Highlights of

Avaya MultiVantage™ Software

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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.

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.

Voice Over Internet Protocol (VoIP)

If the equipment supports Voice over Internet Protocol (VoIP) facilities, you may experience certain compromises in performance, reliability and security, even when the equipment performs as warranted. These compromises may become more acute if you fail to follow Avaya's recommendations for configuration, operation and use of the equipment. YOU ACKNOWLEDGE THAT YOU ARE AWARE OF THESE RISKS AND THAT YOU HAVE DETERMINED THEY ARE ACCEPTABLE FOR YOUR APPLICATION OF THE EQUIPMENT. YOU ALSO ACKNOWLEDGE THAT, UNLESS EXPRESSLY PROVIDED IN ANOTHER AGREEMENT, YOU ARE SOLELY RESPONSIBLE FOR (1) ENSURING THAT YOUR NETWORKS AND SYSTEMS ARE ADEQUATELY SECURED AGAINST UNAUTHORIZED INTRUSION AND (2) BACKING UP YOUR DATA AND FILES.

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.

The equipment described in this manual complies with standards of the following organizations and laws, as applicable:

- Australian Communications Agency (ACA)
- American National Standards Institute (ANSI)
- Canadian Standards Association (CSA)
- Committee for European Electrotechnical Standardization (CENELEC) – European Norms (EN's)
- Digital Private Network Signaling System (DPNSS)
- European Computer Manufacturers Association (ECMA)
- European Telecommunications Standards Institute (ETSI)
- FCC Rules Parts 15 and 68
- International Electrotechnical Commission (IEC)
- International Special Committee on Radio Interference (CISPR)
- International Telecommunications Union - Telephony (ITU-T)
- ISDN PBX Network Specification (IPNS)
- National ISDN-1
- National ISDN-2
- Underwriters Laboratories (UL)

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 Laser products, equipment classification and requirements:

- IEC 60825-1, 1.1 Edition
- 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

Electromagnetic Compatibility (EMC) Standards

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

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

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

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

Federal Communications Commission Statement

Part 15:

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

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

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

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

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

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

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.

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 table.

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
CO trunk	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, 1KN, 1SN	6.0F	RJ48C, RJ48M
120A2 channel service unit	04DU9-DN	6.0Y	RJ48C

If the terminal equipment (for example, the MultiVantage™ Solution equipment) 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.

It is recommended that repairs be performed by Avaya certified technicians.

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

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

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003.
Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This digital apparatus does not exceed Class A limits for radio noise emission set out in the radio interference regulation of the Canadian Department of Communications.

Le Présent Appareil Nomérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils manicures de la class A préscrites dans le reglement sur le brouillage radioélectrique édicté par le ministére des Communications 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://support.avaya.com/elmodocs2/DoC/SDoC/index.jhtml/

All MultiVantage™ system products 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

CE

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) signed by the Vice President of MultiVantage™ Solutions research and development, Avaya Inc., can be obtained by contacting your local sales representative and are available on the following Web site:
http://support.avaya.com/elmodocs2/DoC/IDoC/index.jhtml/

Japan

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

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

Network Connections

Digital Connections - The equipment described in this document can be connected to the network digital interfaces throughout the European Union.
Analogue Connections - The equipment described in this document can be connected to the network analogue interfaces throughout the following member states:

Belgium	Germany	Luxembourg
Netherlands	Spain	United Kingdom

LASER Product

The equipment described in this document may contain Class 1 LASER Device(s) if single-mode fiber-optic cable is connected to a remote expansion port network (EPN). The LASER devices operate within the following parameters:

- Maximum power output –5 dBm to -8 dBm
- Center Wavelength 1310 nm to 1360 nm
- CLASS 1 LASER PRODUCT IEC 60825-1: 1998

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

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The following are trademarks or registered trademarks of Avaya Inc.:

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- Cajun®
- CallVisor®
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- DEFINITY®
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- Softconsole™
- Transtalk™
- VisAbility™
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Call: Avaya Publications Center
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Write: Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA
Attention: Avaya Account Management
E-mail: totalware@gwsmail.com

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Welcome

Overview

MultiVantage™ is the centerpiece of Avaya Next Generation Enterprise Class IP Solutions (ECLIPS). Running on a variety of Avaya™ Media Servers and DEFINITY® Servers and providing control to Avaya™ Media Gateways and Avaya™ Communications Devices, MultiVantage can be designed to operate in a Distributed or Networked call processing environment.

