Avaya MultiVantage™ Solutions Hardware Guide

Issue 1
May 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.

How to Get Help

For additional support telephone numbers, go to the Avaya Web site:
http://www.avaya.com/
Click 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 Protcal (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	Greece	Italy	Luxemburg
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 1.410.568.3680
FAX 1.800.457.1764 or 1.410.891.0207
Write: Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA
Attention: Avaya Account Management
E-mail: totalware@gwsmail.com

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Avaya MultiVantage Solutions Hardware Guide

May 2002

Overview

Avaya Enterprise Class IP Solutions (ECLIPS) offer a range of products that enable businesses to tap the power of IP and converged networks and put them to work for your business. Powered by rock-solid, Avaya MultiVantage™ software, the ECLIPS portfolio of products includes Avaya™ Media Servers and Avaya™ Media Gateways, Avaya VisAbility™ Management Suite (network management tools) and Avaya Communication Devices.

With Avaya MultiVantage™ enterprises can rethink their existing communication networks, and determine where to introduce IP telephony in order to maximize the return on their investment. The choices are nearly unlimited, given Avaya's flexible, modular architecture and commitment to standards-based software and hardware.

Avaya's MultiVantage™ Software is highly scalable and reliable voice application software with rich call processing and contact center functions and widely accepted application-programming interface that supports a range of Avaya and third-party applications. Avaya is bringing continuous voice application innovation to the enterprise. Avaya MultiVantage delivers massive new scalability to provide a number of network devices to range from as few as 20 users to up to a million.

Avaya Media Servers and Avaya Media Gateways provide smart ways to rethink networking by adding top-tier scalability and reliability, while supporting mission-critical applications in a distributed, secure, multi-vendor environment. To provide businesses with maximum flexibility, the Avaya ECLIPS family of server and gateway components follow a modular mix-and-match approach. The wide range of custom configurations can be deployed to meet a spectrum of business needs: from a single-location, who desires an IP network for 200 employees, to a complex multi-national network capable of supporting 10,000-plus users.

The Avaya line of media servers provides a robust application platform based on industry standard operating systems to support distributed IP networking and centralized call processing across multi-protocol networks. These servers are available as an integrated solution with other servers or can operate independently.

Avaya Media Servers features and benefits:

- Redundant, survivable call and media processing supports crucial business continuity
- Standards-based computing supports Linux, Microsoft Windows and Avaya's DEFINITY™ operating systems
- Distributed survivable IP networking supports campus, global multi-site and branch environments

Avaya Media Gateways

These stackable and modular hardware elements deliver application-enabling data, voice, fax, video and messaging capabilities to your network. They support both bearer and signaling traffic routed between packet-switched and circuit-switched networks. These gateways are optimized to enterprise class telephony and provide a variety of flexible deployment options. Options include blended, such as IP and TDM, and 100% IP environments.

Avaya Media Gateways features and benefits:

- Interoperable with standards-based data networks
- Stackable, modular, configurable component solutions
- Redundant capabilities
- Distributed networking
- Supports traditional Avaya system cabinets

Avaya VisAbility™ Management Suite

Designed for a converged network environment Avaya Visability™ Management Suite provides a comprehensive set of standards-based software tools that make it easier for you to manage your complex network infrastructure-including both voice and data communications-through a common web-based user interface. This enables you to improve network uptime, increase staff productivity and reduce your operating costs.

Visibility features and benefits:

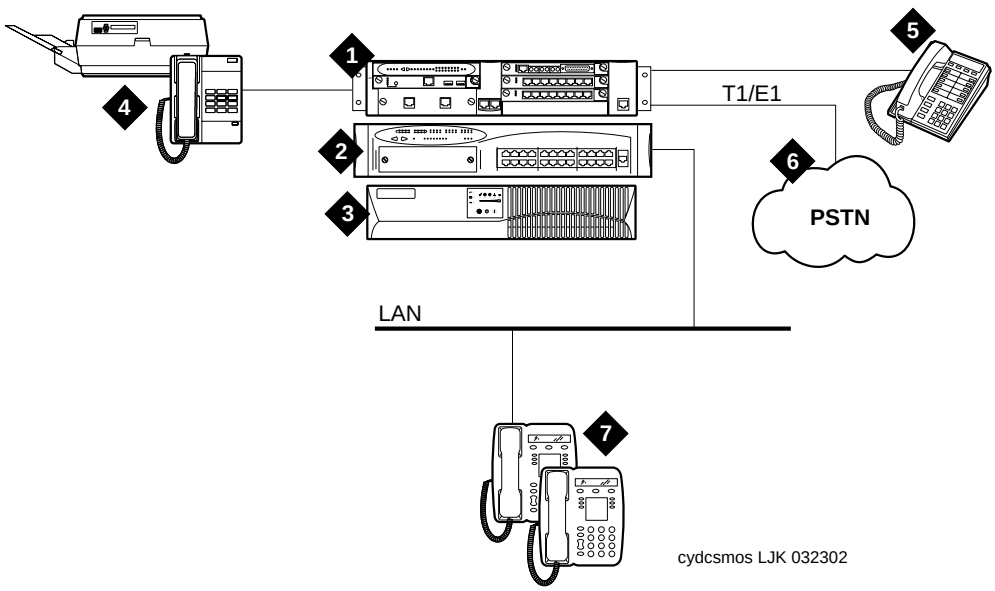
- Web-based system view of Avaya Media Servers, Avaya Media Gateways and IP Phones
- Voice over IP Monitoring provides industry-leading, centralized monitoring of network quality based on endpoint information
- Directory-enabled (LDAP) platform enables integration of management activities with business processes
- On-demand phone feature management provides users with the customization tools they need

Avaya Communications Devices

Innovative and standards based-providing new mobility opportunities Avaya offers a wide choice of flexible, intelligent, mobile and easy-to-use communication devices to meet your company’s unique needs. With analog, digital and IP telephones, the spectrum is covered. Highlights of the portfolio include the Avaya™ IP Softconsole, a software attendant console that brings high-end attendant feature/functionality to converged networks. the Avaya IP Softphone for Pocket PC brings the full functionality of you Avaya office phone to you PocketPC handheld device. The ECLIPS portfolio also includes the Avaya 4630 Screenphone, a full color, touch screen phone with Web-access. Avaya IP communication devices are supported without special power requirements.

Customer Configurations

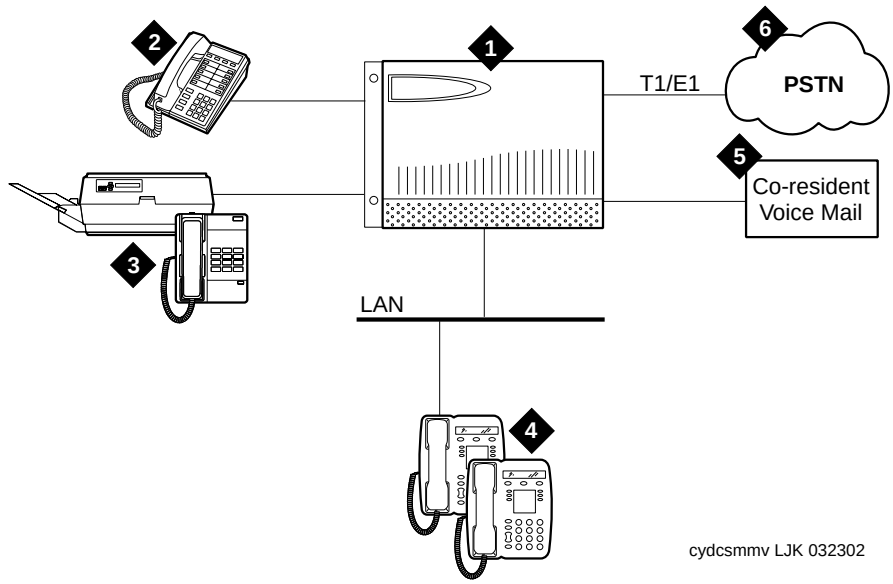
Small Customer Configuration using a S8300 Media Server with a G700 Media Gateway



Small Customer Configuration using a S8300 Media Server with a G700 Media Gateway

Item #	Description
1	S8300 Media Server with a G700 Media Gateway. This configuration is using the T1/E1, DCP and Analog Media Modules.
2	Ethernet Switch: Can be customer or Avaya provided. This device provides for port multiplication by having more than one network segment.
3	UPS: suggested to provide power during an power outage and allow for a graceful shutdown of the server.
4	Aanlog Capabilities: Can provide for analog trunks, stations and lines.
5	DCP phones: Avaya Multi-Function Digital phones.
6	T1/E1 Connectivity: • The T1 (or T-1) carrier is the most commonly used digital line in the United States, Canada and Japan. • The E1 (or E-1) is a European digital transmission format. It is the equivalent of the North American T-carrier system format.
7	Avaya IP telephones

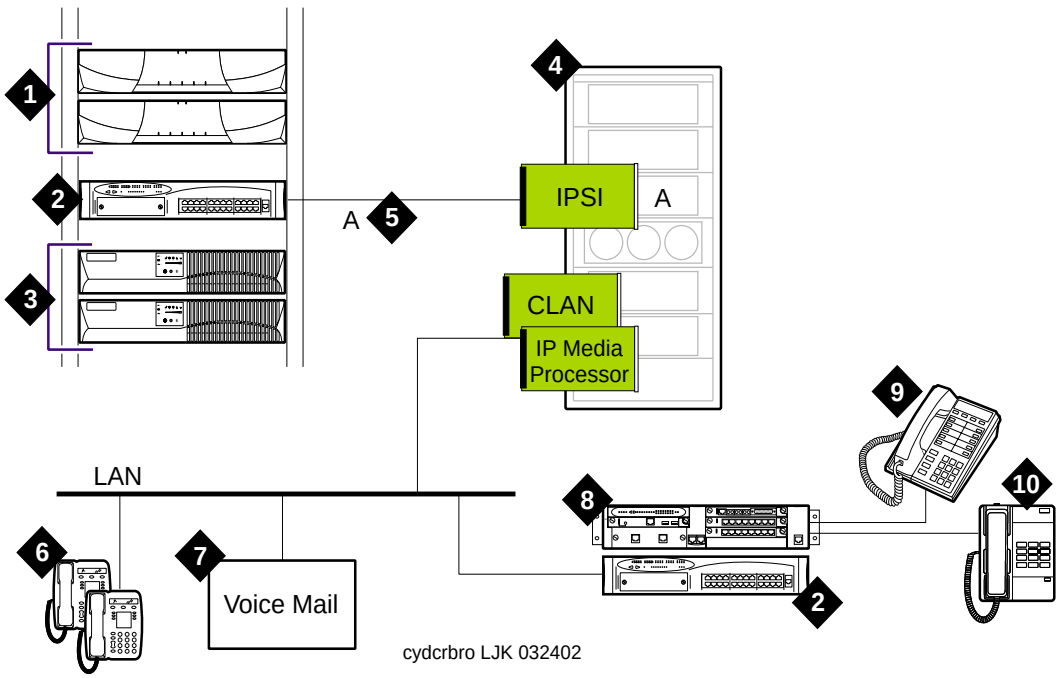
Small Customer Configuration Using S8100 with a G600 Media Gateway



Small Customer Configuration Using S8100 with a G600 Media Gateway

Item #	Description
1	The S8100 Media Server with a G600 Media Gateway: The S8100 Media Server is a two slot processor running the WINDOWS 2000 operating system with co-resident applications such as MultiVantage Software, Intuity AUDIX and Avaya Site Administration.
2	DCP Phones: Supports 2-wire and 4-wire Avaya Multi-function phones.
3	Analog Capabilities: Analog trunks, stations and FAX machines.
4	IP Phones: Avaya 4600 series phones.
5	Voice Mail: Co-resident Intuity 8 port voice mail or an external Voice Mail system.
6	**T1/E1 Capability:** • The T1 (or T-1)carrier is the most commonly used idgital line in the United States, Canada or Japan. • The E1 (or E-1) carrier isa a European digital transmission format. It is the equivalent of the North American T-carrier system format.

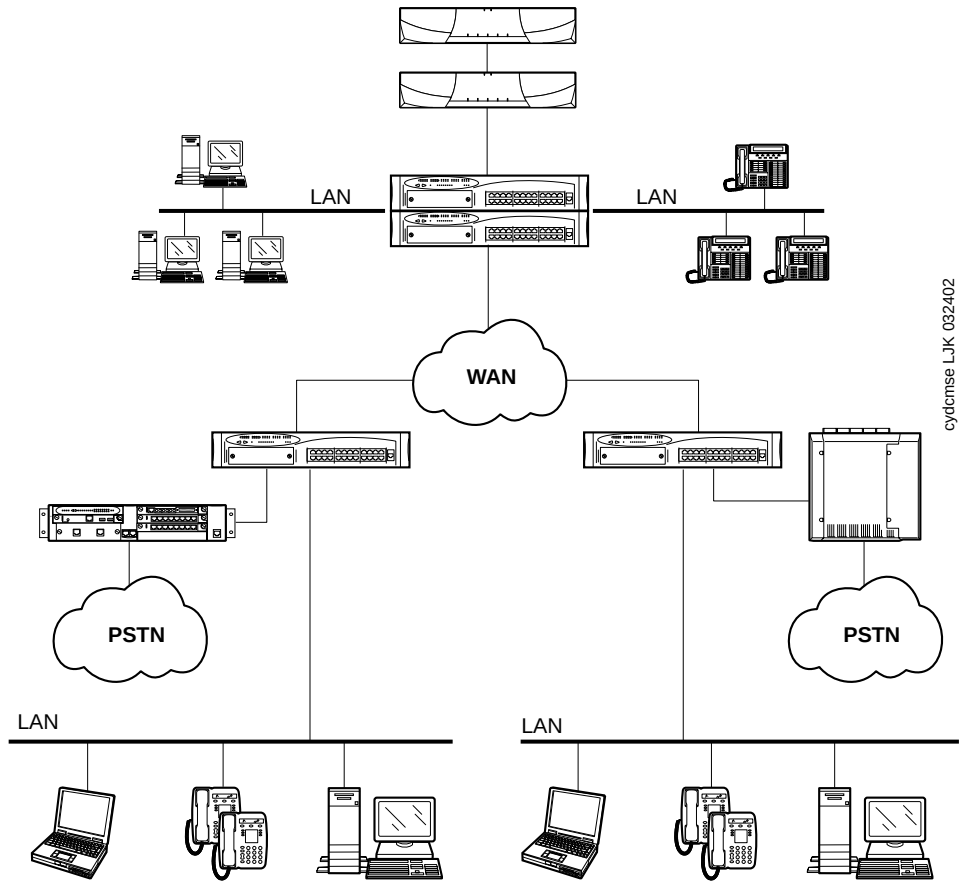
Branch Office Configuration using the S8700 Media Server in a Multi-Connect Configuration connected to a G700 Media Gateway



Branch Office Configuration using the S8700 Media Server in a Multi-Connect Configuration connected to a G700 Media Gateway

Item #	Description
1	S8700 Media Servers in a Multi-Connect Configuration
2	Etherenet Switch: Avaya provided.
3	UPS units: Two UPS units one for each server.
4	MCC1 Media Gateway
5	Dedicated LAN connectivity to the IPSI board in the MCC1 Media Gateway.
6	IP Phones off of the customer's LAN
7	Voice Mail: Intuity AUDIX shown connected via IP.
8	G700 Media Gateway connected via the LAN to the CLAN board located in the MCC1 Media Gateway. The S8300 Media Server in an LSP configuration is located in the G700 Media Gateway. In the event of a loss in communication between the S8700 and the G700 the LSP will provide call processing for the endpoints that register with it.
9	DCP Phones: Avaya Multi-Function Digital Phones.
10	Analog Connectivity: Analog phones, lines and trunks.

