Avaya MultiVantage™ Software on

Avaya™ S8700 Media Server
Avaya™ S8300 Media Server
Avaya DEFINITY® Server R
Avaya DEFINITY® Server SI and
Avaya DEFINITY® Server CSI

for
Avaya MultiVantage Software (Release 1.1.2),
Issue 1.0 (01.2.065.3)
Change Description

Release 1.1.2
555-233-423
Comcode 700250699
Issue 1
August 2002

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.

If you need technical support or assistance and are outside of the United States, contact the equipment vendor from whom you purchased your equipment service maintenance contract. If you need to report toll fraud issues regarding a public telephone, contact the in-country telephone service provider.

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.

To order copies of this and other documents:

Call: Avaya Publications Center
Voice 1.800.457.1235 or 1.207.866.6701
FAX 1.800.457.1764 or 1.207.626.7269
Write: Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA
Attention: Avaya Account Management
E-mail: totalware@gwsmail.com

Highlights

This section presents highlights of features and enhancements to Avaya MultiVantage™ software.

General enhancements

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

EC500 enhancements

The enhancements to the EC500 system for cellular or wireless phones simplify administration and reduce costs. The enhancements include:

- Calls to an XMOBILE station can be extended directly over an ISDN trunk connected to the public network. Regular ARS (automatic route selection) or AAR (automatic alternate routing) tables are used to select the trunk for an EC500 call, which simplifies administration.
- Administration defines whether to keep Call Detail Records (CDR) for EC500 calls.
- Call Filtering helps manage cellular phone costs by limiting the calls extended to the cellular network for EC500 users. Call delivery can be limited, on a per-call basis, to only external calls (from a customer), only internal calls, all calls, or no calls.

Avaya Installation Wizard

The Avaya™ Installation Wizard is a tool for use in new installations (not upgrades) to help reduce complexity, time-to-install, and the cost of installation for the Enterprise Class IP Solutions (ECLIPS) portfolio, currently beginning with the S8300 media server with up to five G700 media gateways in a stack.

The Avaya Installation Wizard delivers the following installation advantages:

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

The Avaya Installation Wizard can guide installers through:

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

For additional information, including a hands-on prototype of the Avaya Installation Wizard, see:

http://support.avaya.com/avayaiw

Change Descriptions

The following problems have been addressed and corrected in Release 1.1.2 of Avaya MultiVantage™ Software (01.0.065.3).

1. Valid entries in the equipment type fields on the 'display software' form give error messages.
2. Some admin help messages related to duplication on the DEFINITY® server SI were invalid if entered on a duplicated system.
3. ISDN trunk originated tandemed DCS calls to analog caller ID stations showed trunk information rather than calling party information.
4. If the BCMS (Basic Call Management System) Abandoned Call Timer was non-zero, and a VDN queued calls to a split/skill, the VDN ACD (Automatic Call Distribution) call count did not match the split/skill call count.
5. One could filter only events below 3399. Now, filter can be applied for events till 9999.
6. After a reset 2, IP phones could not get registered.
7. QSIG Call Completion could not be cancelled in an unknown numbering plan.
8. MultiVantage Software did not handle non-native manufacturer-specific information embedded in the manufacturer-specific extension of the Diversion Leg information. This caused the display to be incorrect.
9. Reboots were hanging occasionally on the S8300/G700 configuration.
10. QSIG diversion calls might use the wrong user permissions to make calls.