Avaya MultiVantage™ Software will carry forward all of cutsomers' current DEFINITY capabilities, plus offer all the enhancements that will enable them to take advantage of new, distributed technologies, increased scalability and redundancy. Avaya MultiVantage evolved from DEFINITY software and delivers no- compromise Enterprise Class IP Solutions.

Avaya MultiVantage 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 document describes the new features and enhancements available with the Avaya MultiVantage™ Software solution, 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 will contain an Avaya™ S8300 Media Server configured as a Local Survivable Processor.

It also contains information about what has changed within the software solution, such as new and changed software screens.

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 sections:

Highlights

Presents short descriptions of each of the new features or changes in the Avaya MultiVantage™ Software solution for this release.

Hardware

Describes hardware changes introduced in the Avaya MultiVantage™ solution.

New and Changed Screens

Provides information about new administration screens, and changes to existing screens.

New and Changed Commands

Provides information about non-administration commands (such as display, list, or status commands) that changed for this release.

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The following are trademarks or registered trademarks of Avaya:

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CentreVu™	CONVERSANT®
DEFINITY®	Intuity™
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- Zydacron registration is pending for Zydacron Corporation.

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1. Access the Avaya web site at **http://www.avaya.com/**
2. Click on Support and then find the latest release of Documentation for your solution.
3. To find a specific book, search for the document number (for example, **555-233-778,** for this book) to view the latest version of the book.

How to Order Documentation

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

This document and any other Avaya documentation can be ordered directly from the Avaya Publications Center toll free at 1-800-457-1235 (voice) and 1-800-457-1764 (fax). Customers outside the United States should use +1.410.568.3680 (voice) and +1.410.891.0207 (fax).

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email: document@avaya.com

fax: 1-303-538-1741 or to your Avaya representative, and mention this document's name and number, ***Highlights of Avaya MultiVantage™ Software***, 555-233-778.

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

How to get help

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For additional support telephone numbers, see the Avaya web site:

http://www.avaya.com

Click on Support, click on Escalation Lists US and International. This web page includes phone numbers for escalation within regions of the United States. For escalation phone numbers outside the United States, click on Global Escalation List. This web page lists the phone numbers for the Centers of Excellence in each Avaya-defined region.

Highlights

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This section presents highlights of features and enhancements added to Avaya MultiVantage™ software, running on Avaya DEFINITY® Servers, as well as the Avaya™ S8000 series Media Servers (with associated Avaya Media Gateways).

General enhancements

This release of Avaya MultiVantage software includes the following general telephony and system-wide enhancements.

Conference/Transfer Enhancements

These are enhancements to telephones for the conference, transfer and hold features that make them easier to use. The enhancements include:

- 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 screen 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.
- Meet-me Conference, which is a special VDN, secured via an access code, and allows a number of parties to be conferenced together with other parties up to the system's Conference limit.

For more information, see the *Administrator's Guide for Avaya MultiVantage Software*.

Dial-Plan Expansion (DPE) Changes

Avaya MultiVantage software provides dial plan expansion to 6 or 7 digits (from 4-/5-digit dial plans). This affects all extensions, including stations, data modules, agent login IDs, vectors, etc. This change increases the total number of extensions that can exist in an Avaya MultiVantage solution. It also allows Avaya servers to participate in networks that already use 6- or 7-digit dial plans, for example, a network of switches made by other vendors. Customers upgrading to Avaya MultiVantage™ software can choose to migrate to the 6-/7-digit dial plan or not. Customers who choose not to migrate during the upgrade process may convert their dial plan to 6/7 digits at a later date.

For more information, see the *Administrator's Guide for Avaya MultiVantage Software*.

Uniform Dial Plan (UDP) Changes

This development modifies the allowed extension length for Uniform Dial Plan (UDP) extensions. Any length of extension from 3 digits to 7 digits and combinations of extension lengths are now allowed.

Users can also replace up to three digits in the UDP tables.