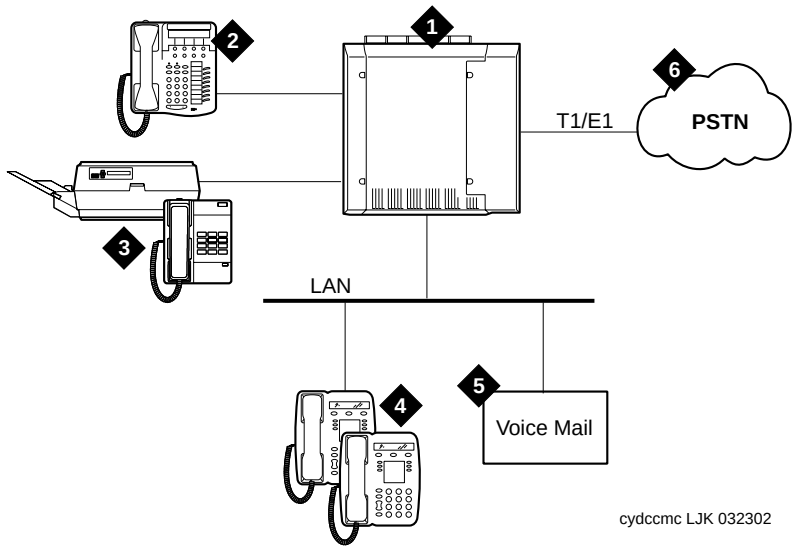
Multi-Site Environment



In a multi-site environment there are a number of standalone solutions networked together using protocals such as QSIG or DCS. Each location has it's own processing capability and seperate managed configurations. These configurations can be a mix of different solutions. Voice mail can be shared or networked.

Shown: The S8700 Media Server in a Multi-Connect configuration (top middle) networked with a S8300 Media Server in a G700 Media Gateway (botton left) and a DEFINITY CSI (bottom right).

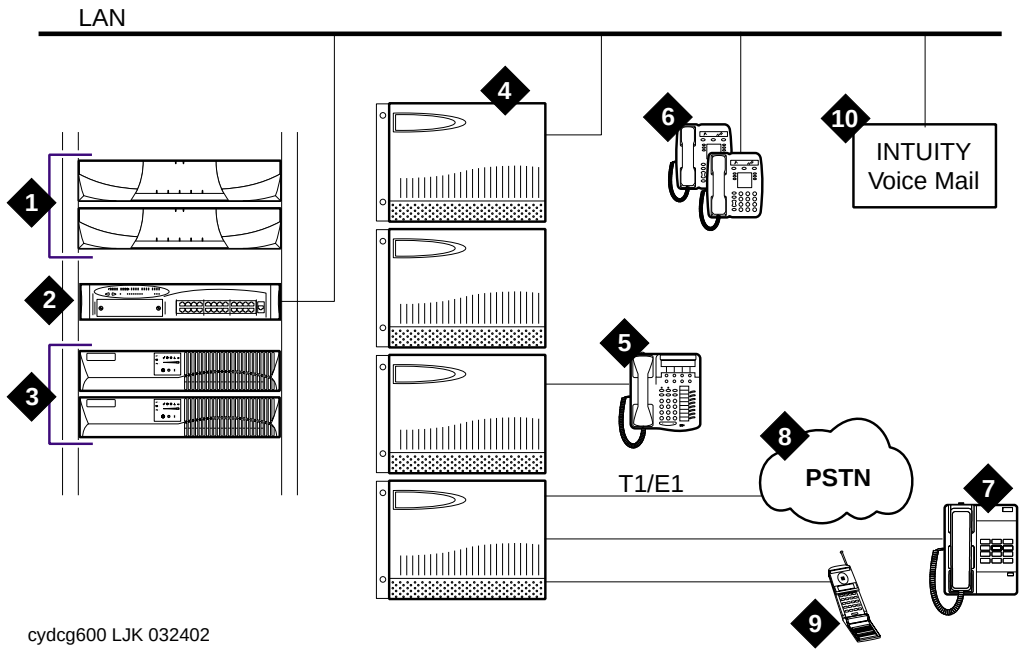
Medium Customer Configuration Using DEFINITY CSI with a CMC1 Media Gateway



Medium Customer Configuration Using DEFINITY CSI with a CMC1 Media Gateway

Item #	Description
1	The DEFINITY CSI with the CMC1 Media Gateway
2	DCP Phones: Supports 2-wire and 4-wire Avaya Multi-function phones.
3	Analog Capabilities: Analog trunks, stations and FAX machines.
4	IP Phones: Avaya 4600 series phones.
5	Voice Mail: Inutity AUDIX shown using IP connectivity.
6	T1/E1 Connectivity: • The T1 (or T-1)carrier is the most commonly used idgital line in the United States, Canada or Japan. • The E1 (or E-1) carrier isa a European digital transmission format. It is the equivalent of the North American T-carrier system format.

Medium Office Configuration using the S8700 Media Server with IP Connect



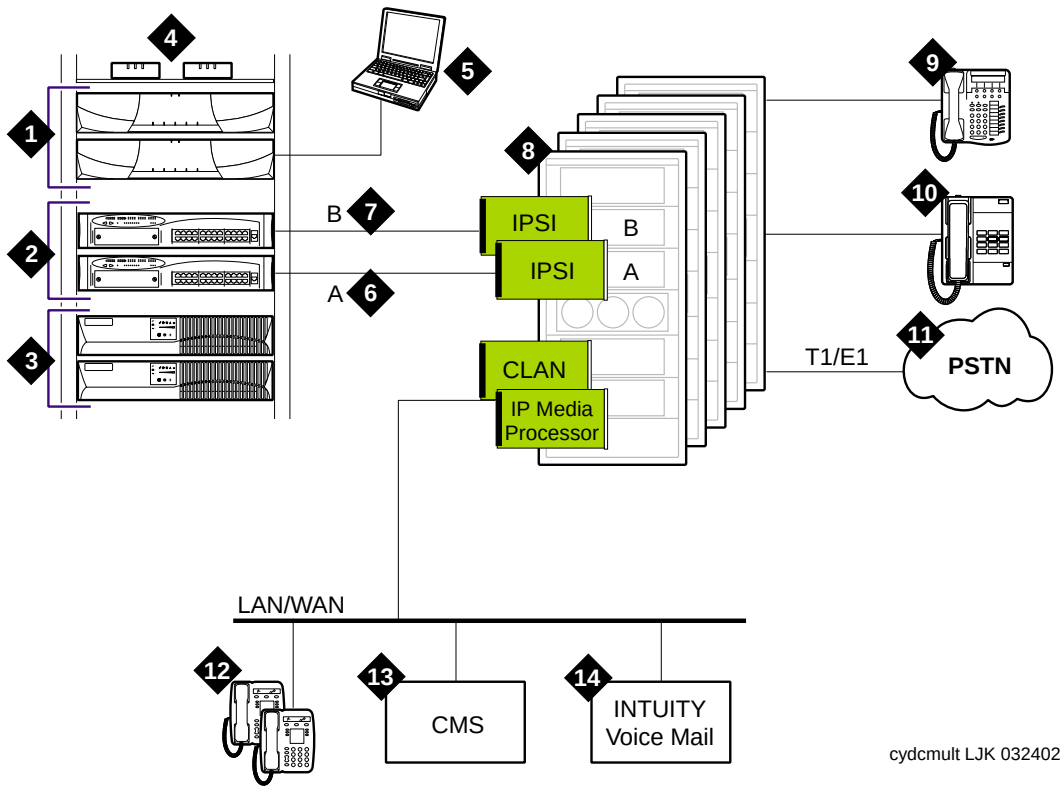
Medium Office Configuration using the S8700 Media Server with IP Connect

Item #	Description
1	S8700 Media Servers
2	Ethernet Switch: A device that provides for port multiplication by having more than one network segment. The Etherent Switch could be provided by Avaya or already exist in the customer's network.
3	UPS: Used to provide power during an outage and allow for a graceful shutdown of the server.
4	G600 Media Gateway: Connected to the S8700 Media Server by IP. This is the only supported connectivity in the IP Connect configuration. The G600 contains 10 slots and can use traditional DEFINITY circuit packs.
5	DCP Phones: Supports 2-wire and 4-wire Avaya Multi-function phones.
6	IP Phones: Avaya 4600 series IP phones.

Medium Office Configuration using the S8700 Media Server with IP Connect — *Continued*

Item #	Description
7	Analog Connectivity: Stations, lines, trunks and FAX machines
8	T1/E1 Connectivity: • The T1 (or T-1) carrier is the most commonly used digital line in the United States, Canada and Japan. • The E1 (or E-1) carrier is a European digital tranmission format. It is the equivalent of the North American T-carrier system format.
9	Wireless
10	Voice Mail System: Intuity AUDIX is shown connecting via IP.

Large Customer Configuration using S8700 Media Server in a Multi-Connect Configuration



Large Customer Configuration using S8700 Media Server in a Multi-Connect Configuration

Item #	Description
11	The S8700 Media Server in a Multi-Connect Configuration.
12	Ethernet Switches: Must be Avaya provided.
13	UPS: Provides power hold over when commercial power shuts down. The UPS units are duplicated in this configuration (one for each server).
14	Modems for services access.
15	Services LAPTOP Connectivity.
16	Dedicated LAN A for control data.
17	Dedicated LAN B for control data.
18	Port Networks

Large Customer Configuration using S8700 Media Server in a Multi-Connect Configuration — *Continued*

Item #	Description
19	DCP Phones: Avaya Multi-function Digital Terminals.
20	Analog Connectivity: Analog phones, FAX machines and trunks.
21	T1/E1 Connectivity: • The T1 (or T-1) carrier is the most commonly used digital line in the United States, Canada and Japan. • The E1 (or E-1) carrier is a European digital transmission format. It is the equivalent of the North American T-carrier system format.
22	IP Phones: Avaya 4600 series IP phones.
23	Call Management System (CMS): Used in a call center environment to collect and store data for reports and management.
24	Voice Mail: Intuity AUDIX shown in this configuration connecting via IP.

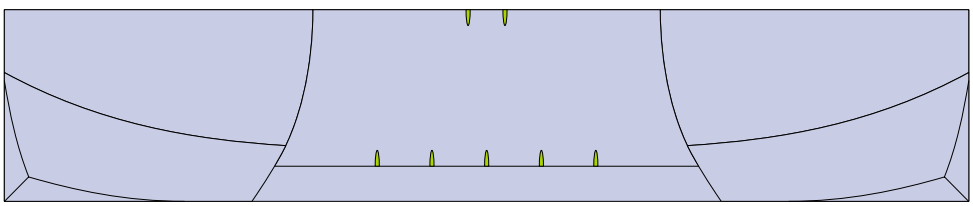
S8700 Media Server for Multi-Connect Configurations

May 2002

Overview

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 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 (PC), and internet addresses. The servers are duplicated in a S8700 Multi-Connect solution..

S8700 Media Server



scds8700 KLC 031402

Detailed Description

The S8700 Media Server for Multi-Connect Configurations uses a Linux platform on an Intel server. It is derived from the current 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 LAN for transport of the control data.

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

High Level Capabilities

The S8700 Multi-Connect provides a large scale solution with a high number of endpoints. Specifically, it supports the following high level capabilities. Refer to the table that follows. For more information about system capacity, refer to the System Capacity Tables.

Capability	S8700 Multi-Connect
Call Processing Feature Set	MultiVantage Software
Duplication options available	Duplex, high, and critical
Port Network Connectivity	Center Stage Switch or ATM, IP Network, Direct
Media Gateways	SCC1, MCC1, G700
Port Network quantity, maximum	44 – Center Stage Switch (CSS) or 64 – ATM-PNC
Survivability Options	Local
Survivable Options	S8300 Media Server in a Local Survivable Processor (LSP) configuration
Remote Office Gateway	Supported
Locations - maximum	44
Port Networks per IPSI	Up to 5 (2 IPSIs used with High or Critical Reliability)
Modem calls	Supported
Wideband connections	Supported

Required System Components

The S8700 Multi-Connect uses the following components:

- Intel Server
- IP Server Interface Card (IPSI) TN2312AP
- Cajun Ethernet Switch P133/P134 or P333/P334
- Powerware 700VA/1500VA online UPS
- USB Modem – ACM Compliant
- Media Gateways: MCC1 or SCC1
- "Avaya MultiVantage Software". For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.

Each of the required components is described in the following sections.

S8700 Media Server

Characteristics of the S8700 Media Server for Multi-Connect Configurations include:

- Sufficient 10/100 Ethernet ports to support IPSI network control links 1 & 2, services access, duplication, and administration/alarming.
- IDE hard disk
- IDE CD ROM
- 100 meters distance limitation between S8700 Media Servers
- Support for global power
- Storage media for OS, customer translations, and maintenance software
- Supports USB port connectivity for modem
- 128 MB Flash Card for removable media
- Provides monitoring and alarming
- Supports remote call out alarming from either server
- SNMP alarming

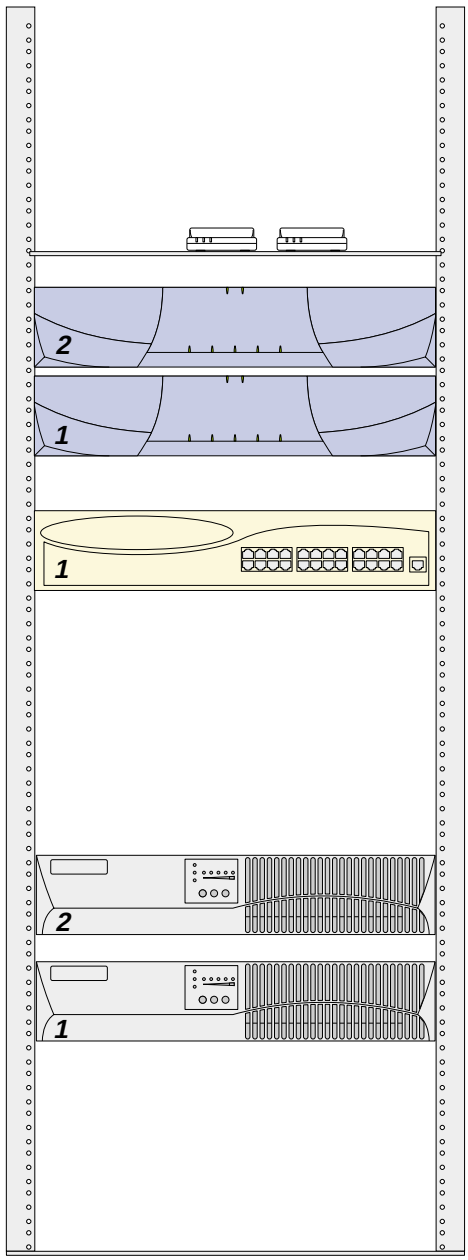
Communication Complex Configuration

S8700 Multi-Connect configurations use two S8700 Media Servers that are mounted in a EIA Standard 310-D 19 inch rack. The two servers can be cabled a maximum of 100 meters apart. The following is a detailed description of the entire 19 inch data rack control configuration.

- One DAJ1 duplication circuit board cable (yellow cable), 5 meters in length with duplex SC fiber connectors on each end. This cable connects the DAJ1s from one S8700 server to the other.
- Two services laptop cables (one per S8700 server) (black cables), 5 meters in length with RJ45 connectors.
- Four green five meter length server-to-Ethernet and UPS-to-Ethernet cables, and 2-64 green cables of varying lengths, in the range of 5-50 meters, connect from Ethernet switch to PN switches and to the front of IPSI circuit packs.
- At least one configured Ethernet switch.

Refer to the following figure for an example of a Communication Control Complex.

Communication Control Complex



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IP Server Interface (IPSI) (TN2312AP)

In the S8700 Multi-Connect the bearer and control paths are separate. Control information for port networks travels over a dedicated LAN through the Ethernet switch and terminates on the S8700 Media Server at one end and a TN2312AP IP Server Interface (IPSI) on the other.

Many of the services required for a port network are available on the IPSI board such as tone generation, tone detection, call classification equivalent to a TN2182C, and clock generation.

Each IPSI will have one 10/100BaseT Ethernet connection to the Ethernet switch. In the S8700 Multi-Connect, each IPSI normally controls five port networks by tunneling control messages over the bearer network to port networks without IPSIs. Further, IPSIs cannot be placed in a port network that has either a Stratum 3 clock or DS1-Converter.

To determine the number of IPSI-connected port networks recommended to support a Multi-Connect configuration, use the following formula:

([number of IPSI connected PNs] divided by 5) + 1

For example: You have 20 PNs, so you need 20/5 = 4. Always round up! In this example, you need 4 + 1 or a minimum of 5 IPSIs to support the 20 PNs. There is an absolute minimum of 2 IPSIs per configuration.

NOTE:
The addition of 1 IPSI provides additional fault tolerance by adding excess IPSI capacity.

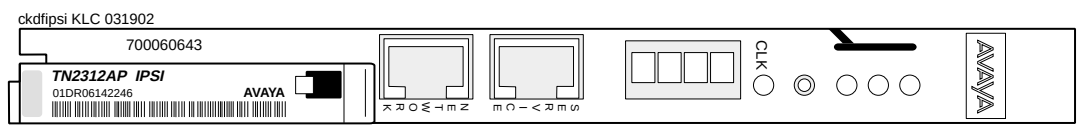
IPSI IP addresses are typically assigned automatically using DHCP service. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance and firmware downloads.