11. Calling party category was incorrect in CDR (Call Detail Record) for Russian Shuttle and Rotary trunks.
12. Calls doing CCRON (Coverage of Calls Redirected Off Net) over ISDN trunks and then resuming coverage to local AUDIX® might not find the right mailbox.
13. User could enter a non-IP extension for "list trace ras ip-station ’extension‛".
14. BCMS Service Level could report 100 percent even if there were abandoned calls.
15. ICSU (Integrated Channel Service Unit) tests attempted to run on a MM710. They now abort properly with code 1412.
16. In rare situations, IP agents could not register due to memory corruption, and some agents could not log in or out.
17. The call forwarding destination number could only be up to 16 digits. Now it can be up to 18 digits.
18. Reset ip-station is a new command, end-user visible and call destructive: all calls are taken down when this command is run. The intent is to unregister and re-register all phones so that they can be upgraded.
19. Neither the 'packet-length' option nor the “board’ option is now allowed to go with 'source' option in the ping ip-address and ping node-name commands.
20. Erroneous tones were heard on a call which resumed from a DCS (Distributed Communications System) coverage point on another node back to the original receiver.
21. SPE (Switch Processing Element) indicator on a SAT (System Access Terminal) using IP to connect to the media server via a TN799 was not updated after an interchange for duplicated SI servers.
22. The check for duplicate node-names across the forms "change node-names ip" and "change node-names audix-msa" forms was case sensitive, but other areas of the system were not, which made node-names differentiated only by case effectively identical. Now the check for duplicate node-names across the forms "change node-names ip" and "change node-names audix-msa" is case insensitive.
23. After an upgrade from a pre-R8 load with a manually-administered 'dadmin' login, another 'dadmin' service level login could be added as long as it used a login string other than 'dadmin'. This resulted in translation corruption.

24. Under certain circumstances, the system sent Notify messages containing hold and unhold notification over trunks administered for countries that do not want to see those Notify messages. This could lead to the call being disconnected.
25. With a media gateway analog phone, after recall dialtone was applied the station could not place a call.
26. It was not possible to transfer from one trunk enabled for Busy Tone Disconnect to another.
27. The text "a=info: <collected digits>" did not appear on two-line DCP (Digital Communications Protocol) phones displays, when "auto Answer" was set to "all" and "Select Last Used Appearance" was set to "y" on the "station" form.
28. The Destination field on the administered-connection form took only a TAC (Trunk Access Code) number. It now takes an extension, TAC+extension, or FAC+extension.
29. An IP Softphone/IPAgent in Telecommuter configuration on a call lost the call when the telephone attempted re-registration.
    Now, if the telephone upgraded to the current firmware, this does not happen.
30. The SAT command "list usage holiday-table X", where X is a holiday table number, was not available.
31. An attendant could not transfer intercepted calls using a CallMaster®.
32. Some pages of the SAT form for changing IP interfaces had fewer than 15 lines on them. Now that is true only for the last page of the form.
33. An IP Softphone/IPAgent in Telecommuter configuration on a call lost the call when the TCP call signaling socket closed.
34. An attempt to use the add eda-external-device-alrm command for a G700 port resulted only in an error message.
35. The ’list history’ log displayed random data for the port for ’ip-a’. Now it shows the IP port assigned to the station.
36. Network outages caused software and TN799 to go out of synchronization, causing extended service outage for remote sites due to briefer network outage. Now the software and TN799 re-synchronize promptly.
37. User could not add a host route on the ip-interface form.
38. When using an ATM-WSP (WAN Spare Processor) in critical reliability or ATM (Asynchronous Transfer Mode) Network Duplication - that is, duplicated Port Network Connectivity - a WSP might not become active in certain situations when it should.