For more information, see the *Administrator's Guide for Avaya MultiVantage Software*.

EC500 feature

EC500 offers users the freedom to work anywhere, anytime, using any type of cellular or wireless phone. With EC500, calls to an office number are extended to a cell phone, allowing users to receive work-related calls wherever the y 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 EC500 cell phone is administered to send office caller ID, the user also has the option of picking up an ongoing EC500 cell phone call on the office phone. All of the new Avaya MultiVantage Solutions offer and fully support this important feature.

The EC500 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. EC500 provides this capability, regardless of the cell phone's Cellular Service Provider or the cellular standard in use.

Leave Word Calling Over QSIG

The "QSIG Leave Word Calling (LWC) MSI and Interworking with DCS+" feature allows users to send and receive LWC messages:

- among Avaya DEFINITY Servers and media servers over a QSIG network
- among Avaya servers over a combination QSIG and DCS+ network
- with Voice Messaging Servers using LWC FAC, and QSIG or Avaya proprietary signaling (adding use over QSIG to that available over DCS).

There is no new administration or hardware required for QSIG LWC MSI and Interworking with DCS+.

This feature maintains current LWC functions in systems that add QSIG to an existing DCS system, and in systems that change from DCS to QSIG. With Avaya MultiVantage software, customers who migrate from DCS to QSIG will retain the LWC feature functionality they currently have.

Analog Busy Automatic Callback without Flash feature

This is a feature 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.

France 25% Trunk Alarming feature

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.

Support Russia DATS/ISDN Network feature

This feature supports ISDN/DATS trunk networks when the tone generator field is set to 15 (Russia) on the system-parameters country-options screen. When the feature is activated, the overlap sending delay and ISDN T302 and T304 timers are modified to support the Russian trunk network.

Capacity Changes

In order to provide customers and Avaya associates with the most up-to-date information, the system capacity limits are not listed in Avaya MultiVantage software documentation; instead, this information will now be available online at http://www.avaya.com/support .

Hardware enhancements

New Avaya Media Servers and Media Gateways

The following hardware products are new components of Avaya MultiVantage Solutions:

- 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).

New Avaya telephones

The following telephones are new components of Avaya MultiVantage Solutions:

- Avaya™ 4620 IP Telephone
- Avaya™ 4602 IP Telephone
- Avaya™ 2420 DCP Telephone.

For more information on these phones, see IP Solution enhancements.

IP Solution enhancements

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 Avaya MultiVantage Solutions with enhanced flexibility in IP addressing and routing, as well as ensuring customer-network compatibility. Quality of Service (QoS) features provide superior call quality for voice-over-IP (VoIP) signaling. Examples of these quality-related standards include 802.1p/Q, VLANs and DiffServ (Differentiated Services).

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.

H.248 Media Gateway Control

New support 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 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. 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.”

Time of Day

Provides users with the capability of synching their Avaya DEFINITY Server or S8000-series Media Server clock(s) with Internet servers providing 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 synched time to the Avaya server.

For more information, see the *Administrator's Guide for Avaya MultiVantage Software* and also the documentation or online help for the software products comprising Avaya VisAbility™ Management Suite.

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 MultiVantage Software*.

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 MultiVantage Software*.

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 Avaya MultiVantage software.

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 voicemail 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 MultiVantage Software*.

Hardware

2

This chapter describes the following hardware changes:

New Avaya S8000 series Media Servers and Media Gateways

Avaya S8300 Media Server and Avaya G700 Media Gateway

The following hardware products are new components of Avaya MultiVantage Solutions:

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

Overview of S8300 Media Server Solutions

The Avaya S8300 Media Server and G700 Media Gateway solution seamlessly delivers a business's voice, fax, and messaging capabilities over an IP network. This unique solution converges the power of the Avaya MultiVantage™ software 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 solution:

- 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. If present, it supports the Avaya MultiVantage software that provides call-processing capabilities for the system. The S8300 can be configured as the primary call controller or as a Local Survivable Processor (LSP) standby server for an S8700 Media Server or for another S8300 Media Server in the configuration.
- Avaya MultiVantage software 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 module.

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 expand the system, or diagnose problems

Avaya S8700 Media Server Configurations

The following hardware products are new components of Avaya MultiVantage Solutions:

- 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- 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 gateway, with the S8300 serving as Local Survivable Processor, rather than primary call controller.

