan parameters

Action	Object	Qualifier
change	dialplan parameters	—
display	dialplan parameters	[‘print’ or ‘schedule’]

uniform-dialplan

Action	Object	Qualifier
change	uniform-dialplan	Type 1-7 digits between 0-9
display	uniform-dialplan	Type 1-7 digits between 0-9 [‘print’ or ‘schedule’]
list	uniform-dialplan	[start *digits*] [len *length*] [insert *digits*] [net *network*] [node *node number*] [to-node *node number*] [count *number*] [‘print’ or ‘schedule’]

IP telephones

NOTE:
This new command is not related to a specific feature.

To apply a firmware fix to all IP telephones at a site, you have to get the phones to reboot. This forces the phones to download new firmware from the TFTP server. The only way to force all IP telephones to reboot is to either busyout each phone one extension at a time, or reset system 2.

A new command, **reset ip-station**, has been created to allow the administrator to be able to unregister all the H.323 stations, IP softphones, and IP telephones in a system. This feature allows the administrator to unregister and re-register their IP telephones, allowing the phones to download new firmware.

The **reset ip-station** command causes the switch to send a message to each IP telephone, telling that IP telephone to reset. After sending the message, the switch unregisters the station. That guarantees that even if the reset message were dropped by the LAN, the set would still be unregistered.

This command runs in the background. When the command is submitted at the SAT, the system responds immediately: "Command completed successfully." That does not mean that all IP telephones have already been reset. It means that the system has started the background process of resetting them.

reset ip-station

Action	Object	Qualifier
reset	ip-station	network region

Changed commands

2420 DCP telephones

terminal-parameters

action	object	qualifier
change	terminal-parameters 6400/607A1/4600/2420	
display	terminal-parameters 6400/607A1/4600/2420	

Note that the change terminal-parameters command requires at least craft login permissions.

Conference/transfer enhancements

display capacity

The Meet-me Conference VDNs are displayed as shown in the following example:

```
display capacity                                              Page   3 of  10
                                   SYSTEM CAPACITY

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

                    CALL COVERAGE
                            Coverage Answer Groups:      0     750      750
                                    Coverage Paths:      3    9996     9999

                                Call Pickup Groups:      0    5000     5000

                                      Call Records:      -       -     7712

           CALL VECTORING/CALL PROMPTING
                    Total Vector Directory Numbers:  19321     679    20000
                Meet-me Conference VDNs per system:      0    1800     1800
                                Vectors Per System:    231     768      999
         BSR Application-Location Pairs Per System:      1     999     1000
```

list usage extension

When listing extension usage, Meet-me Conference VDNs will display that information as shown in the following example:

```
list usage extension 36090

                             LIST USAGE REPORT

      Used By
      VDN - Meet-me Conf   VDN Number           36090
```

list usage vector

When listing vector usage, Meet-me Conference VDNs using a particular vector will display that information as shown in the following example:

```
list usage vector 12

                          LIST USAGE REPORT

      Used By
      Vector              Vector Number       2                 Step 4
      Vector              Vector Number       43                Step 4
      Vector              Vector Number       78                Step 4
      VDN                 VDN Number          25002
```

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