The IPSI has the following characteristics:

- Always resides in the tone/clock slot
- Has 10/100 BaseT interface for connection to server
- Has 10/100 BaseT interface for connection to services laptop
- Supports 8 global Call Classification ports
- Supports network diagnostic capabilities
- Provides PN clock generation and synchronization (Stratum 4 type II only)
- Provides PN tone generation
- Provides PN tone detection/global call classifier/international protocols
- Provides distributed PN packet interface
- Supports IPSI firmware download
- One IPSI per configuration provides serial number support for License File Feature Activation

The following diagram is of the IPSI faceplate.

IPSI Faceplate



Ethernet Switch

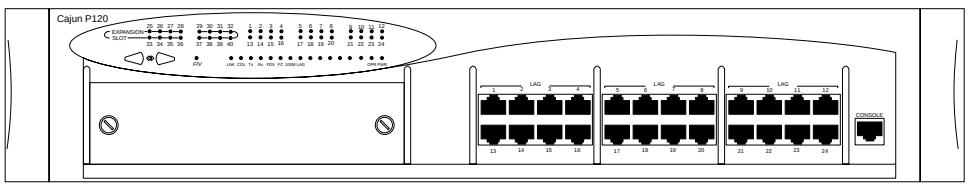
An Ethernet switch provides the connectivity between the servers and the IPSI Circuit packs located in some PNs. For Duplex reliability, one Ethernet switch is provided. For High and Critical reliability, the ethernet switches are duplicated. The S8700 Media Server supports two Ethernet connections to the control network Ethernet switch. One of these connections is used for Duplex reliability and both are used if High or Critical reliability is implemented.

The S8700 Multi-Connect will always require Cajun Ethernet (P13X or P33X) switches as part of the control complex. The Cajun Ethernet switch extends Ethernet connectivity to the port networks where it connects into an IP Server Interface Card (IPSI). One Ethernet switch is required for duplex reliability and two are required for high or critical reliability.

NOTE:
The call control network **must** be on a dedicated network **not** connected to any other networks.

The following diagram is of an Ethernet Switch.

Ethernet Switch



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UPS or Power Backup

S8700 Media Servers always require power backup, both to avoid power problems but also to ensure the ability to gracefully shut down system processes, should power fail. The 700 VA UPS provides approximately 30 minutes of power backup. Combinations of battery extension modules and a 1500 VA UPS provide up to 8 hours of power backup. Refer to the "Avaya UPS Information" section for more information.

Avaya’s UPS units send an alarm, via SNMP trap, when power fails. This initiates a graceful shutdown process of the Linux server, including the call processing software. When a separate 48V DC battery sting is used, it may be possible to send an alarm when voltage is below a threshold, but graceful shutdown will not be mechanized.

USB Modem

Each S8700 Media Server requires a Universal Serial Bus (USB) modem for maintenance access and remote call out. The modems can share a common phone line. Incoming calls are answered by the on-line server, meaning callers can access the off-line server via a telnet session. Each modem connects to a USB port on the S8700 Media Server. USB modems used must conform to the Communication Device Class (CDC) specification, and usually to the Abstract Control Model (ACM) sub-class. If the modem does not comply with the specification, it will not work with the driver provided by the S8700 Media Server.

Media Gateways

The MCC1, SCC1, and G700 Media Gateways are supported in a Multi-Connect configuration. Each is described in the following sections.

MCC1 Media Gateway

The MCC1 Media Gateway has the capacity to house up to 5 carriers and uses DEFINITY circuit packs. Doors on the front and rear of the cabinet protect internal equipment and allow easy access to circuit packs. The MCC1 Media Gateway could contain the following carriers:

- Port Carrier - contains port, service, tone/clock, and EI circuit packs
- Switch Node Carrier - contains switch Node Interface circuit packs composing the Center Stage Switch (CSS).
- Expansion Control Carrier - contains service and port slots

For more information, refer to "MCC1 Media Gateway".

SCC1 Media Gateway

The SCC1 Media Gateway consists of a single carrier. Up to 4 SCC1 Media Gateways can be connected together in one location. The SCC1 Media Gateway provides vertical slots for holding DEFINITY circuit packs. Cabinet clips connect the cabinets together and at the rear, a ground plate connects between cabinets for ground integrity. There are two types of SCC1 Media Gateways:

- Expansion Control Cabinet - contains service and port slots
- Port Cabinet - contains ports and interfaces to an Expansion Control Cabinet.

For more information, refer to "SCC1 Media Gateway".

G700 Media Gateway

The G700 Media Gateway contains slots for 4 Media Modules. A customer can mix and match Media Modules as well as stack and add additional G700 Media Gateways as their business grows in size. The S8700 Media Server with Multi-Connect can support up to 5 G700 Media Gateways. For more information, refer to "G700 Media Gateway in S8700 Multi-Connect".

S8700 Multi-Connect Reliability

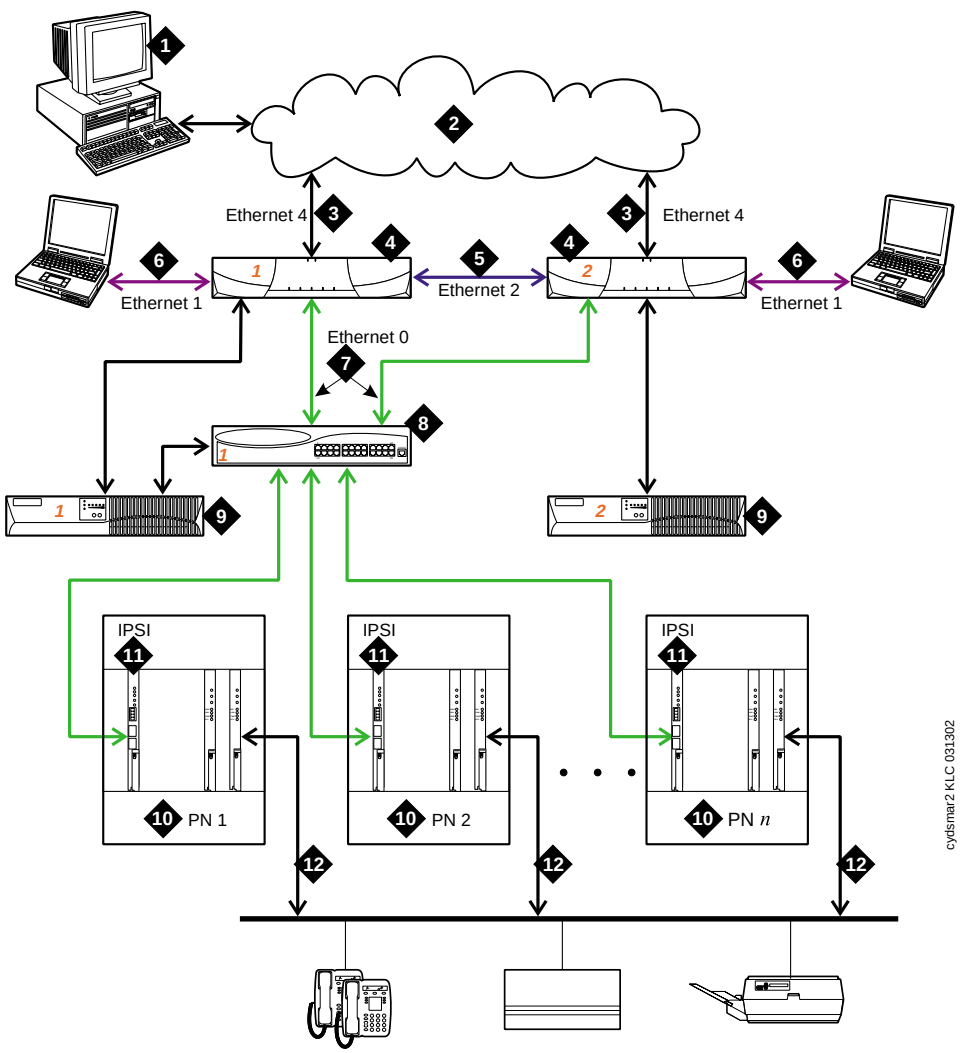
The following sections describe S8700 Multi-Connect reliability configurations: duplex reliability, high reliability, and critical reliability options.

Duplex Reliability Configuration

The duplex reliability option is the most basic option. It uses two S8700 Media Servers, one Ethernet switch, two UPS units (one for each server), and one IPSI in each IPSI-connected port network. Duplicating the UPS ensures no single point of failure in the processing platform.

The Ethernet connections for a S8700 Multi-Connect duplex reliability configuration are shown in the following figure and are described in the table that follows.

S8700 Multi-Connect Duplex Reliability Configuration



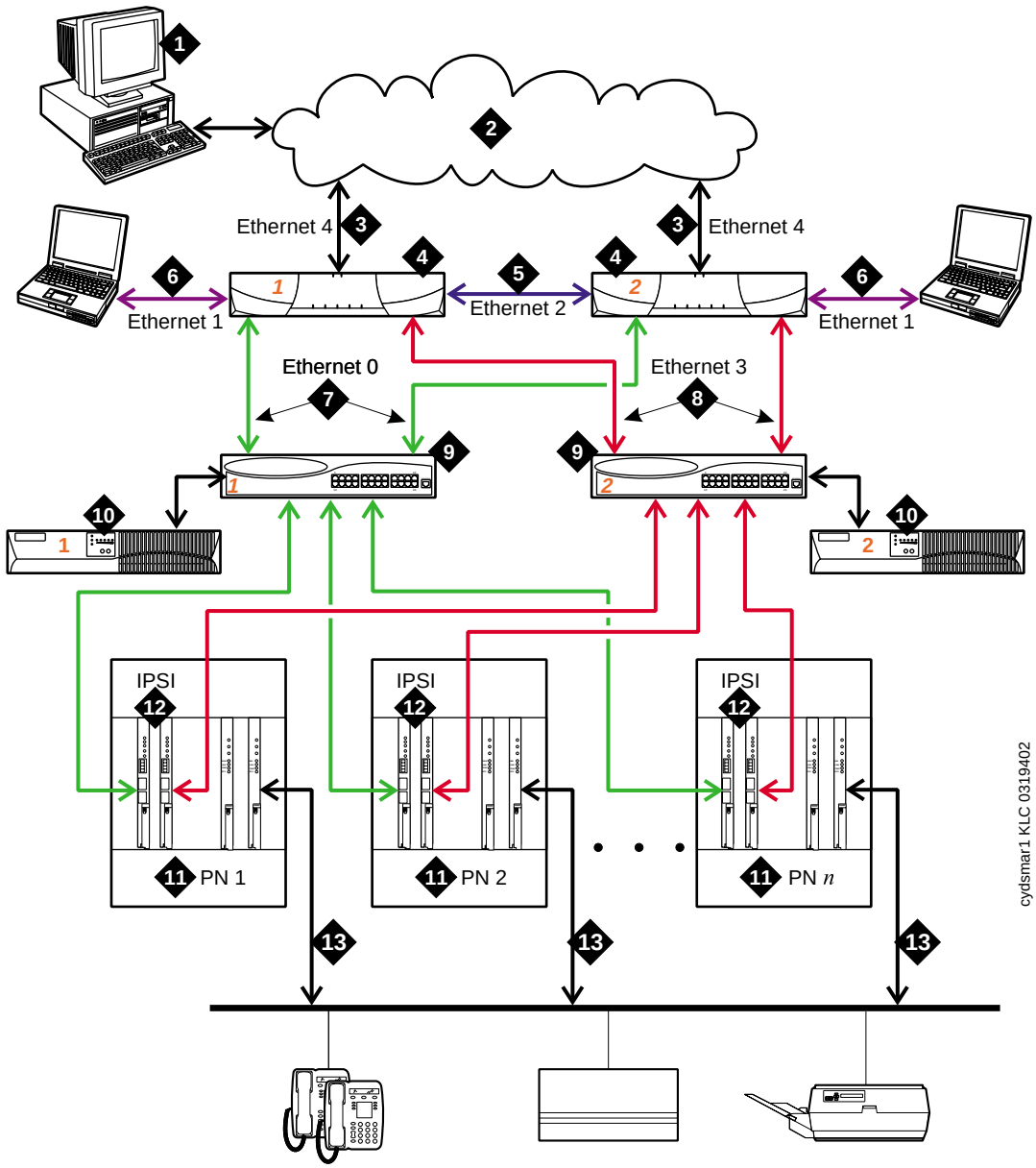
#	Description of Connection
1	Administration PC — used to access S8700 Media Server over the corporate LAN
2	Corporate LAN
3	Corporate LAN interface: default Ethernet 4 — S8700 Media Server's Ethernet link to the LAN. Used for administration and can be used for alarming via simple network message protocol (SNMP) traps to the Initialization and Administration System (INADS). The Ethernet connection to the corporate LAN in this figure is a non-dedicated network. IP addresses for the various S8700 Multi-Connect Media Server components must be administered with extreme care to prevent conflicts with other equipment that shares the LAN. In the default S8700 Multi-Connect configuration, all other Ethernet connections are dedicated, operating on their own closed LANs.
4	S8700 Media Server — two are always present, one in active mode and the other on standby.
5	Duplication interface: default Ethernet 2. The dedicated Ethernet connection between the S8700 Media Servers.
6	Services interface: default Ethernet 1. The server's dedicated Ethernet connection to a laptop. This link is active only during on-site administration or maintenance.
7	Network control A interface: default Ethernet 0. The server's Ethernet connection to 1 or 2 Ethernet switches. This private LAN carries the control signals for the S8700 Multi-Connect PNs.
8	Ethernet switch — at least one is required to support the S8700 Multi-Connect control network. If many PNs are present, two Ethernet switches may be daisy-chained together to provide sufficient Ethernet connections to the IPSI boards in the PN.
9	UPS — Keeps the S8700 Media Servers and Ethernet switches functional through brief power outages.
10	PN — provides telecommunications functions of the S8700 Multi-Connect Media Server.
11	IPSI — IPSI circuit pack carries the control network signals to the PNs. Provides tone-clock functionality.
12	Bearer Connectivity

High Reliability Configuration

The high reliability option builds on the duplex reliability option. It duplicates components so that no single point of failure exists in the control network. It uses two S8700 Media Servers, two IPSIs in each IPSI-connected port network, two Ethernet switches, and two UPS units. Voice and data bearer traffic between port networks is carried on a simplex network made up of one Expansion Interface in each port network. The EIs are cabled via lightguide fiber to either the CSS or an Asynchronous Transfer Mode (ATM) switch.

The Ethernet connections for a S8700 Multi-Connect high reliability configuration are shown in following figure and are described in the table that follows.

S8700 Multi-Connect High Reliability Configuration



#	Description of Connection
1	Administration PC — used to access S8700 Media Server over the corporate LAN
2	Corporate LAN
3	Corporate LAN interface: default Ethernet 4 — S8700 Media Server's Ethernet link to the LAN. Used for administration and can be used for alarming via simple network message protocol (SNMP) traps to the Initialization and Administration System (INADS).
4	S8700 Media Server — two are always present, one in active mode and the other on standby.
5	Duplication interface: default Ethernet 2. The dedicated Ethernet connection between the S8700 Media Servers.
6	Services interface: default Ethernet 1. The server's dedicated Ethernet connection to a laptop. This link is active only during on-site administration or maintenance.
7	Network control A interface: default Ethernet 0. The server's Ethernet connection to 1 or 2 Ethernet switches. This private LAN carries the control signals for the S8700 Multi-Connect PNs when possible. Control network A is considered the primary control network because it connects to the primary IPSI board in a PN.
8	Network control B interface: default Ethernet 3. The server's Ethernet connection to a duplicated set of Ethernet switches. This private LAN carries control signals for the PNs when the primary control network is unavailable. Control network B connects to the secondary IPSI board in a PN. When the system problem is resolved, primary control is returned to control network A.
9	Ethernet switch — at least one is required to support each control network.
10	UPS — Keeps the S8700 Media Servers and Ethernet switches functional through brief power outages. Usually, UPS 1 powers server 1 and its Ethernet switch and UPS 2 powers server 2 and its Ethernet switch.