39. The license file sometimes set the "Maximum number of VAL Boards:" field incorrectly.
40. If a cell phone number was mapped to several different Xmobile stations and none of its mappings had mapping mode "both" or "origination", then the cell phone number was not retrieved when the "list xmobile mapping full-string" command was executed.
41. The system sometimes sent a STATUS ENQUIRY message so quickly after sending the SETUP message that the far end responded with a STATUS message giving a call state of NULL, causing the system to tear down the call.
42. UPS (Uninterruptable Power Supply) inline error handling did not check that the SNMP (Simple Network Management Protocol) trap had a valid IP address before taking any action (such as shutting down the server due to UPS power failure).
43. The command "status station xxx" did not show the VoIP (Voice over IP) address of a endpoint connected to a VoIP call.
44. Entering 'busyout ipserver-interface' or 'release ipserver-interface' did not busy out or release tone components on the IPSI (IP Server Interface) board. Now it does so for the **standby** IPSI board. However, it does not affect tone components on the active IPSI board.
45. The list config for a VAL (Voice Announcement over the LAN) board contained two identical field IDs, which prevented determining that the board was programmable.
46. Button labels on the 2420 DCP terminal were not administrable. Now they are, via the 'display/change display-messages button-labels' form.
47. EECCR ('Error Encountered Cannot Complete Request') might occur when using the "ping ip-address xxxx" command.
48. The following commands were inappropriately available when Port Network Support in an S8300/G700 configuration was set to "n": status ip-network-region, display failed-ip-network-region, test failed-ip-network-region.
49. An ASAI (Adjunct Switch Application Interface) request for Third Party Merge set up using ARS (Automatic Route Selection) digit conversion failed when going to local station that was busy on another call.
50. When displaying on the SAT a cabinet that contained a CSS (Center Stage Switch), the Duplicate fields on the form contained no data.

51. On the ip-network-region form, the upper limits of certain fields were:

Call Control PHB [Per Hop Behavior] Value	99
VoIP Media PHB Value	99
BBE [Better than Best Effort] PHB Value	999
RTCP [Real Time Control Protocol] Report Period	99

And on the system-parameters ip-options form, the upper limit for the Default RTCP Report Period field was 99.

Now the upper limits for these fields are:

Call Control PHB Value	63
VoIP Media PHB Value	63
BBE PHB Value	63
RTCP Report Period (secs)	30
Default RTCP Report Period (secs)	30

Upon upgrade, if values are larger than the new upper limit, the values are set to the new maximum.

52. Calls using telecommuting access and similar features did not handle incoming FACILITY messages when routed over DCS.

53. After interchange on an S8700 media server, standby might call in obsolete ACP (Avaya Call Processing) alarms.

54. Security violations or disabling a Multi Vantage login did not disable access to the Linux layer if the login had shell access.

55. When a TN2501 VAL board was inserted, the recorded announcements might not be reflected in the output generated by the SAT commands list integrated-annc-boards and display integrated-annc-boards.

56. The following commands were available on platforms other than the S8700 and S8300 media servers: add/change/display/list/remove media-gateway, list configuration media-gateway, status mg-announcements. They are no longer available except on those media servers.

57. The Whisper Page feature did not work on the S8300/G700 configuration.

58. AAR (Automatic Alternate Routing), ARS, and Toll Analysis entries with a trailing wildcard (for example, 817xxxx) were displayed with a final # instead (817xxx#). This was a display-only problem; it did not affect the way calls were routed.

59. Collected digits in the UUI (User to User Information) were not sent in an ISDN SETUP message when routed over an ISDN trunk via adjunct application "route-select". Now collected digits in the UUI are sent along with the IN-VDN time, VDN name, and other LAI (Look-Ahead Interflow) info, if an adjunct performs "route-select" to an ISDN trunk. This routed call behaves as a route to digit with coverage **with** UUI.

60. When a S8700 was booted and the ACP logins were synchronized with the Servers /etc/passwd file, logins that didn't have shell access were given shell access by default.