Overview of S8700 IP Connect

The S8700 Media Server for IP Connect Configurations is an all-IP, 19 inch data rack solution that is part of the Avaya Enterprise Class IP Solutions (ECLIPS). 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, this solution saves customers the cost of building a separate telephony network. As an all-IP solution, 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 MultiVantage™ software
- 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* .

Overview of S8700 Multi-Connect

The Avaya™ S8700 Media Server for Multi-Connect Configurations (S8700 Multi-Connect) uses a standard microprocessor engine with an Intel® processor on a commercial server. It provides the building block for a flexible, highly reliable Avaya MultiVantage solution 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 solution.

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* .

New Avaya Telephones

The following phones are new components of Avaya MultiVantage Solutions:

- Avaya™ 4620 IP telephone
- Avaya™ 4602 IP telephone
- Avaya™ 2420 DCP telephone with downloadable firmware.

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

New and Changed Administration Screens

3

New screens

For Avaya MultiVantage™ Solutions, many screens are new or changed.

- For the conference/transfer enhancements, there are no new screens. Please see Changed screens.
- For the dial-plan expansion (DPE) and Uniform Dial Plan (UDP) changes, there are the following three new screens:
  - Dial Plan Analysis Table
  - Uniform Dial Plan Table
  - Dial Plan Parameters screen

  Please also see Changed screens.
- For the analog busy automatic callback without flash feature, there are no new screens. Please see Changed screens.
- For new or enhanced hardware, there are no new screens. Please see Changed screens.
- For CLAN QoS and CIDR support, there are no new screens. Please see Changed screens.
- For location by IP network region, there are no new screens. Please see Changed screens.
- For time of day (TOD) clock synchronization of new and enhanced Avaya MultiVantage solutions, the new/changed screens are within the Avaya VisAbility™ Management Suite. Please see the documentation and online help for the suite's software products for more information about screens.

- For the France 25% Trunk Alarming and Russia DATS/ISDN network features, there are no new or changed screens. Please refer to Chapter 1, ‘‘Highlights’’.
- For the 4620 IP telephone, there is the following new screen:
  — Overview of Display-Messages Button-Labels (Language Translations).
- For the 4602 IP telephones, there are no new screens. Please see Changed screens.
- For the 2420 DCP telephone, there are the following two new screens:
  — TFTP Server Configuration
  — Overview of Display-Messages Button-Labels (Language Translations).
- For IPSI administration, two new screens were added. Among other things, the IPSI administration and system-parameters screens allow you to:
  — ‘‘Add IPSI translations to MultiVantage™ software’’
  — ‘‘Enable control of IPserver interfaces’’.

Overview of new screens for 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 screen. In the new Avaya MultiVantage software and future releases, you use the Dial Plan Analysis Table, the Dial Plan Parameters screen, and the Uniform Dial Plan Table for these tasks.

Dial Plan Analysis Table

The Dial Plan Analysis Table is a new screen that replaces both a Dial Plan Record screen and the Second Digit Table. This screen 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
```

Screen 1. 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, * and #	Enter 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**	Enter the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

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.
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).

Valid entries	Usage
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.
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.

Uniform Dial Plan Table

The Uniform Dialing Plan field must be y on the System-Parameters Customer-Options screen before you can administer this table.

The UDP provides a common 3- 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
```

Screen 2. 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)	Enter the number you want the switch to match to dialed numbers.

Len

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

Del

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

Insert Digits

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

Net

Enter 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 enter **enp**, you must enter the ENP node number in the Node Num field. The Insert Digits field must be blank, and Conv must be **n**.

Conv

Valid entries	Usage
y/n	Enter y to allow additional digit conversion

Node Num

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

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

Dial Plan Parameters screen

The Dial Plan Parameters screen 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
```

Screen 3. Dial Plan Parameters screen

Local Node Number

Enter 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**	Enter 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

Enter 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**	Enter the number of a destination switch.

ETA Routing Pattern

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

Valid entries	Usage
1 – **254**	Enter 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.

Overview of new 2420 phone screens

There are two new screens for the Avaya 2420 IP telephone:

- TFTP-Server
- Display-Messages Button-Labels (Language Translations)

TFTP Server Configuration

This screen 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:
```

Field description

Local Node Name

The local node name must be a valid entry from the IP Node Names screen. The node must be assigned to a CLAN ip-interface or **procr** (processor CLAN).