#	Description of Connection
11	PN — provides telecommunications functions of the S8700 Multi-Connect Media Server. For High Reliability, each IPSI-connected PN contains a pair of IPSI circuit packs, one primary circuit pack and a duplicate secondary circuit pack as a backup. For Critical Reliability, the bearer network among the port networks is also duplicated. This means that two EI circuit packs or two ATM circuit packs are present in each PN instead of just one.
12	IPSI — IPSI circuit pack is duplicated in every IPSI-connected PN in a high reliability or critical reliability configuration. The primary IPSI is connected to control network A. It is the preferred IPSI because it has better control over activating the clock than the secondary IPSI. The secondary IPSI is connected to control network B. This IPSI takes over in case of problems with the primary control network. The S8700 Media Server regularly tests the duplicated IPSI to make sure it is ready for service, before returning control to the primary IPSI.
13	Bearer Connectivity

Critical Reliability Configuration

The critical reliability option is similar to the high reliability option. In addition, it duplicates the bearer IP network channels among the PNs. It also uses two S8700 Media Servers, two IPSIs, two Ethernet switches, and two UPS units.

Connectivity

The S8700 Multi-Connect Media Server uses IP, CSS, and ATM networks to connect to port networks. Each is described in more detail.

Port Networks

With S8700 Multi-Connect, all port networks are comprised of either the MCC1or SCC1 Media Gateways. In the case of a migration from the Avaya MultiVantage™ DEFINITY® Server R to the new S8700 Multi-Connect processor, the cabinet that was the PPN will be converted to a port network by replacing the control carriers (two if duplicated) with port carriers.

The new IPSI TN2312AP extends Ethernet Control connectivity from the new S8700 Media Server processor to the PNs. The IPSI replaces the TN2182B Tone Clock in each of the port networks that are IPSI connected. Not all port networks require the IPSI board. One IPSI-connected port network can control up to 4 other port networks.

When high or critical S8700 Media Server duplication is desired, two IPSI boards are required in each port network. In this case, the second IPSI goes into the B carrier in the slot reserved for an additional Tone Clock.

IP Network

MultiVantage Software sends control signals to the IPSI circuit packs in various PNs. It also receives control signals and forwards these messages to the proper port circuits.

Center Stage Switch Network

The CSS is a connection hub that provides port network communication. It can be used when the Multi-Connect Solution is composed of more than three port networks configuration. Often it is incorporated into smaller configurations to allow for growth. The CSS consists of from one to three switch nodes (SN). SNs are composed of one or two switch node carriers, depending on whether the system is being duplicated for critical reliability. Port network expansion depends on internal SN-to-SN traffic, but guidelines follow:

- One SN — expands from 1 to up to 15 port networks
- Two SNs — expands to up to 29 port networks
- Three SNs — expands to up to 43 port networks

ATM Network

The Asynchronous Transfer Mode (ATM) switch is a replacement option for the CSS or the direct-connect switch. Several Avaya ATM switch types can provide port network connectivity. Non-Avaya ATM switches that comply with the ATM standards set by the European Union can also provide port network connectivity.

With S8700 Multi-Connect, ATM-PNC allows any ATM switch or ATM network that complies with specified standards and capacities serve as the means to connect port networks. In this type of configuration, the ATM switch or network replaces the CSS. ATM-PNC is used to connect port networks within a single switch. The Wide Area Network (WAN) spare processor is not supported.

S8700 Multi-Connect Recoverability

In addition to the high reliability of the duplicated S8700 Multi-Connect Media Servers, the S8300 Media Server in a Local Survivable Processor (LSP) configuration can be used to provide backup service to the endpoints connected to a G700 Media Gateway. Additional recovery capability is embedded in the MultiVantage software that resides on both the S8700 and S8300 Media Server in a LSP configuration. Each is described in the following.

S8300 Media Server in a Local Survivable Processor (LSP) Mode

The LSP, located in the G700 Media Gateway, provides survivability when the S8700 Media Server is inaccessible. An LSP cannot control call processing for non-IP endpoints homed in traditional port networks. Each LSP has a copy of the S8700 Media Server's translations, which are updated regularly from the main S8700 Media Server via a virtual link through a dedicated IP network. Normally, all LSPs are in idle mode, not processing any calls. When the Media Gateway's Processor (MGP) or IP endpoints perceive the S8700 Media Server to be unreachable, they attempt to register with an LSP. Each LSP runs in license-normal mode until IP Phones or MGPs register with the LSP, which triggers the LSP to move into license-error mode. The LSP will run in license-error mode for a maximum of six days before transitioning into a no-license mode. When the customer performs a system reset on the LSP, the six-day license timer is reset after the primary controller, the S8700 Media Server, responds to a ping and no devices are registered with the LSP. The LSP does not actively take over when the primary controller becomes unreachable, but waits for MGPs and IP endpoints to register with it. Each S8700 Media Server can have multiple LSPs. Switchback from the LSP is a manual operation. Reset 3 on the LSP forces devices registered with it to return to their primary controller.

Power Outages

The system is designed to recover from a power outage or other failure instantly, regardless of the source of the failure. Each PN includes a set of segmented, parallel buses. If one of the paired segments fails, the other bus segment continues to handle communications. The UPS units supply power to the control complex.

Survivable Remote EPN

The Survivable Remote Expansion Port Network (SREPN) allows either an MCC1 or SCC1 port network to provide service to the customer when connectivity links fail. When the links to the PN are restored and stable, a logic switch in the SREPN is manually reset and the PN is reconnected to the links from the switch. The logic switch can either be reset locally at the SREPN or remotely via a dial-up connection to the SREPN.

The SREPN must be administered separately, not as a duplicated PN, to function in a disaster recovery scenario. It does not function as a SREPN without the administration of stations, trunks, and features to support its operation. SREPN is not compatible with ATM Port Network Connectivity (ATM-PNC).

System Management

Avaya's VisAbility™ Management Suite offers a comprehensive set of web-based network and system management solutions for the converged voice and data environment. Avaya VisAbility Management provides standard based infrastructure for integrated management applications.

Avaya's VisAbility Management Suite components include:

- **Avaya MultiVantage Fault & Performance Manager** - Fault Management will display a hierarchical view of devices and their status allowing users to view and isolate alarms and errors. Performance Management provides a comprehensive set of performance reports for trending and isolation of performance issues.
- **Avaya MultiVantage Configuration Manger** - Web-based configuration and administration for multiple devices providing a single point of entry for centralized management of distributed network and campus environments.
- **Avaya Directory Enabled Management** - Provides a converged voice and data directory enabled middleware solution that provides a platform for simplifying information and configuration management of Avaya solutions, allowing a real-time, integrated, directory-based read/write access to Avaya communications server based databases
- **Avaya Terminal Configuration** - Provides a web-based client application that allows end users to access Avaya MultiVantage devices to configure some of their personal station set preferences and features. Avaya Terminal Configuration runs on top of the Avaya Directory Enabled Management software. Permission to access a limited set of features is controlled by the administrator along with the ability to print a copy of the terminal users button labels.
- **Avaya MultiService Network Manager -** Provides discovery of IP devices showing a system view on a network map and is a central launch point for VisAbility applications.
- **Avaya VoIP Monitoring Manager** - Web-based tool for assisting the data and telecommunications engineers in isolating and identifying problems with VoIP calls.
- **Avaya Policy Manager** - Policy based network management system allowing users to configure the QoS parameters for the entire distributed network from a centralized location.

Refer to the Avaya Web site for more information:

Device Management for the Server

A Web server and a SNMP agent are provided for the server. The server provides a set of HTML Web pages that provide access to configuration of the Linux platform as well as telnet access to a Unix shell (bash) and some shell scripts that are used for platform management.

Avaya MultiService Network Manager Infrastructure

The optional Avaya MultiService Network Manager infrastructure can provide access to the management facilities that are used to administer and operate the system. CVSA is a

client/server component also referred to as the Avaya Management Portal (AMP), and is the entry point to the system.

The CVSA server, or back-end, component runs on a separate PC or Solaris workstation. The AMP is a Java applet based client that provides access to some basic CVSA services and serves as launching pad for a collection of network management applications.

The AMP provides the following services:

- Network viewing with standard and customizable filters
- Application launching
- Trap logging and management

Media Server Web Interface

Basic administration tasks are performed using the Media Server Web Interface. This browser-based server administration tool is an easy-to-use Graphical User Interface for performing commands such as:

- Checking server and process status
- Administering network features for the S8700 Media Server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services and installing license and authentication files.
- Installing new software and reconfiguring the server as needed
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs and running tests if needed.

The Web interface contains an extensive help system that describes all the Web screens and Media Server procedures.

Avaya VoIP Monitoring Manager

The Avaya VoIP Monitoring Manager is a QoS monitoring/feedback Windows 2000 software tool that allows the customer to visualize the real-time operation of VoIP systems. The tool provides information on QoS parameters related to VoIP quality. The Avaya VoIP Monitoring Manager provides the ability to view QoS related information through a client GUI application from the customer's LAN or via remote access. The Avaya VoIP Monitoring Manager is also capable of being configured to generate traps, associated with VoIP QoS, sent to any NMS.

The Avaya VoIP Monitoring Manager can receive RTCP packets from the following entities: IP telephones, IP soft phones, VoIP engines on media gateways and prowler boards. The RTCP data for current calls is published in the RTP MIB via the SNMP agent running on the server. The historical RTCP data is published in the VMON MIB. Both forms of RTCP data can be viewable by either field support, a system administrator or services personnel.

System Capacity

The following table presents a subset of S8700 Multi-Connect BHCC capacities.

Type	S8700 Multi-Connect
All Analog	342,000
General Business - non-ISDN	300,800
General Business - ISDN	192,300
ACD	140,500
ICM	112,100
OCM	230,500
General Business - IP	53,400
Call Center - IP	61,800

NOTE:
Any configuration that includes IP Solutions applications such as, Road warrior, Telecommuter or H.322 trunking, will impact the Busy Hour Call Completion system and processor capacity. If a customer is utilizing these applications in a high traffic system, it is recommended that the Avaya Technology and Consulting (ATAC) team be involved to review any potential traffic impact.

Adjunct Systems and Solutions

Avaya provides the following adjunct systems. For a description of each, refer to the Adjunct section.

- Voice messaging and response such as INTUITY AUDIX
- Call center systems such as Call Management Systems (CMS), CentreVu CMS, CentreVu Agent, CentreVu Supervisor, CentreVu Explorer, BCMSVu
- System Printer (via terminal server)
- Journal Printer (via terminal server)
- Call Accounting Systems (via terminal server)
- Call Detail Recording (CDR) (via terminal server)
- Remote Network Monitoring Services (RNMS)
- Avaya Secure Remote Access

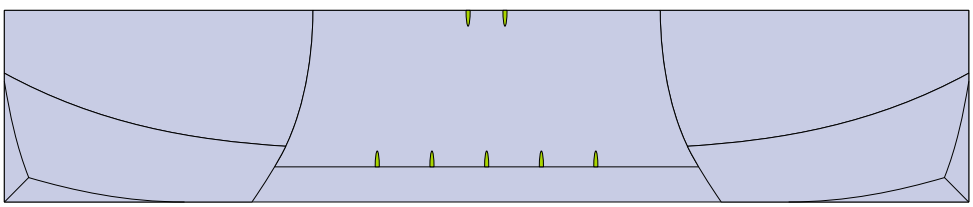
S8700 Media Server for IP Connect Configurations

May 2002

Overview

The Avaya™ S8700 Media Server for IP Connect Configurations (S8700 IP Connect) is an Internet Protocol (IP) PBX solution that routes voice, data, and video traffic between a mix of IP and traditional endpoints such as digital, analog, and ISDN. Its Intel-based server, modular media gateways, and IP-based connectivity provide building blocks that meet a variety of customer telephony needs.

S8700 Media Server



scds8700 KLC 031402

Detailed Description

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 up to 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 up to 100 meters cable distance apart.

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:

- S8700 Media Server (always duplicated)
- G600 Media Gateway
  - — Up to 4 G600 Media Gateways per PN
  - — Maximum of 64 PNs
- Scalable up to 12,000 IP endpoints
- Scalable up to 4,000 traditional stations and trunks
- 2 UPSs (one per Server)
- Avaya MultiVantage™ software
- Utilize customer IP network
- Leverage existing assets such as circuit packs and endpoints

Required System Components

The S8700 IP Connect consists of the following system components:

- Two duplicated S8700 Media Servers
- Two UPS units - one for each server
- USB Modem that is ACM compliant
- G600 Media Gateway
- At least one TN2312AP IP Server Interface (IPSI)
- At least one TN799DP C-LAN
- At least one TN2302AP IP Media Processor to support inter- and intra-port network bearer connectivity
- "Avaya MultiVantage Software". For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.

Each of the required components is described in the following sections.

S8700 Media Server

Characteristics of the S8700 Media Server for IP Connect Configurations include:

- Sufficient 10/100 Ethernet ports to support IPSI network control links 1 & 2, services access, duplication, and administration/alarming
- SNMP alarming
- IDE hard disk
- IDE CD ROM
- 100 meters distance limitation between S8700 Media Servers
- Support for global power
- Storage media for OS, customer translations, maintenance software, etc.
- Supports USB port connectivity for modem
- 128 MG Flash Card for removable media
- Provides monitoring and alarming
- Supports remote call out alarming from either server
- Modem connectivity for maintainability when an IP network is not available

The S8700 Media Server must be mounted in a EIA-310-D compliant 19 inch rack.

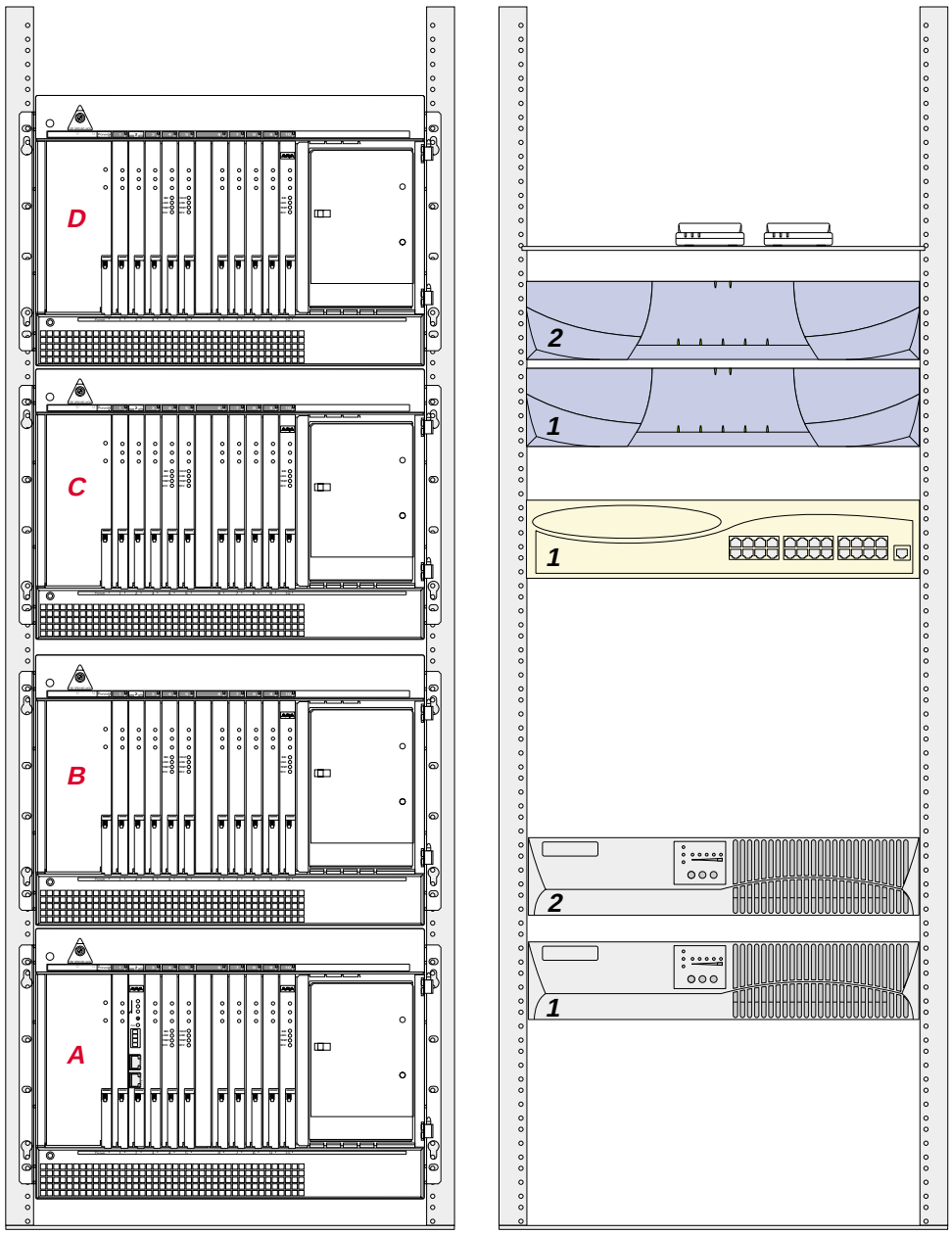
Communication Complex Configuration

Configurations for S8700 IP Connect use two S8700 Media Servers that are mounted in a 19 inch rack. The following is a detailed description of the communication complex configuration.