61. It was not possible to trace port 32 on a media module.
62. On the system-parameters mode-code form, if a value assigned to a non-displayed field (#04, #05, #07, #08) was entered for any of the mode code fields, the cursor would go to a non-displayed field with the error "Duplicate entry".
63. The CDR with reliable session did not come up for S8700 or S8300 configurations.
64. A co-resident DLG (DEFINITY LAN Gateway) ASAI link had to be hyperactive for 10 consecutive 1-second intervals before a hardware error (513) was logged. Now a hardware error is logged for each 1-second interval that the co-resident DLG ASAI link is hyperactive.
65. Sometimes when one added or removed IPSIs, the SAT displayed EECCR.
66. When translations were saved on the primary controller and the filesync to the LSP (Local Survivable Processor) failed, the error message shown was the same as that shown for a failure to filesync the translations to the duplicated server. Now, if the translations make it to the duplicated server but not to an LSP, the message shown informs the user that the transfer to the LSP is the portion that failed. If the duplicated server does not receive the translations, the message informs the user that the failure was to the duplicated server, just as it did previously. If both fail, the message refers to the failure in the transfer to the LSP.
67. The command 'list measurements occupancy pktint' was allowed for the S8700 and S8300 configurations, although it was not valid. Now the 'list measurements occupancy' command does not allow 'pktint' as an identifier for those configurations.
68. If an agent's idle time became greater than 9 hours, the adjusted idle-time calculated by the BSR (Best Services Routing) polling switch was set to -200.
69. If a CMS (Call Management System) measured call was conferenced with a CMS measured agent and then subsequently transferred to a VDN, and the caller dropped while listening to the announcement associated with a disconnect vector step, CMS did not track the call.
70. On S8700 IP Connect configurations, when an IP phone was active on call app 1 and a second call was received on call app 2, audio could be lost on the current call.
71. When an attendant agent put an ACD call on hold, it did not realert the attendant agent after the Timed Reminder on Hold Interval expired. This could lock up a call appearance for some other agent.
72. When a registered IP telephone rebooted, an URQ (UnRegistration reQuest) denial event was logged incorrectly.

73. Users had to administer the second digit table as a separate entity via add, change, display, and remove commands. Now, as part of the Dial Plan Expansion, the second digit table is no longer necessary. Therefore the second-digit keyword has been eliminated for the commands mentioned above.
74. When a call not measured by CMS was redirected (coverage, forwarding) to a CMS-measured endpoint, no setup event was reported to CMS.
75. For S8700 and S8300 media servers, abbreviated dial enhanced lists defined in list 2 were not preserved on a save trans/reset sys.
76. Vector controlled trunk-to-trunk transfers failed if the trunk was using Busy Tone Disconnect.
77. The "monitor bcms" command truncated agent extensions longer than 5 digits. Now 6- and 7-digit agent extensions are listed in their entirety.
78. Status health did not have the same key maps as other status forms.
79. When a license was added, if either the "IP Attendant Consoles?" field was turned OFF and the associated field "Maximum Concurrently Registered IP eCons:" was greater than zero, or the "IP Stations?" field was turned OFF and the associated field "Maximum Concurrently Registered IP Stations:" was greater than the default value of 5, a license exceedance error should have been raised, but was not.
80. If Per-Call CPN/BN (Calling Party Number/Billing Number) was administered in the Incoming Call Handling table, and an EC500 called a destination matching the entry, then the call never rang the destination. EC500-originated calls now ignore the Per-Call CPN/BN entry.
81. In the S8700 and S8300 configurations, the pbx_id, tac, and trunk member number were neither displayed nor announced correctly.
82. After massive translation changes, BCMS agents split data was incorrect and caused warm starts.
83. The reserved slot administration did not take into account that slot 6 in a rack-mount-carrier is a double wide slot.
84. When a BSR interflow was attempted with the Network Call Redirection / Network Call Deflection (NCR/NCD) or Network Call Redirection / Network Call Transfer (NCR/NCT) feature, and the interflow failed due to (for NCT) the second leg of the call not staying up or (for NCT or NCD) some "reject" or "error" FACILITY return message from the PSTN after NCR invocation, the call was not queued to the BSR "local best" skill (if there was one) as it should be for failed BSR interflow attempts.
85. Interworking between DCS and QSIG Leave Word Calling failed.