Valid entries	Usage
1-15 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 screen.

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

TFTP Server Port

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

File to Retrieve

Valid entries	Usage
	Enter 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.

Overview of Display-Messages Button-Labels (Language Translations)

This multi-page screen allows you to define language translations for the 4620 IP and 2420 DCP telephone feature buttons.

Changed screens

For Avaya MultiVantage™ Solutions, many screens changed. This section describes only the new fields or new valid values for each screen that changed for this release of Avaya MultiVantage software.

- For the conference/transfer enhancements and the various telephones that support them, the following screens have changed:
  - — System parameters customer options
  - — Meet-me Conference VDN
  - — Meet-me Conference Call Vector
  - — Display messages
  - — Feature access codes
  - — Language translations - self-administration
  - — Language translations - softkey labels
  - — Language translations - view buttons
  - — Feature-related system parameters.
- Because of the dial plan expansion (DPE), numerous screens have changed to accommodate the wider fields needed for the longer extension numbers. Sometimes this also changed other screen formatting. For more detailed information, see *Administrator's Guide for Avaya MultiVantage Software*.
- For the analog busy automatic callback without flash feature, the following screens have changed:
  - — Feature-Related System Parameters
  - — Station screen.
- For new or enhanced hardware, like the Avaya™ S8300 Media Server with an Avaya™ G700 Media Gateway, the following screen has changed:
  - — IP Network Region screen.
- For C-LAN QoS and CIDR support, the following screens have changed:
  - — IP Routing screen
  - — IP Network Region screen.
- For location by IP network region, the following screen have changed:
  - — IP Network Region screen.
- For the 4620 and 4602 IP telephones, the following screens have changed:
  - — Overview of 4620 phone changed Station screens
  - — Overview of 4602 phone changed Station screens.

- For the 2420 DCP phone, the following screens have changed:
  - — Feature Access Code screen
  - — Feature-Related System Parameters screen
  - — Station screen
  - — Terminal Parameters screen
  - — List usage node-name
  - — List usage ip-address.

Overview of conference enhancements screen changes

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

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

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 screen, the G3 Version field must be set to **V11** or higher (see ''System parameters customer options'' on page 35).

- 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 screen, the G3 Version field must be set to **V11** or higher and the Enhanced Conferencing option must be **y** (see ''System parameters customer options'' on page 35).

System parameters customer options

To enable the Conference Enhancements, the G3 Version field must be set to **V11** on the System-Parameters Customer-Options screen. 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 screen. 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 screen, the G3 Version field is **V11** or higher.

Valid entries	Usage
y/n	Enter **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 screen.

```
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 screen, the Enhanced Conferencing field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

NOTE:
If the VDN extension is part of 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	Enter **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 Vectory Directory Number, 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	Enter 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 screen, 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 screen 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 screens 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 screen, the Enhanced Conferencing field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this vector. If Meet-me Conference 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 screen.

```
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 screen.

```
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. Enter 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.

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: ****
```

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:
```

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.

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.

Overview of Analog Busy Callback changed screens

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

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:
```

Screen 4. Feature-Related System Parameters screen

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	Enter **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
	Enter 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	Enter a voice mail hunt group extension.

Station screen

```
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
```

Screen 5. Station screen

Busy Auto Callback Without Flash

(I don't need to show the field's location. It will be listed by alphabetical order.) Appears only if, on the Feature-Related System Parameters screen, 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	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