- One DAJ1 duplication circuit board cable (yellow cable), 5 meters in length with duplex SC fiber connectors on each end. This cable connects the DAJ1s from one S8700 server to the other.
- Two services laptop cables (one per S8700 server) (black cables), 5 meters in length with RJ45 connectors.
- Four green five meter length server-to-Ethernet and UPS-to-Ethernet cables, and 2-64 green cables of varying lengths, in the range of 5-50 meters, connect from Ethernet switch to PN switches and to the front of IPSI circuit packs.
- At least one configured Ethernet switch. For example:
    - Avaya provided Cajun P133 or P134
    - Avaya provided Cajun P333 or P334

Refer to the following figure for an example of a communication complex configuration.

Communication Complex Configuration



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UPS or Power Backup

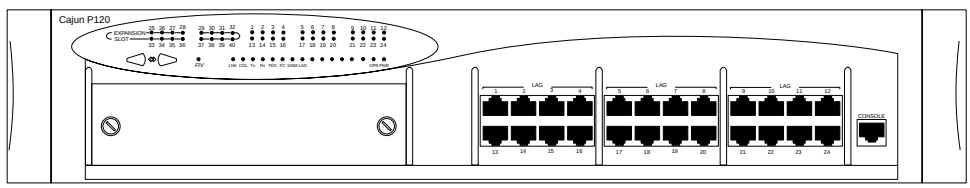
S8700 Media Servers always require power backup, both to avoid power problems but also to ensure the ability to gracefully shut down system processes, should power fail. The 700 VA UPS provides approximately 30 minutes of power backup. Combinations of battery extension modules and a 1500 VA UPS provide up to 8 hours of power backup. Refer to the "Avaya UPS Information" section for more information.

Avaya's UPS units send an alarm, via SNMP trap, when power fails. This initiates a graceful shutdown process of the Linux server, including the call processing software. When a separate 48V DC battery sting is used, it may be possible to send an alarm when voltage is below a threshold, but graceful shutdown will not be mechanized.

Ethernet Switch

Connectivity from the S8700 Media Servers to the port network IPSI circuit pack is provided via the customer's LAN. The minimum form of a customer's LAN in a small configuration is an Ethernet switch, which provides connectivity between the servers and the IPSI circuit packs located in some port networks. The Ethernet switch should support 802.ip/Q, VLAN, 10/1000 mbps.

Ethernet Switch



swdcp120 KLC 031902

USB Modem

Each S8700 Media Server requires a Universal Serial Bus (USB) modem for maintenance access and remote call out. The modems can share a common phone line. Incoming calls are answered by the on-line server, meaning callers can access the off-line server via a telnet session. Each modem connects to a USB port on the S8700 Media Server. USB modems used must conform to the Communication Device Class (CDC) specification, and usually to the Abstract Control Model (ACM) sub-class. If the modem does not comply with the specification, it will not work with the driver provided by the S8700 Media Server.

Port Networks

PNs provide the telecommunications functions to the S8700 Media Server. The S8700 Media Server for IP Connect supports a maximum of 64 Port Networks with up to 4 G600 Media Gateways in each for a total of up to 256 maximum G600 Media Gateways. Mixing of G600 Media Gateways with traditional Expansion Port Network cabinets – CMC1, SCC1 and MCC1, is not supported.

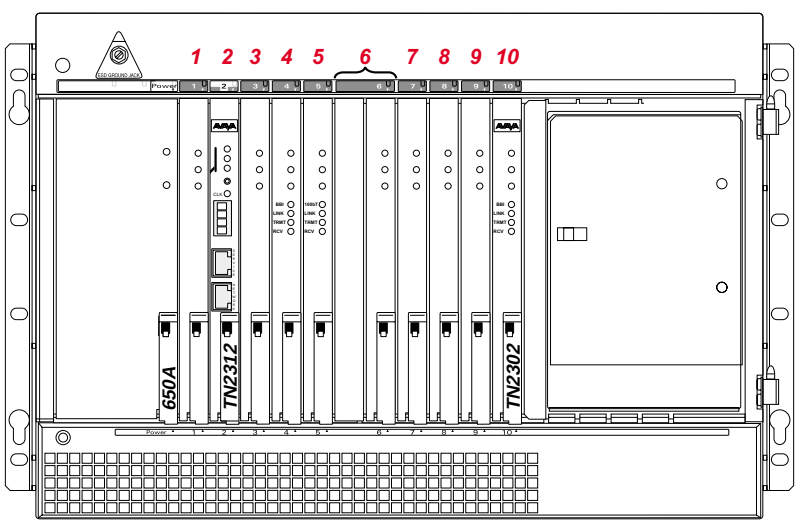
PNs have the following characteristics:

- Contain circuit packs to which communications devices or networks connect
- Contain communications busses for connecting circuit packs to each other, in other words, switching
- Are physically located within cabinets that supply power and environmental diagnostic equipment
- Connect to the S8700 Media Server through the IPSI via customer's LAN

G600 Media Gateway

Refer to the following diagram of a G600 Media Gateway.

G600 Media Gateway



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The G600 Media Gateway has the following characteristics:

- 19 inches wide, 13 inches high and 21 inches deep. It is designed for rack mounting only.
- 10 universal slots plus one power supply slot are available
- Maximum of 64 port networks.
- Maximum of 4 G600 Media Gateways per PN. Each PN must be collocated in the same 19 inch data rack due to TDM cable length

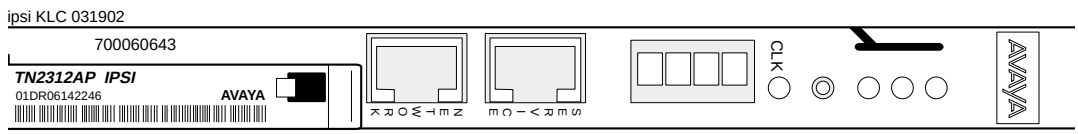
- A PN consists of a control G600 Media Gateway that is designated A, and second, third, and fourth optional G600 Media Gateway designated as B, C, and D respectively.
- Cabinet address IDs within the port network are set via a DIP switch on the backplane of each G600 Media Gateway.
- Circuit packs are inserted and removed from the front of the cabinet. Cabinet I/O is through the back and through a front cable pass-through slot on the right.
- The G600 Media Gateway is AC powered only. There are no internal batteries and internal DC power is not an option. External UPS or DC powering is supported.
- An RJ45 patch panel is recommended for cross-connect to LAN or wall field.

IP Server Interface (TN2312AP)

The IP Server Interface (IPSI) provides transport of CCMS messages over IP allowing the S8700 Media Server to communicate with the PNs. The IPSI is required to provide control network signaling over the customer's LAN and WAN. Tone generation, tone detection, global call classification, as well as stratum 4 type clock generation are provided on the IPSI board.

The following diagram is of the IPSI faceplate.

IPSI Faceplate



The IPSI has the following characteristics:

- Always resides in the tone/clock slot
- Has 10/100 BaseT interface for connection to server
- Has 10/100 BaseT interface for connection to services laptop
- Supports 8 global Call Classification ports
- Supports network diagnostic capabilities
- Provides PN clock generation and synchronization (Stratum 4 type II only)
- Provides PN tone generation
- Provides PN tone detection/global call classifier/international protocols
- Provides distributed PN packet interface
- Supports IPSI firmware download
- One IPSI per configuration provides serial number support for License File Feature Activation

IPSI IP addresses are typically assigned automatically using DHCP service. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance and firmware downloads.

C-LAN (TN799DP)

The C-LAN circuit pack, TN799DP provides call control for all IP endpoints connected to the S8700 Media Server for IP Connect. There is a maximum number of 64 C-LANs per configuration. The number of C-LANs required depends on the number of devices connected as well as the options being utilized by the endpoint. It may be advantageous to segregate IP voice control traffic from device control traffic as a safety measure.

The default value for C-LAN socket usage of H.323 Tie Trunks is determined by dividing the total number of H.323 Tie Trunk utilizing sharing by 31. Each IP endpoint requires the use of some number of C-LAN sockets, which is the software object used to connect a TN799 board to the IP Network. The TN799DP board supports up to 500 sockets.

The C-LAN differs from an IP Media Processor in that the former controls the call and the latter provides the codecs used for the call's audio.

To take advantage of downloadable firmware capability there must be at least one TN799DP C-LAN and access to the public Internet for firmware downloads to other downloadable circuit packs. Downloads and instructions have been posted to:

http://www.avaya.com/support/
Click on Online Services and then Download Software needed.

IP Media Processor (TN2302AP)

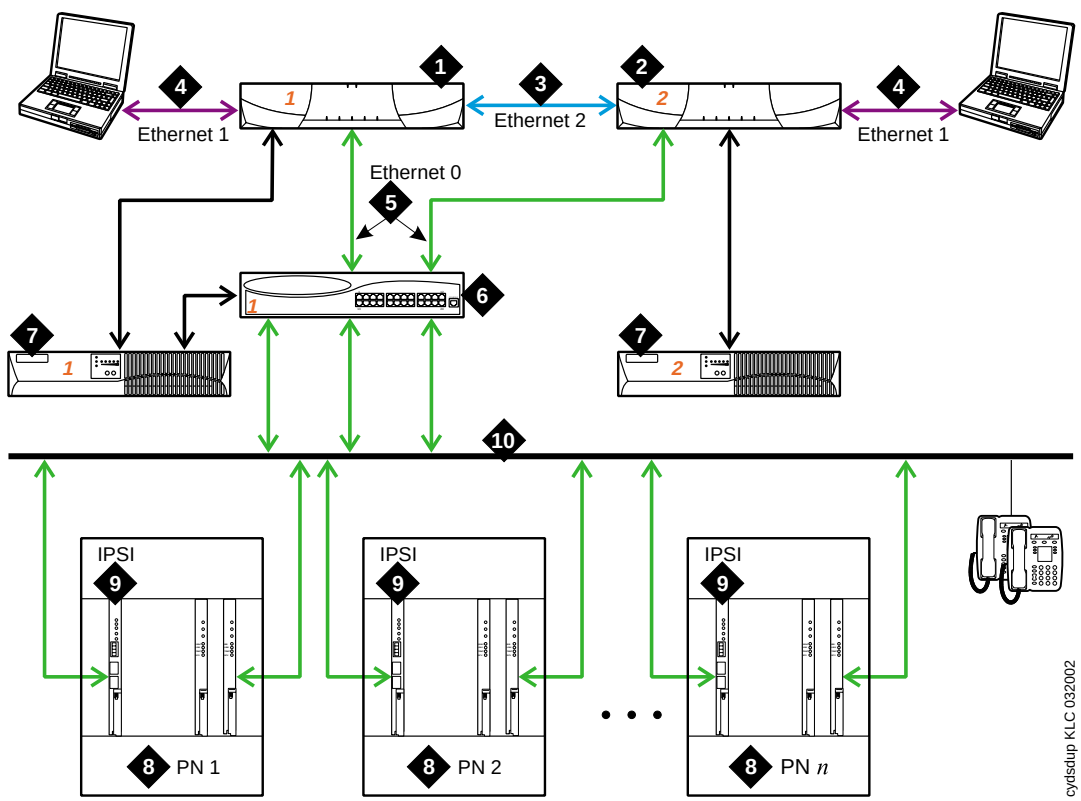
The S8700 Media Server for IP Connect requires resources on an IP Media Processor (TN2302AP) circuit pack for inter-port network bearer communications. The TN2302AP Media Processor includes a 10/100 BaseT Ethernet interface to support H.323 endpoints for IP trunks and H.323 end-points. The TN2302AP can perform echo cancellation, silence suppression, dual-tone multi-frequency (DTMF) detection, and conferencing.

S8700 IP Connect Reliability

The S8700 Media Server is duplicated. Tone-clock functionality in each G600 Media Gateway is configured simplex. It is provided for by the IPSI circuit pack. The control network, which consists of at least one 19 inch data rack mountable Ethernet switch, and only one IPSI circuit pack inserted into a G600 Media Gateway, is configured simplex. IP Bearer network is configured simplex via the IP Media Processor circuit pack, indirectly replacing traditional inter-PN connectivity via Expansion Interface circuit pack. Bearer connectivity such as Direct Connect, CSS, or ATM PNC are not supported by the S8700 Media Server with IP Connect Solution.

The S8700 IP Connect duplex reliability configuration is shown in the following figure and is described in the table that follows.

S8700 IP Connect Duplex Reliability Configuration



#	Description of Connection
1	S8700 Media Server 1 - two are always present, one in a active mode and the other on standby.
2	S8700 Media Server 2 - two are always present, one in a active mode and the other on standby.
3	Duplication Interface - The Ethernet connection between the two S8700 Media Servers.
4	Services Interface - The S8700 Media Servers dedicated Ethernet connection to a laptop. This link is active only during on-site administration or maintenance and the services interface can link to the non-active server through a telnet session.
5	Connection from the Servers to the Ethernet Switch.
6	Ethernet Switch - A device that provides for port multiplication by having more than one network segment. In an IP Connect environment, the Ethernet Switch should support 802.1 ip/Q, VLAN and 10/100 mbps.
7	UPS units - two required.
8	Port Network - optional configuration of Media Gateways that provides increased switch capacity.
9	IPSI - provides transport of CCMS messages over IP allowing the S8700 Media Server to communicate with the Port Networks.
10	Customer LAN

System Management

Avaya's VisAbility™ Management Suite offers a comprehensive set of web-based network and system management solutions for the converged voice and data environment. Avaya VisAbility Management provides standard based infrastructure for integrated management applications.

Avaya's VisAbility Management Suite components include:

- **Avaya MultiVantage Fault & Performance Manager** - Fault Management will display a hierarchical view of devices and their status allowing users to view and isolate alarms and errors. Performance Management provides a comprehensive set of performance reports for trending and isolation of performance issues.
- **Avaya MultiVantage Configuration Manger** - Web-based configuration and administration for multiple devices providing a single point of entry for centralized management of distributed network and campus environments.
- **Avaya Directory Enabled Management** - Provides a converged voice and data directory enabled middleware solution that provides a platform for simplifying information and configuration management of Avaya solutions, allowing a real-time, integrated, directory-based read/write access to Avaya communications server based databases
- **Avaya Terminal Configuration** - Provides a web-based client application that allows end users to access Avaya MultiVantage devices to configure some of their personal station set preferences and features. Avaya Terminal Configuration runs on top of the Avaya Directory Enabled Management software. Permission to access a limited set of features is controlled by the administrator along with the ability to print a copy of the terminal users button labels.
- **Avaya MultiService Network Manager -** Provides discovery of IP devices showing a system view on a network map and is a central launch point for VisAbility applications.
- **Avaya VoIP Monitoring Manager** - Web-based tool for assisting the data and telecommunications engineers in isolating and identifying problems with VoIP calls.
- **Avaya Policy Manager** - Policy based network management system allowing users to configure the QoS parameters for the entire distributed network from a centralized location.

Refer to the following Web site for more information:

Device Management for the Server

A Web server and SNMP agent are provided for the server. The server provides a set of HTML Web pages that provide access to configuration of the Linux platform as well as telnet access to a Unix shell (bash) and some shell scripts that are used for platform management.

Avaya MultiService Network Manager Infrastructure

The optional Avaya MultiService Network Manager infrastructure can be used to provide access to the management facilities that are used to administer and operate the system. CVSA is

a client/server component also referred to as the Avaya Management Portal (AMP), and is the entry point to the system.

The CVSA server, or back-end, component runs on a separate PC or Solaris workstation. The AMP is a Java applet based client that provides access to some basic CVSA services and serves as launching pad for a collection of network management applications.

The AMP provides the following services:

- Network viewing with standard and customizable filters
- Application launching
- Trap logging and management

Media Server Web Interface

Administration tasks are performed using the Media Server Web Interface. This browser-based server administration tool is an easy-to-use Graphical User Interface for performing server tasks such as:

- Checking server and process status
- Administering network features for the S8700 Media Server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services and installing license and authentication files.
- Installing new software and reconfiguring the server as needed
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs and running tests if needed.

The Web interface contains an extensive help system that describes all the Web screens and Media Server procedures.