86. If one attempted TAC dialing from a DCP station on a port network over a T1 trunk to a DEFINITY Server R, then TAC dialed back to an S8700/G700 configuration, and then dialed the extension of any station, the call routed to the wrong station or the attendant console.
87. The COR (Class Of Restriction) fields for Authorization Codes were reset to 0 during translation migration from DEFINITY Server R to S8700.
88. The 'change media-gateway N' form did not have slots 8 and 9 available for input.
89. The ACTR (Automatic Customer Telephone Rearrangement) feature did not work properly when TTI (Terminal Translation Initialization) or administration terminal merged a station with a port and the Automatic Moves field was set to 'once'.
90. At times it was not possible to make trunk gateway calls using Remote Office, due to the signaling group(s) being out-of-service.
91. When two parties were in conference, and a third party used a bridge appearance to join the conference, if one of the original (non-bridged) parties dropped the call, the whole conference call was dropped.
92. When one administered IPSIs using DNS (Domain Name System) names, the addition to the /etc/hosts file failed.
93. There were two DTMF tones heard on media gateway DCP when pressing a dial pad button on an IP phone.
94. The value of "Abbreviated Dial Programming by Assigned Lists?" in the system-parameters features form was reset back to n(o) after system reset 1 or 2, or periodic maintenance.
95. Administering data-modules or station dtdms with "X" ports could result in EECCR for various list commands.
96. An error 217 was seen when a media gateway booted.
97. If a UDP (Uniform Dial Plan) extension was entered in a remote voice mail hunt group, and that extension did not have a node number associated with it, an unhelpful error message was displayed. Now a helpful error message points the administrator to the root of the problem.
98. A remote service observer on a tie trunk did not hear VOA (VDN of Origin Announcements).
99. When the low value from the license for IP_Agent, IP_Phone, IP_Romax, IP_Soft, or IP_eCons was greater than the high value from the license, the associated translation field on the customer options form should have been automatically set to zero and the field restricted to DISPLAY ONLY.
100. On the G700, no tone_on_hold was heard on shuffled IP phone → IP trunk → IP station held calls.

101. When downloading translations from S8700 to the LSP occurred, major alarms were generated in regard to missing EPNs.

102. It was not possible to administer a loop-start CO (Central Office) trunk or PCOL (Personal Central Office Line) on the MM711 media module with country code 18 (China).

103. The Network Call Redirection (NCR) feature did not support the Telcordia or ANSI Explicit Call Transfer (1998) PSTN feature as required by many Avaya customers for use with the Nortel DMS-100 or Lucent 5ESS PSTN switches. (This feature is similar to the R8.3/R9.1 NCR "Network Call Transfer" feature currently used with the DMS-250 and DEX600 PSTN [Public Switched Telephone Network] switches.)

104. When migrating from DEFINITY Server R to S8700, the aut-msg-wt button was corrupted.

105. A call to a 7315D or 7317D set could cause message buffer exhaustion if the display message was corrupted. Now the message buffer exhaustion problem does not occur, even if the display message is corrupted. The user just gets a blank display.

106. When a "reset sys 4" was issued from the SAT screen on an S8700/G700 configuration, the media gateway could not register.

107. The conference "toggle-swap" button could not be removed by self-administration.

108. If Music On Hold was administered on a media gateway, it did not work on a call that originated from a port network, only on calls that originated from that (or another) media gateway.

109. Calls between IP phones did not get music on hold.

110. Duplicated CDRs were generated when Reliable Session Protocol was in use.

111. During migration from DEFINITY server R to S8700, the "BCMS/VuStats Abandon Call Timer (seconds)" field on the "system-parameters features" form was set to an invalid value.

112. IP Phones could not get registered after a warm restart.

113. Making calls from an IP phone to an ISDN trunk using the trunk access code, the user could not break second dial tone. The call would not complete.

114. When there were more than 6 bridged call appearances and a call came in from a DCP phone on a G700, the controller crashed.