Overview of S8300 Media Server changed screens

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

IP Network Region screen

Field descriptions for page 1

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

                                      IP Network Region

      Region: 10
        Name: North

Audio Parameters                 Direct IP-IP Audio Connections? y
   Codec Set: 4                            IP Audio Hairpinning? y
    Location: 3
  UDP Port Range                                   RTCP Enabled? y
         Min: 2048_              RTCP Monitor Server Parameters
         Max: 3028                Use Default Server Parameters? n
                                             Server IP Address: 999.999.999.999
DiffServ/TOS Parameters                            Server Port: 5005
 Call Control PHB Value: 34_          RTCP Report Period(secs): 5
   VoIP Media PHB Value: 46
          BBE PHB Value: 43      Resource Reservation Parameters
                                                     RSVP Enabled? y
            802.1p/Q Enabled? y            RSVP Refresh Rate(secs): 15
Call Control 802.1p Priority: 7    Retry upon RSVP Failure Enabled? y
  VoIP Media 802.1p Priority: 6                RSVP Profile: guaranteed-service
                 802.1Q VLAN: 0
```

Screen 6. IP Network Region screen

Region

Displays the number of the region being administered.

Valid entries	Usage
1-250	Numeric identifier for the region.

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.

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
1-64	(For Avaya S8300 Media Server with an Avaya G700 Media Gateway, Avaya S8700 Media Server for Multi-Connect configuration, and Avaya S8700 Media Server for IP Connect configuration only. The range of valid entries for other Avaya servers will differ.) 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 PPN cabinet. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server with an Avaya G700 Media Gateways all have their locations administered on their corresponding screens.

UDP 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.

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.

DiffServ/TOS Parameters

Call Control PHB Value

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

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control Per-Hop Behavior (PHB) 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	Enter the decimal equivalent of the DiffServ Audio PHB value.

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.

802.1p/Q Enabled

When this field is **y**:

- The `802.1Q VLAN ID` field defaults to zero (0).
- The DiffServ ControlPoint (DSCP) value is taken from the Network Region form.
- The Ethernet frame format changes. If corresponding changes are not made to the attached Ethernet switch, service to the C-LAN is interrupted. Ethernet frames will only be tagged with VLAN and user-priority values if this field is **y**.

For all H.323 call signaling

- Ethernet frames are marked with the user priority configured in the `Call Control 802.1p Priority` field.
- IP packets are marked with the VLAN tag configured in the `802.1Q VLAN` field.

NOTE:
802.1Q is not appropriate for a hub-based network.

Valid entries	Usage
y/n	Enter **y** for 802.1p MAC-layer prioritization and 802.1Q Virtual LAN specification for this region.

Call Control 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**.

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

VoIP Media 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**.

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

802.1Q VLAN

Appears only if the 802.1p/Q Enabled field is **y**.

Valid entries	Usage
0-4095	Specifies the 802.1Q virtual LAN value.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to use the voice channel directly between IP endpoints for audio transmissions, bypassing Avaya servers.

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.

RTCP Reporting 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.

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	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If you enter **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	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.

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	Enter 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	Enter 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	

Overview of CLAN QoS and CIDR changed screens

This section shows changes to existing IP administration screens, 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 screens. 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 screen.

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 screen. The IP Network Region screen 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 screen

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:
```

Screen 7. IP Routing screen

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 MultiVantage Solutions.

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 screen

Field descriptions for page 1

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

                                     IP Network Region
      Region: 10
        Name: North
    Location: 3

             AUDIO PARAMETERS             AUDIO RTCP MONITOR SERVER PARAMETERS
                   Codec Set: 4                     RTCP Reporting Enabled? y
          UDP Port Range Min: 2048_           Use Default Server Parameters? n
          UDP Port Range Max: 3028            Server IP Address: 999.999.999.999
          Direct IP-IP Audio? y                                Server Port: 5005
        IP Audio Hairpinning? y                     RTCP Report Period(secs): 5