Avaya VoIP Monitoring Manager

The Avaya VoIP Monitoring Manager is a QoS monitoring/feedback Windows 2000 software tool that allows the customer to visualize the real-time operation of VoIP systems. The tool provides information on QoS parameters related to VoIP quality. The Avaya VoIP Monitoring Manager provides the ability to view QoS related information through a client GUI application from the customer's LAN or via remote access. The Avaya VoIP Monitoring Manager is also capable of being configured to generate traps, associated with VoIP QoS, sent to any NMS.

The Avaya VoIP Monitoring Manager can receive RTCP packets from the following entities: IP telephones, IP soft phones, VoIP engines on media gateways and prowler boards. The RTCP data for current calls is published in the RTP MIB via the SNMP agent running on the server. The historical RTCP data is published in the VMON MIB. Both forms of RTCP data can be viewable by either field support, a system administrator or services personnel.

System Capacity

The following BHCC values for MultiVantage with S8700 IP Connect are provided as a preliminary introduction. For information, refer to the System Capacities Table.

Type	S8700 IP Connect
All Analog	342,000
General Business - non-ISDN	300,800
General Business - ISDN	192,300
ACD	140,500
ICM	112,100
OCM	230,500
General Business - IP	53,400
Call Center - IP	61,800

NOTE:
Any configuration that includes IP Solutions applications such as, Road warrior, Telecommuter or H.323 trunking, will impact the Busy Hour Call Completion system and processor capacity. If a customer is utilizing these applications in a high traffic system, it is recommended that the Avaya Technology and Consulting team be involved to review any potential traffic impact.

Adjunct Systems and Solutions

Avaya provides the following adjunct systems. For a description of each, refer to the Adjunct section.

- Voice messaging and response such as INTUITY AUDIX
- Call center systems such as Call Management Systems (CMS), CentreVu CMS, CentreVu Agent, CentreVu Supervisor, CentreVu Explorer
- BCMS VU Terminal and Servers
- Call Accounting Systems using TCP/IP connectivity
- System Printer (via terminal server)
- Journal Printer (via terminal server)
- Call Accounting Systems (via terminal server)
- Call Detail Recording (CDR) (via terminal server)
- Remote Network Monitoring Services (RNMS)
- Avaya Service Remote Access

S8300 Media Server with G700 Media Gateway

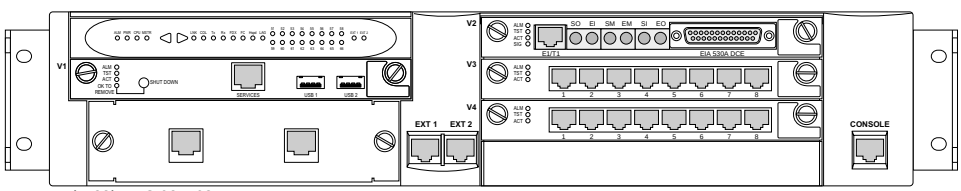
May 2002

Overview

The Avaya™ S8300 Media Server with the G700 Media Gateway is a unique solution set that converges the power of Avaya MultiVantage™ software with the power of distributed switching from the Cajun™ P330 line. It provides Internet Protocol (IP) PBX functionality using the H.323 and H.248 open standards and a Linux standard operating system. This solution is scalable, stackable, modular, and able to support a variety of distributed configurations.

The S8300 Media Server with the G700 Media Gateway is a Voice-Over-IP (VoIP) gateway that acts as an IP-PBX, a VoIP gateway to a variety of circuit switched interfaces. It also acts as an IP media management resource for conferencing, tone detection and generation, call classification, and audio transcoding..

S8300 Media Server with the G700 Media Gateway



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Detailed Description

The S8300 Media Server with the G700 Media Gateway delivers data, voice, fax, and messaging capabilities over an IP network while integrating with a variety of data infrastructures. It is mounted in a EIA-310-D 19 inch rack. It has redundant architectures and some of the parts are hot-swappable, which can be replaced in the field without disrupting system performance.

An S8300 Media Server that resides in the G700 Media Gateway provides call control for small systems, a maximum of 100 stations and 100 trunks. For more system capacity information, refer to the System Capacity Tables.

When more stations and trunks are desired, an S8700 Media Server in a Multi-Connect Configuration provides the control.

The S8300 Media Server with the G700 Media Gateway accommodates Cajun P330 Expansion Modules for full LAN/WAN connectivity.

An S8300 Media Server with the G700 Media Gateway incorporates the following features:

- Open IP standards H.248 and H.323
- Layer 2 switch
- Seamlessly integrates traditional circuit switched interfaces such as analog stations, analog trunks, FAX, Avaya multifunction digital stations, E1/T1 trunking, ISDN-PRI with IP switched interfaces such as IP telephones and IP trunking.
- Provides the high reliability of the Cajun Octaplane™ which has the capability to bundle the stackable components using a proprietary 4 dual-direction 4 gbps into a larger logical switch which is presented as a single network element to system management. Wired in a ring configuration, the Octaplane provides redundancy and re-routing should one of the members within the stack needs to be replaced or added.

Configuration Description

S8300 Media Server with the G700 Media Gateway may have the following components:

- S8300 Media Server
- G700 Media Gateway
- Media Modules
  - — DCP Media Module (MM712)
  - — Analog Trunk/Telephone Media Module (MM711)
  - — T1/E1 Media Module (MM711)
  - — VoIP Media Module (MM760)
- Octaplane including Cascade Stacking Modules and cables
- "Avaya MultiVantage Software". For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.

Each of the components is described in the following sections.

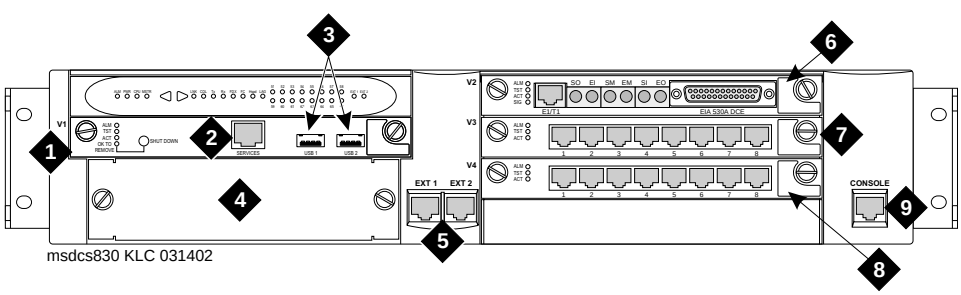
S8300 Media Server

The S8300 Media Server is a Pentium-based processor that runs on a Linux operating system. It is placed in slot V1 in the G700 Media Gateway. When installing or removing the S8300 Media Server, the S8300 Media Server and LED board must be removed or installed as a single unit.

The S8300 Media Server provides MultiVantage Software that is derived from DEFINITY® functionality. It provides a feature-rich environment. The S8300 Media Server does not support port networks or extended port networks.

The S8300 Media Server is **NOT** hot swappable. The following figure shows the S8300 Media Server in a G700 Media Gateway.

s8300 Media Server in a G700 Media Gateway



13 Media Module Slot V1. In this case, it contains the S8300 Media Server

NOTE:
An S8300 Media Server or an S8300 Media Server in a Local Survivable Processor (LSP) configuration can only be used in the Media Module slot labeled V1.

14 Service Ports

15 USB Ports

16 Cajun Expansion Module Slot

17 Dual 10/100Base-T Ethernet Switch Ports

18 Media Module Slot V2

19 Media Module Slot V3

20 Media Module Slot V4

21 Console (Serves as Serial On-Site Administration Interface)

The S8300 Media Server comes standard with:

- Avaya MultiVantage Software
- Administration and maintenance provisioning software
- 20G Hard drive
- 256 MB RAM
- Web server
- Linux OS (Red Hat v6.2)
- H.248 Gateway Controller (Megaco)
- H.323 Gate Keeper
- TFTP server

S8300 Media Server Module Interfaces

There is no external connector support for VGA video or PS2 keyboard and mouse. All customer and services access to the S8300 Media Server is via the network connections. There are USB connectors for optional customer connectivity such as the service's modem for remote access to the S8300 Media Server as well as an Ethernet connector for service's access on the faceplate.

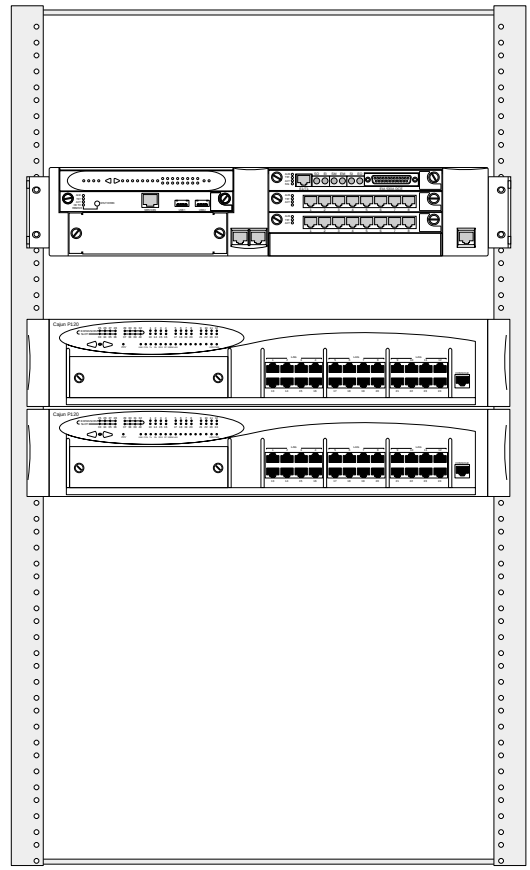
G700 Media Gateway

This section describes the G700 Media Gateway. It also briefly addresses other components within the G700 Media Gateway. The basic architecture of the G700 Media Gateway is as follows:

- Contains four Media Module slots
- Allows the insertion and functioning of modules that traditionally fit in the Cajun Expansion/Personality slot and the Cajun Stacking/Octaplane slot
- Fits in a standard EIA-310-D 19 inch, 23 inch rack with applicable brackets
- Usable on a table top
- Standard based 10/100 Ethernet Interface connection types. A wall field or breakout panel is not required.
- Emissions are in the Class B level in some specified configurations
- Uses front-loaded media modules
- Developed using ISO-compliant processes
- Uses materials that meet or exceed UL 94-VO requirements
- All interfaces with the exception of power, earth ground, and Octaplane are taken off the front of the unit. This facilitates movement and changes with the unit after initial installation.
- All cables that are plugged into the front of the G700 Media Gateway, whether in a table-top or rack-mount environment, are routed off to the side of the unit. A specified package of screws is included as part of this kit. One exception is the Octaplane, which is installed in the back of the box.

The G700 Media Gateway has a physical design that is similar to the Cajun stackable switching products. Refer to the following figure for an example.

G700 Media Gateway



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The G700 contains:

- VoIP resources
- A layer 2 switch
- Four Media Module slots
- A single, standard Cajun Expansion Module interface slot
- Provides for audio services on the LAN, such as conferencing, tones, and announcements

The G700 Media Gateway is designed to offer options and scalability. A customer can mix and match Media Modules as their business grows in size, as well as stack and add additional G700 Media Gateways if using the S8700 Media Server with Multi-Connect.

19 Inch G700 Media Gateway

The G700 Media Gateway is functional on its own, with other G700 Media Gateways, or in a stack that is mixed with G700 Media Gateways and Cajun P330 family devices such as the P333T, P333R, and P334. No more than 10 units can be in a stack, of which 5 can be G700 Media Gateways.

To provide power to IP telephones without additional cabling, stack the G700 Media Gateways with the Avaya P333T-PWR.

Media Modules are optional components that can be mixed and matched inside the G700 Media Gateway. The S8300 Media Server with the G700 Media Gateway uses Media Modules for traditional interfacing of service provider network access solutions such as T1/E1, ISDN-PRI, and Trunk Lines.

All Media Modules are field replaceable. All Media Modules are hot swappable and use universal slots. The S8300 Media Server, Cajun Expansion Module, and the Cascade Stacking modules have special consideration for inserting and shutting down.

Cajun Expansion Module

The G700 Media Gateway is architecturally based on the Cajun P330 family of data products. Therefore, customers have the flexibility of using any of the Cajun Expansion Modules with the G700 Media Gateway. This means that customers can add additional Cajun LAN/WAN expansion modules directly to the G700 Media Gateway without requiring additional hardware.

CAUTION:
Cajun modules - both Expansion Modules and Octaplane Stacking Modules - are NOT hot-swappable. They are service-disrupting and can reset the entire G700 Media Gateway upon insertion or removal. Power down the system prior to any insertion or removal of Cajun modules.

Cajun Stacking Module (Octaplane)

Each Cajun P330 switch and G700 has an expansion slot at the rear for the Octaplane stacking fabric and is 4 gbps in each direction. The use of a separate expansion slot for the Octaplane Stacking Module means that all of the front panel ports are available for network connections.

The design of the Octaplane Stackable Switching System enables the removal of a failed P330 or G700 Media Gateway from the stack without disrupting the integrity of the stack. Special considerations apply. The broken link is bypassed and the transmission continues within the stack.

The following are the major highlights of Octaplane Module Stacking:

- Octaplane
- Stacking module inserted in the back
- Connection: up & down or ring close
- Immediate close of ring in case of link or box failure

Cabling for the Cajun Stack

Cable	Description	Length	Length (metric)
X330SC Short Octaplane Cable (30 cm)	Short Octaplane cable - light- colored, used to connect adjacent switches or switches separated by one Backup Universal Power Supply (BUPS) unit.	12 in	30 cm
X330LC Long Octaplane Cable (2 m)	Long Octaplane cable - light- colored, used to connect switches from two different physical stacks	6 ft	2 m
X330RC Redundant Octaplane Cable (2 m)	Redundant cable - black, used to connect the top and bottom switches of a stack.	6ft	2 m
X330L-LC Extra Long Octaplane Cable (8 m)	Extra-Long Octaplane cable - light-colored, used to connect switches from two different physical stacks	24 ft	8 m
X330L-RC Long Redundant Octaplane Cable (8 m)	Long Redundant cable - black, used to connect the top and bottom switches of a stack.	24 ft	8 m
Cajun Stacking Module X330STK	Stacking Module provides two backplane links		

Power Supply

The G700 Media Gateway uses an AC power supply. A power supply located in the G700 Media Gateway converts AC input power to voltages needed by the system.

Motherboard

The Motherboard resides internally within the G700 Media Gateway. This board is responsible for the following:

- The VoIP Engine that performs IP/UDP/RTP processing, echo cancellation, G.711 A/mu, G.729 and G723.1 encode/decode, fax relay, silence suppression, jitter buffer management, packet loss concealment, etc. The VoIP Engine will support 64 channels. If more then 64 channels are needed then a VoIP Media Module will be required.
- The Gateway Processor complex which controls all the resources inside the Gateway. The Gateway Processor functions include the Media Module Manager, Tone/Clock and H.248 signaling to the Gateway Controller.
- An Avaya P330 processor complex which is based on the Cajun P330 data switch architecture. It provides an eight port Layer 2-switch function and manages the Expansion and Cascade modules.
- Provides the electrical and physical connectivity for the four media module slots.

NOTE:
The Motherboard is not replaceable in the field.

For additional information about the VoIP Media Module, refer to "VoIP Media Module (MM760)" on page 95.

Fans

The G700 Media Gateway contains four 12-volt fans. These are monitored and can be reported by SNMP to a management station.

LEDs

The S8300 Media Server with the G700Media Gateway uses two types of LEDs:

- Media Module LEDs
- System-Level LEDs

Media Module LEDs

Media Module LEDs have the following characteristics:

- The MM710 Media Module and S8300 Media Server have additional LEDs that function differently than the traditional LEDs.
- Each Media Module has at least three LEDs to indicate module/port status or maintenance and administration modes.
- The location, spacing, and labeling is fixed for all LEDs on every Media Module.
- The LEDs are mounted on the Media Module printed wiring board and are placed so that they show through an opening.

System-Level LEDs

The LED board provides visual indication of system and data port status, but also allows the customer to switch between status indication modes. In addition, the System Level-LEDs have the following characteristics:

- An LED board is located in the upper left front area of each G700 Media Gateway. The front of the LED board displays the LEDs in an oblong fascia panel.
- The LED board is intended to provide visual indication of system and Ethernet port status and to allow the customer to switch between status indication modes.

The LED panel needs to be removed when installing or removing the S8300 Media Server or Standby S8300 Media Server. The two components should be installed or removed as a unit.

NOTE:
The LED panel is not the same size as standard Media Module slots. Trying to insert a Media Module into the LED slot, or vice versa, will not work.