115. The MAC (Media Access Control) address displayed on the media gateway form was backwards.

116. The 'list data-module' and 'list station' commands could cause the system to hang due to real time issues.

117. Music "on hold" continued after the attendant answered the call when the held call timer expired.

118. Several fields on the second page of the trunk group form were ignored for calls going to Xmobile stations. Instead the calls used these values: Send Name (y), Send Calling Number (y), Outgoing Channel ID Encoding (preferred). They now use the administered values, except that, if the Xmobile station has the Display Module field set to 'y', the Send Name field is still ignored: the station incoming display is sent. However, if the Display Module field is set to 'n', then the Send Name field value determines whether or not a standard trunk display is sent.

119. If the Vector Disconnect Timer on page 9 of the system-parameters feature form was not blank, Meet-me VDN conference calls automatically terminated when the timer expired.

120. On occasion, list measurements collected little or no data.

121. In the MST (Message Sequence Tracer) trace, pairs of H.248 messages could appear in reverse order.

122. The signal might outpulse a # (pound sign) that was appended by QSIG rerouting.

123. The packet interface went out of service on reset system 2.

124. The Group Page feature worked for only 5 members of the page group.

125. The Protect flag on TN2501 VAL announcements did not work consistently.

126. If internal auto answer was activated on an IP(46xx) station, and the user logged off and then logged back in, then the associated lamp would relight, although the feature was actually off.

127. In an S8700 IP Connect system, using MF (Multi-Frequency) trunks and TAC, dialing from a G600 not containing the MF trunk resulted in no talkpath.

128. After reboot of an S8300/G700 configuration, the green "OK to shut down" LED flashed indefinitely and the shutdown button did not work.

129. Anti-thrashing curbs kicked in for up to 24 hours (rather than the standard 1 hour). In addition, the "server" command did not always display the message "(curbs in)" when, in fact, curbing was in force.

130. If there were IP boards in the same network region as a G700, and they were all disabled (which might happen due to a maintenance test), calls from the G700 to a port network in another network region rang, but failed to connect.

131. Pre-interchange filesyncs accompanied software-requested restarts that escalated to interchanges, which resulted in unnecessarily extended outages. The window of no new call processing has been shortened by several seconds, and the risk of outage has been greatly reduced.

132. If an IP phone had a bridge appearance set up with the extension of a DCP phone, and answered a call from an IP phone, a second call to the original IP phone while the IP-to-IP call was connected did not provide ringback to the caller.

133. The call forwarding destination number could be up to 16 digits. It can now be up to 18 digits.

134. If IP trunks were interconnected using G.729 but using G.711 within their own regions, a call attempted on IP phones between an S8700/G700 configuration and an S8300/G700 configuration did not get talkpath. Or, in one of these configurations, when an IP phone was in one region and the media gateway was in another region, dialtone did not occur.

135. If the upgrade was not "made permanent", no alarm was raised. Now, if it is not "made permanent" within 2 hours, an alarm is raised.

136. If someone called an IP phone that was already on a call, the IP phone rang once, but the caller did not hear ringback.

137. On an S8700/G700, 'save trans' did not work.

138. When DEFINITY server BSR polled using TSC (Temporary Signaling Connection) to an S8700 configuration or vice versa, the agent idle time was corrupted due to byte ordering compatibility issues. Also, IN-VDN and UCID (Universal Call ID) data exhibited similar byte ordering corruption when transmitting between DEFINITY servers and S8700 configurations.

139. If a softphone user registered as a telecommuter and provided an invalid callback number, system attempts to use the invalid number could overwhelm the system to the point of paralysis. Now the invalid number is attempted only once.

140. It was possible to administer a 6- or 7-digit number in the Media Complex Extension field of the Station form. This was invalid because R1 and R2 IP Softphones do not support extensions longer than 5 digits in the server/endpoint protocol. Now a Media Complex Extension cannot exceed 5 digits.