DIFFSERV/TOS PARAMETERS                  AUDIO RESOURCE RESERVATION PARAMETERS
      Call Control PHB Value: 34_                                RSVP Enabled? y
             Audio PHB Value: 46                     RSVP Refresh Rate(secs): 15
                                              Retry upon RSVP Failure Enabled? y
          802.1P/Q PARAMETERS                    RSVP Profile: guaranteed-service
            802.1p/Q Enabled? y               RSVP unreserved (BBE) PHB Value: 43
Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 6
                 802.1Q VLAN: 0
```

Screen 8. IP Network Region screen

Region

Displays the number of the region being administered.

Valid entries	Usage
1-250	Numeric identifier for the region.

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.

Min UDP Port Range

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

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

Max UDP Port Range

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

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

Call Control PHB Value

Provides scalable Internet service discrimination without per-flow state and signaling at every hop. Call Control Per-Hop Behavior (PHB) and Audio PHB values have decimal equivalents. Use the IP TOS field to support the DiffServ CodePoint (DSCP).

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control PHB 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 VoIP Media codepoint.

Enter the decimal equivalent of the VoIP Media PHB value

Valid entries	Usage
0-63	Enter the decimal equivalent of the VoIP Media PHB value.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter y to use the voice channel directly between IP endpoints for audio transmissions, entirely bypassing Avaya servers,

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 Avaya DEFINITY Server's TDM bus.

802.1p/Q Enabled

When the `802.1p/Q` field is enabled (**y**):

- The `802.1Q VLAN ID` field defaults to zero (0).
- The DSCP value is taken from the Network Region form.
- The Ethernet frame format changes. If corresponding changes are not made to the attached Ethernet switch, service to the C-LAN is interrupted. Ethernet frames will only be tagged with VLAN and user-priority values if the `802.1p/Q Enabled` field is set to **y**.

For all H.323 call signaling

- Ethernet frames are marked with the user priority configured in the `Call Control 802.1p Priority` field.
- IP packets are marked with the VLAN tag configured in the `802.1Q VLAN` field on the Network Region form.

NOTE:
802.1Q is not appropriate for a hub-based network.

Valid entries	Usage
y/n	Enter y for 802.1p MAC-layer prioritization and 802.1Q Virtual LAN specification for this region.

Call Control 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**. Ethernet frames will only be tagged with a user-priority tag if one is configured in the `Call Control 802.1p Priority` field.

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

Audio 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**.

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

802.1Q VLAN

Appears only if the 802.1p/Q Enabled field is **y**.

Valid entries	Usage
0-4095	Specifies the 802.1Q virtual LAN value.

Overview of Location by Region changed screens

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

IP Network Region screen

Field descriptions for page 1

```
change ip-network-region 1                                 Page   1 of   2
                               IP Network Region

      Region: 1
        Name:

Audio Parameters                 Direct IP-IP Audio Connections? n
   Codec Set: 1                            IP Audio Hairpinning? y
    Location: 3
UDP Port Range                                     RTCP Enabled? y
         Min: 2048               RTCP Monitor Server Parameters
         Max: 65535               Use Default Server Parameters? n
                                              Server IP Address: ___.___.___.___
DiffServ/TOS Parameters                             Server Port: _____
 Call Control PHB Value:               RTCP Report Period(secs): __
   VoIP Media PHB Value:
          BBE PHB Value:          Resource Reservation Parameters
                                                      RSVP Enabled? y
             802.1p/Q Enabled? n            RSVP Refresh Rate(secs):
 Call Control 802.1p Priority:      Retry upon RSVP Failure Enabled? y
   VoIP Media 802.1p Priority:                  RSVP Profile: guaranteed-service
                  802.1Q VLAN:
```

Screen 9. IP Network Region screen

Location

Provides the ability to have correct time and date information on a registered IP endpoint worldwide.

Valid entries	Usage
1-64 or blank	For Avaya S8300 Media Server with an Avaya G700 Media Gateway, Avaya S8700 Media Server for Multi-Connect configuration, and Avaya S8700 Media Server for IP Connect configuration only. The range of valid entries for other Avaya servers will differ.) 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.

Overview of 4620 phone changed Station screens

There is a new entry in the Type field on the Station screen. Page 2 of the Station screen 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 screen.

NOTE:
This is just the portion of the Telephone table on the Station screen that has changed.

Table 1. telephones

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

Overview of 4602 phone changed Station screens

There is a new entry (4602) in the Type field on the Station screen.

```
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
```

Notice the Restrict Last Appearance field defaults to **n**.

```
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
```

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 screen.

NOTE:
You cannot change an analog phone administered with hardware to a virtual extension if TTI is y on the Feature-Related System Parameters screen. Contact your Avaya representative for more information.

NOTE:
This is just the portion of the Telephone table on the Station screen that has changed.

Table 2. telephones

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

Overview of 2420 phone changed Station screens

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

- Feature Access Code screen
- Feature-Related System Parameters screen
- Station screen
- Terminal Parameters screen
- list usage node-name
- list usage ip-address

Feature Access Code screen

There is a new Station Firmware Download Access Code field on the Feature Access Code screen.