Gateway Software

Gateway software is responsible for individual Media Gateway operations, terminating H.248 on the G700 Media Gateway, and for interacting with maintenance operations.

Maintenance Software

The Media Server with the G700 Media Gateway has a dual maintenance strategy in that there is maintenance software that runs on both the G700 Media Gateway platform and the Media Server for the subsystems on the platform. This platform software performs initialization and motherboard maintenance, along with internal environmental monitoring.

The media modules, on the other hand, are tested and brought into service by the Media Server maintenance software after the G700 Media Gateway registers with the Media Server. While the G700 Media Gateway maintenance software is aware of the media modules, the modules and associated ports are controlled by the Media Server. Error logs are maintained on the Media Server.

Media Modules

Media Modules convert the voice path of the traditional circuits such as analog trunk, T1/E1, DCP to a TDM bus. The VoIP engine then converts the voice path from the TDM bus to a compressed or uncompressed and packetized voice over IP on Ethernet connection.

The Media Modules are housed in the G700 Media Gateway and interact with the motherboard and backplane.

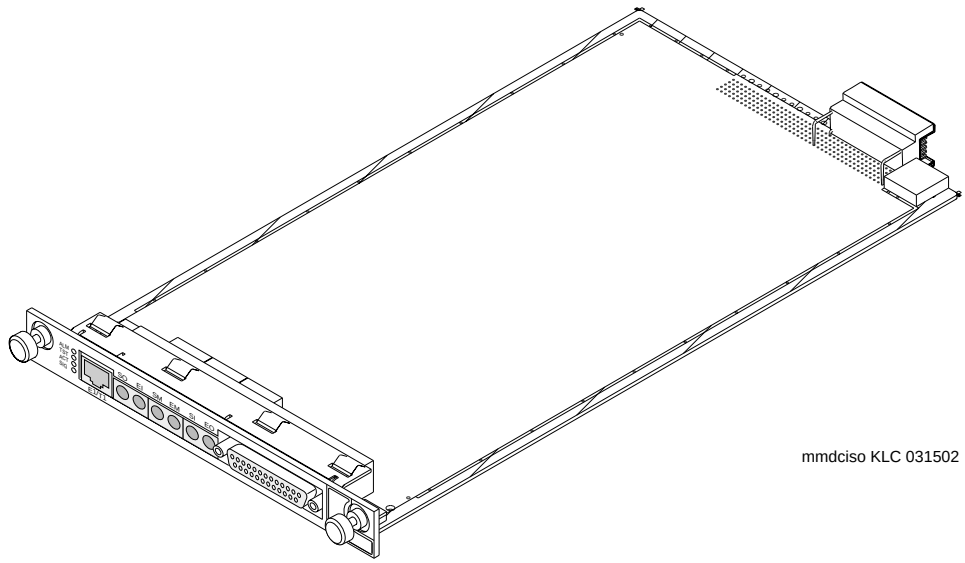
CAUTION:
Wear a wrist strap for Electrostatic Discharge Protection (ESD) when handling any Media Module.

There are four Media Modules:

- T1/E1 Media Module (MM710)
- Analog Trunk/Telephone Media Module (MM711)
- DCP Media Module (MM712)
- VOIP Media Module (MM760)

CAUTION:
Media Modules are locked into the chassis via a set of spring-loaded captive screws on the front of each module. Follow all warnings and cautions when attempting to remove these screws.

Media Module Top View



Media Module Front View

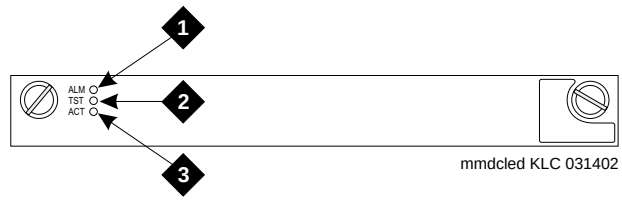


Figure notes

1 ALM - Alarm LED

2 TST - Test LED

3 ACT - * LED

LEDs

Although some Media Modules contain additional LEDs, each Media Module faceplate has a standard pattern of three LEDs that indicate the following conditions.

- Red indicates a fault condition - this LED is also on when the Media Module is physically inserted, and should turn off when the board initializes.
- Green indicates a test condition
- Yellow indicates an in use condition

Pathways for Media Module Control

Two pathways provide Media Module control:

- The TDM bus-based control channel interface (modeled after DEFINITY)
- An Ethernet interface

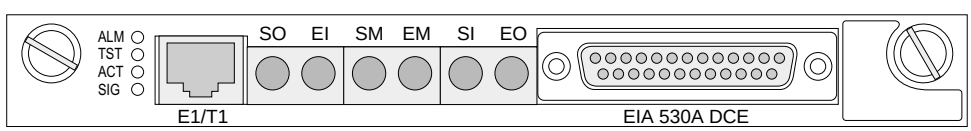
Media Module Descriptions

The following sections describe each of the Media Modules used in the S8300 Media Server with the G700 Media Gateway.

T1/E1 Media Module (MM710)

The MM710 media module is designed to terminate a T1/E1 connection. It has built-in Channel Service Unit (CSU) so an external CSU is not necessary. Refer to the following figure for an example of the MM710.

MM710



mmdc710 KLC 020402

The MM710 Media Module offers these functions:

- Software selectable T1 or E1 operation
- Integrated CSU
- Both A-law (E1) and Mu-law (T1) gain control and echo cancellation ability
- Bantam Racks
- D4, ESF, or CEPT framing
- 700A Loopback Jack
- ISDN PRI capability (23B + D or 30B + D)
- Line Coding: AMI, ZCS, B8ZS (T1) or HDB3 (E1)
- Trunk signaling for supporting US and International CO or tie trunks
- Echo Cancellation in either direction
- Fractional T1 support
- OIC DB 25-pin interface

For more information, refer to "T1/E1 Media Module (MM710)".

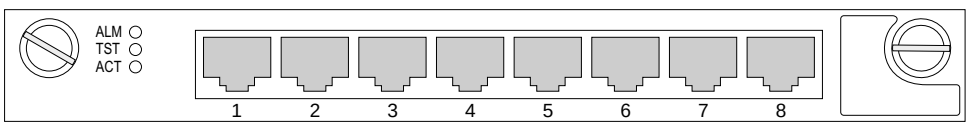
LEDs

Four LEDs are supported on the faceplate. They include the three standard Media Module LEDs and the SIG LED which indicates the MM710 is receiving a valid signal.

Analog Trunk/Telephone Media Module (MM711)

The MM711 Media Module provides analog trunk and telephone features and functionality. Refer to the following figure for an illustration of the MM711.

MM711



mmdc711 KLC 022702

The MM711 provides the administrator with the capability to configure any of the eight ports of this analog circuit pack as follows:

- Central Office trunk, either loop start or ground start
- Analog Direct Inward Dialing (DID) trunks, either wink start or immediate start
- 2-wire analog Outgoing CAMA E911 trunks for connectivity to the PSTN
- MF signaling is supported for CAMA ports
- Analog, tip/ring devices such as single-line telephones with or without LED message waiting indication

The MM711 also supports:

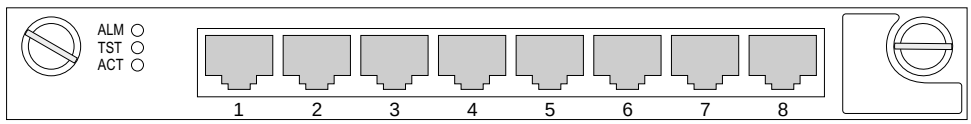
- Type 1 and Type 2 Caller ID
- Ring voltage generation for a variety of international frequencies and cadences
- A hard-wired ground wire is added for each IROB to earth ground

For more information, refer to "Analog Trunk/Telephone Media Module (MM711)".

DCP Media Module (MM712)

The MM712 Media Module allows connectivity of up to eight two-wire Digital Communications Protocol (DCP) voice terminals. Refer to the following figure for an example of the DCP Media Module.

MM712



mmdc712 KLC 022702

NOTE:
DCP is the Avaya proprietary signaling protocol used to interface the DCP-compatible Avaya digital voice to the S8300 Media Server with the G700 Media Gateway.

For more information, refer to "DCP Media Module (MM712)".

VoIP Media Module (MM760)

The MM760 Media Module is a clone of the motherboard VoIP engine. It provides an additional 64 VoIP channels with G.711 compression. Refer to the following figure for an example of a MM760.

MM760

mmdc760 KLC 022702

The capacity is 64 G.711 TDM/IP simultaneous calls, or 32 compression codec, G.729 or G.723, TDM/IP simultaneous calls. These call types can be mixed on the same resource; in other words, the simultaneous call capacity of the resource is 64 G.711 Equivalent Calls.

For more information, refer to "VoIP Media Module (MM760)".

S8300 Media Server in an LSP Configuration

The S8300 Media Server in a Local Survivable Processor (LSP) configuration uses the S8300 as its hardware component and requires a software license to activate the standby feature. This software allows the S8300 Media Server with a G700 Media Gateway to be a survivable call-processing server for remote and branch customer locations when used with the S8700 Media Server in a Multi-Connect configuration.

In the event that the link is broken between the remote G700 Media Gateway and the S8700 Media Server with Multi-Connect as the primary controller, or if the S8700 Media Server fails, the S8300 Media Server in an LSP configuration activates. The S8300 Media Server in an LSP configuration only takes over for those telephones connected to the G700 Media Gateway in which it is housed.

While the hardware for the S8300 Media Server and the S8300 Media Server in an LSP configuration is identical, it has the following software differences:

- Different IP Address than the S8300 Media Server it is supporting
- License File has six days of use once a G700 registers with it
- Receives Saved Translations from the primary server

The S8300 Media Server in an LSP configuration only supports the S8700 Media Server with Multi-Connect.

CAUTION:
The S8300 in an LSP configuration differs from other Media Modules because of its height. Plug the S8300 Media Server in an LSP configuration into **only** the left-most slot under the LED board and panel. When installing or removing, treat the S8300 Media Server and LED board as a single unit.

IP phones obtain their own IP address from a DHCP server. The DHCP server also sends a list of controllers and S8300 Media Servers in an LSP configuration and associated IP addresses. The IP phone then registers to the controller that corresponds to the first IP address in this list. When connectivity is lost between the controller and the endpoint, the endpoint registers with the second IP address in the list, and so on. This list is administrable for phones on the DHCP server.

The S8300 Media Server in an LSP configuration is bought with a special license that enables MultiVantage to run in a backup mode. When the S8300 Media Server in an LSP configuration becomes active and begins call processing, an alarm is raised notifying that the S8300 Media Server in an LSP configuration is in service, and a timer is started. This timer runs for six days, at which time call processing ceases. If the problem is fixed within that six-day period, the S8300 Media Server in an LSP configuration can be reset, all endpoints will then register with the primary controller which is the S8700 Media Server with Multi-Connect, and the timer will be reset to allow for six days upon another failure.

While the S8300 Media Server in an LSP configuration is not call preserving in all circumstances, although IP-to-IP calls will be preserved, the fail-over from the S8700 Media Server with Multi-Connect, which is the primary controller, to the S8300 Media Server in an LSP configuration is an automatic process and does not require human intervention. The fail-back from the S8300 Media Server in an LSP configuration to the primary controller is not automatic and requires a system reset on the S8300 Media Server in an LSP configuration. This breaks the communication between the S8300 Media Server in an LSP configuration and all registered endpoints and causes the endpoints to register with the primary controller. During fail-back to the primary controller, all calls will be dropped with the exception of IP-to-IP calls.

Translation changes made on the primary controller will be propagated to the S8300 Media Server in an LSP configuration through an automated process. Following the daily translation save on the primary controller, the time of which can be administered by the customer, or after any save trans is issued, the primary controller will copy translations to all LSPs.

The physical installation for the S8300 Media Server in an LSP configuration differs from other Media Modules because of its height. The S8300 Media Server in an LSP configuration still plugs into a Media Module slot but is restricted to the left most slot under the LED board and panel. When installing or removing the S8300 Media Server in an LSP configuration, the S8300 Media Server in an LSP configuration and the LED board must be removed as a single unit. This is because the LED faceplate will hit the tall components on the S8300 Media Server in an LSP configuration if it is not removed as well.

The S8300 Media Server and the S8300 Media Server in an LSP configuration cannot reside in the same G700 Media Gateway.

Customer Configuration Options

The S8300 Media Server with the G700 Media Gateway provides a standards-based, IP communications infrastructure without compromising the customer's applications, reliability, and multi-service networking. It can be installed for small customers or large customers with either branch offices or in a multi-site configuration.

The S8300 Media Server with the G700 Gateway provides the following Networking benefits:

- Provides the customer with the feature transparency across the network via QSIG or DCS+.
- Consistent user experience with the same user interface
- Unified System Management
- Applications - controlled by Avaya MultiVantage Software. Offers full enterprise feature set and it standards-based using H.248 for gateway control.
- Ease of moves, adds and changes with the same Administration interface.

Example Configurations

Small Customer Configuration

The following shows a small customer location with under 100 phones and 100 trunks.

Small Customer Configuration

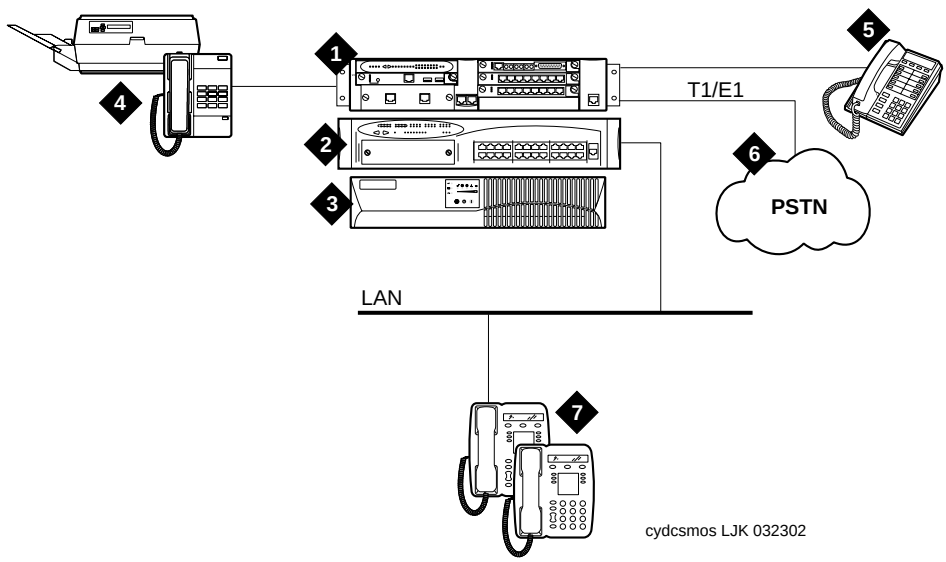


Figure notes

1 S8300 Media Server with a G700 Media Gateway. This configuration is using the T1/E1, DCP and Analog Media Modules.

2 Ethernet Switch: Can be customer or Avaya provided. This device provides for port multiplication by having more than one network segment.

3 UPS: suggested to provide power during an power outage and allow for a graceful shutdown of the server.

4 Analog Capabilities: Can provide for analog trunks, stations and lines.

5 DCP phones: Avaya Multi-Function Digital phones.

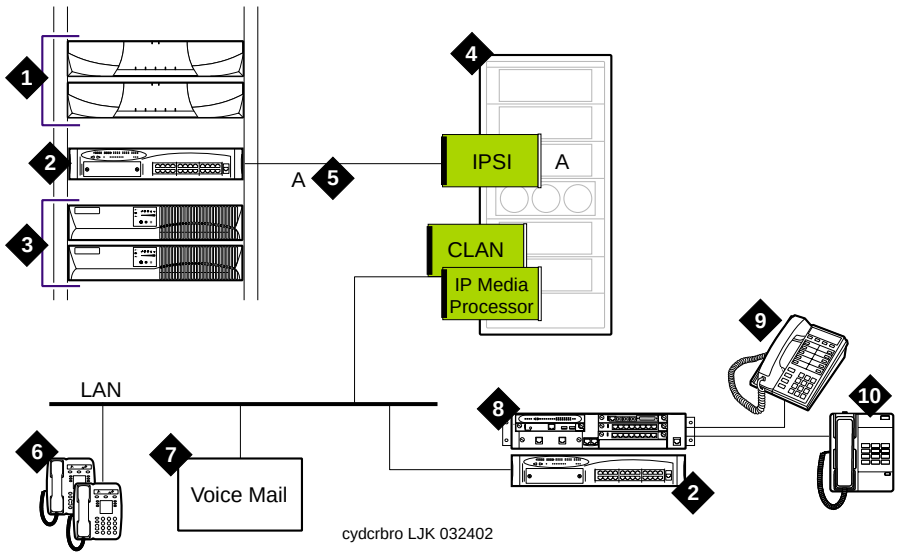
6 T1/E1 Connectivity:
- The T1 (or T-1) carrier is the most commonly used digital line in the United States, Canada and Japan.
- The E1 (or E-1) is a European digital transmission format. It is the equivalent of the North American T-carrier system format.