141. The list-trace-station button activation command did not work for IP and R300 stations.

142. Port numbers, socket IDs, IP addresses, and other IP information gathered by the MST were scrambled, that is, not in the correct byte order, in S8700 and S8300 configurations.

143. In a S8300/G700 configuration, the list measurements ip codec/dsp-resource/signaling-groups commands were inappropriately available if Port Network Support was set to "n".

144. Garbage characters could get into the qualifier of the "list history" log whenever an 'ip-a' entry was logged.

145. If there was a ringback step in the Meet-me vector, when the first user joined the Meet-me conference, that user continued to hear ringback until another party joined the call (or some other event, such as an announcement, interrupted the ringback).

146. The list-trace-station button activation command was not logged in the command information history log.

147. CMS connectivity initialization might fail due to lost messages.

148. The system might reset itself if an unacceptably large message came in over a TCP link. That message is now discarded without a system reset.

149. When a user dialed a Meet-me conference using abbreviated dialing, the access code remained on the display. Now the access code is blocked and intercept is given.

150. PASTE data was missing some Feature Access Codes (FACs), some button types, and some station types for the IP Agent softphone application.

151. When a large burst of IP call center traffic occurred, software errors were logged and the TN799 was reset, causing all adjunct links through the TN799 to be dropped briefly.

152. Auto-answered IP Agents with as-needed service link did not hear zip tone or VOA when ACD calls terminated on their sets.

153. After considerable administration of BCMS agents, it was possible to experience switch restarts when an inactive BCMS agent was removed by the software to make room for a new agent logging in.

154. On a 2420 telephone, if the user pressed the Messages button while on an active call, the phone dropped the call and dialed AUDIX. Such a button press is now ignored.
Also, if the user pressed the Messages button to log into AUDIX while off hook, and then picked up a second line, the second line dialed AUDIX as well. The second line now returns dial tone.

155. More than the maximum allowed NCA-TSCs (Non-Call Associated Temporary Signalling Connections) could be administered on the signalling group form.

156. The command 'dup station' failed when the station duplicated was a 2420 or 4602 telephone.

157. Calls between IP phones might drop if the called phone had one line active already and had a bridged appearance to any phone not on the same media gateway and network region as itself.

158. If "ATM-SW" (ATM Switch) was entered as either the Primary or Secondary Synchronization source, attempts to modify either field could result in an error.

159. A non-call-associated LWC (Leave Word Calling) call from QSIG to DCS failed, and did not give feedback to the user.

160. When removing the Primary Output Format from the "system-parameters cdr" form with either the "Use ISDN Layouts" or "Use Enhanced Formats" enabled, the error message "Required data not specified" was displayed. More useful messages are displayed now.

161. Busy tone was not heard when calling from a G700 to a busy station across an IP trunk.

162. Development-only fields were displayed on SAT for S8700 and S8300 configurations.

163. Security alarms levels that were success (WRN), failure (MIN), and lockout (MAJ) are now success (no alarm), failure (WRN), and lockout (MAJ).

164. If there were too many VDNs and stations administered, the server rebooted.

165. The user could not change the dial tone value on a media gateway.

166. Any account in the susers group can now modify the product ID.

167. In the S8300/G700 configuration, the misoperation alerting feature did not work.

168. Any IP address could log into the ACP SNMP login on the Linux platforms. Now only IP address 127.1 can log into it.

169. Remote login administration on S8700 configurations formatted the entry in /etc/passwd file incorrectly.

170. If a system with a X-Mobility PHS (Personal Handyphone System) adjunct also had stations with data modules (such as 8411D stations with analog data ports), internal data belonging to these stations could get corrupted. The administrator was unable to display or change the corrupted stations.

171. In an S8300/G700 configuration, rotary dialing timed out when the number being dialed was longer than about 4 digits.