```
  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:
```

Screen 10. Feature Access Code screen

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.	Enter the code you want to use for station firmware downloads.

Feature-Related System Parameters screen

There are two new fields on this screen:

- 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	Enter **y** to allow on-hook dialing for 607, 2420, 4600-series, 6400-series, and 8400-series phones.

Station screen

There is a new entry (2420) for the Type field on the Station screen.

```
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
```

Screen 11. Station screen

Page 2 of the Station screen is unchanged in Avaya MultiVantage software. Examples of the third, fourth and fifth pages (if applicable) follow.

```
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** 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 screen.

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 screen. Contact your Avaya representative for more information.

NOTE:
Reviewers: This is just the portion of the "Telephones" table in the Station screen description that has changed.

Table 3. telephones

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

Terminal Parameters screen

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

List 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 screen.

```
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 screen.

```
list usage node-name tftpserv

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

List usage ip-address

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

Administering and Enabling IPSI(s)

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 Avaya MultiVantage™ software is first initialized it ***does not know*** *the primary and secondary control subnet addresses. Submitting the following screen will cause the Avaya MultiVantage™ software 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 and Secondary (if equipped) Subnet Addresses 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 Avaya MultiVantage call processing software does not contain the subnet information displayed. After verifying the displayed information, submitting this form will cause the Avaya MultiVantage software to be updated with the subnet information displayed.

CAUTION:

*If the information displayed in the Primary and Secondary 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. 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 MultiVantage™ software

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 IPserver interface circuit packs’’ on page 74

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:  8B01
      Host: ipsi-A08b
   DHCP ID: ipsi-A08b
```

2. Verify that the fields associated with the Primary and Secondary (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.

Repeat the **add ipserver-interface <Port Network>** for each IPSI controlled Port Network.

Test Connectivity to IPserver interface 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 on **Execute Pingall.** Select **Other Server(s), All IPSIs, Ethernet switches** and click on **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 on **View IPSI Version**. Select **Query All** and click on the button **View IPSI Version**.
   - Verify the firmware release for each TN2312AP IPSI. If upgrade is required , refer to the upgrade procedures in documentation.

Enable control of IPserver interfaces

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.

New and Changed Commands

4

New commands

This following section describes the commands that are new for the Avaya MultiVantage™ software.

New commands for conference enhancements

List VDNs

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 Conference VDN

The status of a Meet-me Conference VDN can be displayed as shown in the following example. In this example, there are 3 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:
```

Resetting VDNs

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***

New commands for DPE

Dial Plan Analysis Table

Action	Object	Qualifier
change	dialplan analysis	—
display	dialplan analysis	[‘print’ or ‘schedule’]

Dial Plan Parameters

Action	Object	Qualifier
change	dialplan parameters	—
display	dialplan parameters	[‘print’ or ‘schedule’]

Uniform Dial Plan

Action	Object	Qualifier
change	uniform-dialplan	Enter 1-7 digits between 0-9
display	uniform-dialplan	Enter 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’]

New commands for CLAN QoS and CIDR

The output of two commands now contain new columns of Subnet Mask data in Avaya MultiVantage software. The syntax for the two commands are unchanged:

a. **netstat ip-route** (lists IP routes from all circuits)

b. **netstat ip-route board** <board location> (lists routes from one 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
```

Screen 12. netstat ip-route command screen

New commands for 4620 phones

Language Translations (display-messages button-labels)

action	object	qualifier
change	display-message view-buttons	
display	display-message view-buttons	

New commands for 2420 phones

Language Translations (Display-Messages Button-Labels)

action	object	qualifier
change	display-message view-buttons	
display	display-message view-buttons	

TFTP-Server

action	object	qualifier
change	tftp-server	
display	tftp-server	

Please see also Changed commands for 2420 phones.

Changed commands

This following section describes the commands that have new qualifiers or new output for the Avaya MultiVantage software release.

Changed commands for conference enhancements

Display capacity command

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 command

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 command

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
```

Changed commands for 2420 phones

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.

Please also see New commands for 2420 phones.