172. If large numbers of stations and VDNs were administered, it could prevent the switch from booting.

173. Duplicates of a DECT Xmobile station whose Configuration Set field was blank had their Configuration Set field set to -1.
174. With two analog phones on a G700, one could not get talkpath back on a call with the other after using flash to put the call on “soft” hold.
175. Multiple-bit errors reported in an S8700/G700 configuration were ignored. Now they raise a major alarm and force an interchange.
176. Ringback was delayed when placing a call from one G700 to another on the same media server.
177. After migration from a DEFINITY server R to an S8700 configuration, the minor alarm threshold on the "set options" form was missing.
178. The S8700/G700 configuration rebooted due to low-level protocol mishandling of fragmented packets.
179. In duplicated IPSI port networks, IPSIs did not switch between test cycles
180. The S8700 and S8300 configurations used Avaya Site Administration (ASA) release 1_10-11. They now use ASA release 1.11-5.
181. The user could register the unsupported R1.0 dual connect IP telephone, but now can register only the supported single connect R1.5 IP telephone.
182. Vector-collected digits, when displayed on line two of the agent’s display at the "Info: " prompt, could cause the system to do an interchange.
183. Interchanges did not succeed if there were multiple LSPs registered to a media server.
184. TTR (Touch Tone Receiver)-level alarms that should be suppressed were present on the LSP.
185. Recording and repeatedly replaying on-board announcements caused the system to reboot. Also, an attempt to play back an on-board announcement multiple times played it back only once.
186. Outgoing MFC (Multi-Frequency Coded) calls using Mexico configuration on a G700 did not work.
187. The software failed when using IP hardphone as the voicepath for an IP softphone in the Telecommuter mode.
188. Links did not come into service if a VAL board had a lower link number than a TN799 board.
189. The first port network in a system, unless 1, was not displayed by the 'list config pn' command.
190. Messages from the login root: almenable cluttered the output of the command 'listhistory'.
191. A TN2501 VAL board did not come into service because it was confused with the media gateway announcement board.

192. CMS connectivity initialization failed with measurement of too many trunk groups (2000) and trunk members (8000).

193. Logging in as an "suser" (init/inads/craft) gave errors when logging in and errors when executing some commands, and it was necessary to specify full paths for commands.

194. The output from the 'status pnc' command did not show alarm counts for system links.

195. In S8300 and S8700 configurations, the 'remove license' command was visible to services users.

196. If the MCT (Malicious Call Trace) Delay Sending Release timer was non-zero, and a normal incoming call covered to a coverage point, then the call was released based on the "COR: Access to MCT" setting of the called party and not the COR (Class Of Restriction) of the coverage point.

197. Certain initialization conditions sometimes resulted in problems with CMS connectivity initialization and one-way talkpaths.

198. A one-IPSI system might show Port Network Connectivity (PNC) duplication constraints for Bearer Fault Detections (BFDs), falsely indicating partially functional PNC. Now BFD constraints are not applied to PNC State Of Health when no BFDs exist.

199. Rapid on-hook dialling on a G700 DCP telephone could result in dropping the second dialled digit.

200. Calls did not connect over CAMA (Centralized Automatic Message Accounting) trunks.

201. The command 'list sys-link' listed system links in type order. It now lists them in priority order.

202. When a port network containing a TN799 board was out of service, the TN799 reset itself as part of the coming into service, but IP calls on that port network and links using that TN799 failed unless there was an additional, manual reset of the TN799.

203. Data administered via the 'change paging loudspeaker' command on a DEFINITY server could not be successfully retrieved.

204. In S8700 configurations, users could lock themselves out of doing subsequent interchanges by killing the 'server –i' command before it was done.

205. The zip tone was heard at the port network DCP originator when making a call from port network DCP to media gateway DCP.

206. Port network and media gateway announcements could not be heard by G700 stations.

207. Systems without initialized time-of-day clocks could not set the time. The command 'set time' gave EECCR.

208. Could not make trunk calls between two media gateways.

209. Users dialing DCP phones off-hook did not hear DTMF if onhook dialing was enabled.