7 Avaya IP telephones

Remote Branch Office

A remote branch office location is apart of a larger enterprise network. In this configuration the S8700 Media Server in a Multi-Connect Configuration resides in the main location and controls the G700 Media Gateway in the remote location. It is highly desirable for the branch office to be survivable in the event of lost connectivity to the S8700 Media Server. Survivability is accomplished with the use of the S8300 Media Server in a LSP configuration

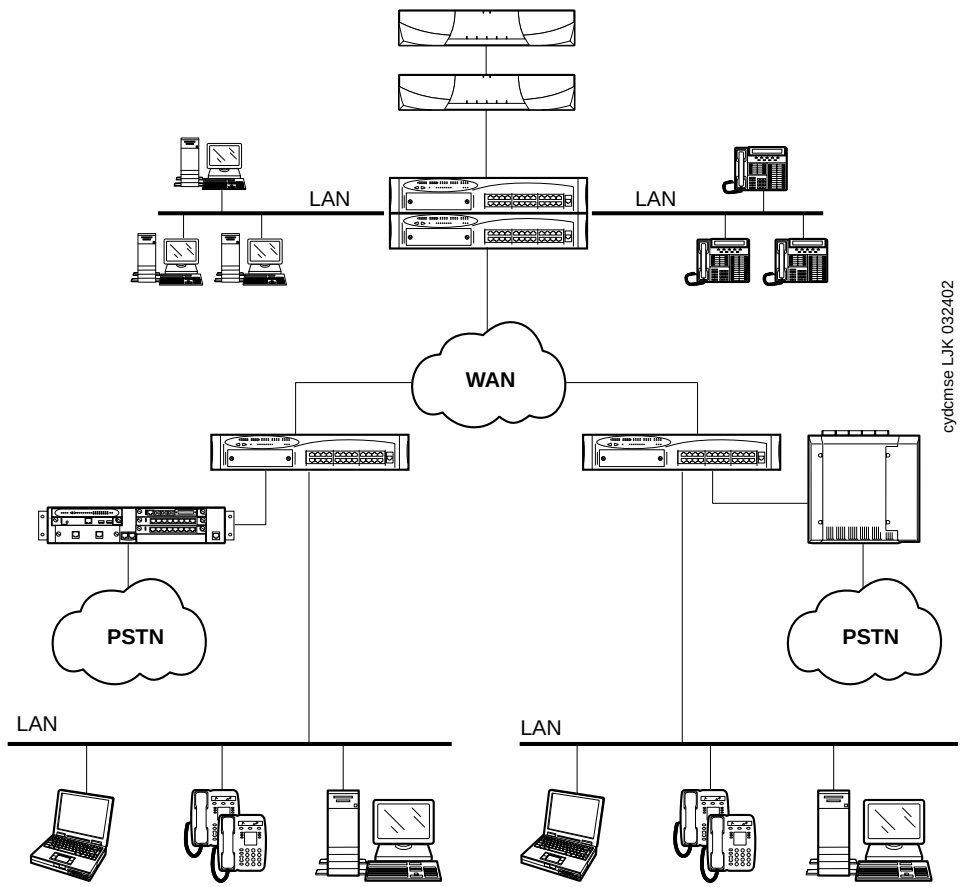
Remote Branch Office



Multi-Site Configuration

The following is an example of a multi-site customer configuration.

Multi-Site Configuration



In a multi-site environment there are a number of standalone configurations that are networked using protocols such as QSIG and DCS+. Each location has it's own processing capability and separate managed configuration. These configurations can be a mix of different solutions.

System Management

Avaya's VisAbility™ Management Suite offers a comprehensive set of web-based network and system management solutions for the converged voice and data environment. Avaya VisAbility Management provides standard based infrastructure for integrated management applications.

Avaya's VisAbility Management Suite components include:

- **Avaya MultiVantage Fault & Performance Manager** - Fault Management will display a hierarchical view of devices and their status allowing users to view and isolate alarms and errors. Performance Management provides a comprehensive set of performance reports for trending and isolation of performance issues.
- **Avaya MultiVantage Configuration Manger** - Web-based configuration and administration for multiple devices providing a single point of entry for centralized management of distributed network and campus environments.
- **Avaya Directory Enabled Management** - Provides a converged voice and data directory enabled middleware solution that provides a platform for simplifying information and configuration management of Avaya solutions, allowing a real-time, integrated, directory-based read/write access to Avaya communications server based databases
- **Avaya Terminal Configuration** - Provides a web-based client application that allows end users to access Avaya MultiVantage devices to configure some of their personal station set preferences and features. Avaya Terminal Configuration runs on top of the Avaya Directory Enabled Management software. Permission to access a limited set of features is controlled by the administrator along with the ability to print a copy of the terminal users button labels.
- **Avaya MultiService Network Manager -** Provides discovery of IP devices showing a system view on a network map and is a central launch point for VisAbility applications.
- **Avaya VoIP Monitoring Manager** - Web-based tool for assisting the data and telecommunications engineers in isolating and identifying problems with VoIP calls.
- **Avaya Policy Manager** - Policy based network management system allowing users to configure the QoS parameters for the entire distributed network from a centralized location.

Refer to the Avaya Web site for more information:

Device Management for the Server

A Web server and SNMP agent are provided for the server. Because this processing environment is the same as that being used by S8700 Media Server, the S8300 Media Server uses the same SNMP agent. The server provides a set of HTML Web pages that provide access to configuration of the Linux platform as well as telnet access to a Unix shell (bash) and some shell scripts that are used for platform management.

Avaya MultiService Network Manager Infrastructure

S8300 Media Server with the G700 Media Gateway uses the optional Avaya MultiService Network Manager infrastructure to provide access to the management facilities that are used to administer and operate the system. CVSA is a client/server component also referred to as the Avaya Management Portal (AMP), and is the entry point to the system.

The CVSA server, or back-end, component runs on a separate PC or Solaris workstation. The AMP is a Java applet based client that provides access to some basic CVSA services and serves as launching pad for a collection of network management applications.

The AMP provides the following services:

- Network viewing with standard and customizable filters
- Application launching
- Trap logging and management

Media Server Web Interface

The S8300 Media Server tasks are performed using the Media Server Web Interface. This browser-based server administration tool is an easy-to-use Graphical User Interface for performing server administration tasks such as:

- Checking server and process status
- Administering network features for the S8300 Media Server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services and installing license and authentication files.
- Installing new software and reconfiguring the server as needed
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs and running tests if needed.

The Web interface contains an extensive help system that describes all the Web screens and Media Server procedures.

Avaya VoIP Monitoring Manager

The Avaya VoIP Monitoring Manager is a QoS monitoring/feedback Windows 2000 software tool that allows the customer to visualize the real-time operation of VoIP systems. The tool provides information on QoS parameters related to VoIP quality. The Avaya VoIP Monitoring Manager provides the ability to view QoS related information through a client GUI application from the customer's LAN or via remote access. The Avaya VoIP Monitoring Manager is also capable of being configured to generate traps, associated with VoIP QoS, sent to any NMS.

The Avaya VoIP Monitoring Manager can receive RTCP packets from the following entities: IP telephones, IP soft phones, VoIP engines on media gateways and prowler boards. The RTCP data for current calls is published in the RTP MIB via the SNMP agent running on the server. The historical RTCP data is published in the VMON MIB. Both forms of RTCP data can be viewable by either field support, a system administrator or services personnel.

Adjunct Systems

Avaya provides the following adjunct systems. For a description of each, refer to the Adjunct section.

- Voice messaging and response such as INTUITY AUDIX
- System Printer (via terminal server)
- Journal Printer (via terminal server)
- Call Accounting Systems
- Call Detail Recording (CDR)

S8100 Media Server

Overview

The Avaya MultiVantage™ S8100 Media Server is not available until August, 2002. DEFINITY ONE and IP600 are currently available. For more information, refer to DEFINITY ECS System Description, 555-233-200.

Avaya MultiVantage™ S8100 Media Server with either a CMC1 or G600 Media Gateway is an all in one solution for the small, small of large, or medium-sized office. The S8100 Media Server uses WINDOWS 2000 as an operating system. Avaya MultiVantage™ Software, Avaya INTUITY™ AUDIX® Messaging Application, and Avaya Site Administration are co-resident applications on the WINDOWS 2000 platform. An optional Ethernet connection from the S8100 Media Server to the customers LAN provides easy access for administration purposes.

Detailed Description

As part of the Avaya Enterprise Class IP Solutions (ECLIPS) portfolio, the S8100 Media Server with the CMC1 or G600 Media Gateway delivers a full range of global communications capabilities including traditional voice and trunking needs as well as voice over IP technology. Growth potential of up to 450 stations and 300 trunks give the small end customer room to expand. Investment protection is secured when existing phones, cables and circuit packs can be reused in larger solutions.

Highlights of the S8100 Media Server with the CMC1 or G600 Media Gateway:

- S8100 Media Server - Windows 2000 operating system
- Co-resident applications
  - Integrated Voice Mail system - 8 ports, 100 hours of storage
  - Avaya Site Administration - Downloadable via the LAN through the Peer WEB interface this application allows for administration ease using add station wizards and a GUI interface.
  - "Avaya MultiVantage Software". For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.
  - DHCP and TFTP servers
- Customer Configurations
  - Single sites
  - Multi-site and branch offices - uses MultiVantage software allowing for consistent, common user experience across a customer's network
  - VOIP Technology- With the addition of a CLAN and IP Media Processor boards, the S8100 Media Server with the CMC1 Media Gateway provides complete IP gateway and IP gatekeeper functions to support 100% voice over IP capabilities
- Call Center
  - Up to 100 agents
  - CMS and BCMSVu
  - On-board announcement functionality - The Avaya S8100 Media Server supports 8 ports of integrated announcements and 1 hour of noncompressed speech via an on-board speech and signal processor. These announcements are stored on the S8100 Media Server hard drive and can be backed up like MultiVantage system translations files are. The processor supports (28.8 Mbytes). In addition, standard *.wav files may be imported and used for system announcements.
- Network Connectivity
  - ATM
  - IP
  - Supports Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) access and Distributed Communications System (DCS) and QSIG private networking

Required System Components

The S8100 Media Server with the CMC1 or G600 Media Gateway has the following components. A brief description of each is provided.

- TN744E Call Classifier circuit pack - integrates the tone generator, tone detection-call classifier, system clock, and synchronization functions. The tone generation function puts tones on time slots of the TDM bus.
- S8100 Media Server
    - TN2314 Processor circuit pack
    - Windows 2000 operating system
    - Occupies two slots in either the CMC1 Media Gateway or G600 Media Gateway
    - Intel Pentium III 500 MHz and Motorola processors
    - 256 MB synchronous dynamic RAM (SDRAM)
    - Services Ethernet RJ45 jack located on the faceplate to facilitate switch installation and ease of maintenance
    - RS232 port for external modem - Used for INADS access during warranty and when a maintenance contract is purchased
    - 20 GB hard drive
    - Virtual ports for INTUITY AUDIX and system announcements
    - Connectors for an optional (customer provided) keyboard, monitor and mouse
- The G600 Media Gateway has the following characteristics:
    - Up to 3 G600 Media Gateways can be connect together in one location
    - 7 available slots in the first G600 containing the S8100 Media Server and the TN 744E Tone Clock board
    - 10 slots in the second and third G600 Media Gateways
    - Can be rack or floor mounted
    - Weight: 40-50 pounds (18-22.5 kilograms)
    - Dimensions: 12 x 19 x 22 inches (30 x 48 x 55 centimeters)
- The CMC1 Media Gateway has the following characteristics:
    - Up to 3 CMC1 Media Gateways can be connect together
    - 7 slots available in the first CMC1 Media Gateway containing the S8100 Media Server and the TN 744E Tone Clock board
    - 10 slots available in the second and third CMC1 Media Gateways
    - Weight: 50 -60 lbs.
    - Dimensions: 11 x 25 x 25 inches
    - Can stand on the floor or mount on the wall
- Modem - Robotics Model 839 Sportster 33.6 Fax Modem ordered separately

Optional Components

The S8100 Media Server with either CMC1 or G600 Media Gateway can use the following components:

- Avaya TN circuit packs
- For VOIP the following packs are used:
    - C-LAN (TN799DP)
    - IP Media Processor (TN2302AP)
- Avaya Terminals
    - Analog - 6200, 7100 and 8100 series
    - Digital - 6400 and 8400 series
    - IP - 4600 series
- Avaya R300 Remote Office Communicator (Avaya R300) - acts like a simple switch at the remote site to connect remote stations and local access trunks. The Avaya R300 unit supports VOIP and DCP, as well as analog line and trunk connections. In addition, each Avaya R300 supports 12 remote dial access data channels. The S8100 Media Server with either CMC1 or G600 Media Gateway supports up to 16 Avaya R300s.
- Uninterruptible Power System (UPS) - provides uninterruptible power between the AC power source and the S8100 Media Server with either CMC1 or G600 Media Gateway. It is highly recommended that a UPS be used. It provides dependable power and in the event of a power failure, provides battery backup for a graceful shutdown.
- Ethernet Switch

Reliability

High reliability and availability is a cornerstone of MultiVantage solutions. The hardware is designed to detect and correct errors as they occur, to minimize the number of components that cause system outage, and to simplify fault isolation to a replaceable component. Error detection and correction, system reconfiguration, and alarming escalation paths provide necessary performance elements. The software is designed to recover from intermittent failures and to continue providing service with a minimum of disruption.

The maintenance subsystem manages three categories of maintenance objects: hardware maintenance objects (MO), software processes, and data relationships. Hardware MOs are tested, alarmed and removed from service by the software. When the problem is isolated, the object is replaced. If a software process encounters trouble, it is recovered or restarted. Data relationships are audited and corrected.

The maintenance philosophy is embodied in the S8100 Media Server with either CMC1 or G600 Media Gateway. New subsystems maintenance management needs were added. For example, the following design elements help ensure high availability of the Windows 2000 Server operating system:

- A secondary on-board processor complex supports initialization, monitoring, and recovery functions for all applications running on the Windows 2000 Server operating system. The secondary processor takes corrective action when problems are detected in a way to minimize user impact.
- DiskKeeper code is incorporated and runs regularly to eliminate disk fragmentation problems.
- Applications running on the operating system are thoroughly pretested to ensure proper performance. This operating system is closed to any applications other than the manufacturer-provided ones to avoid interference of operation.
- The Windows 2000 Server event log is proactively scanned for potential service affecting items. If found, alarms are generated, and, if necessary, a service technician is dispatched.

As another example of reliability, the CMC1 Media Gateway uses a three fan, hot replaceable, assembly. The fans automatically sense temperature and adjust their operating speed accordingly. If one fan fails, the other two speed up and are more than adequate to provide sufficient cooling for weeks or more. In parallel, an alarm may be created dispatching a technician to replace the fan unit.

Recoverability

The S8100 Media Server with either CMC1 or G600 Media Gateway provides the following recoverability:

- System survival of minor power disruptions without service interruption.
- Automatic restoration of the last saved version following a power outage.
- Scheduled centralized backups of critical system information at remote sites. In an emergency, multiple copies of translations, INTUITY AUDIX subscriber information, and the Windows 2000 Server registry are available. Saved information can be quickly restored. Backups can be saved on the provided PCMCIA card or to another server over the LAN.
- IP trunk fail-over to the PSTN (QOS thresholds can be set to drive shift to and from PSTN).
- Option of Emergency Transfer equipment that cuts up to 6 analog lines directly through to CO analog trunks.

Administration

Administration can be done using Avaya Site Administration, Peer WEB server or a telnet session.

Avaya Site Administration

Avaya Site Administration (ASA) is a Windows-based system management tool that provides an easy-to-use interface with the S8100 Media Server and INTUITY AUDIX. The built-in wizards globally update records, add users, and complete other administrative and maintenance tasks. Shortcuts can be created to frequently used commands and to templates for frequently used tasks. ASA uses the Graphically Enhanced DEFINITY Interface (GEDI); however, the standard system administration terminal (SAT) interface remains available through terminal emulation.

Peer WEB Server

Avaya S8100 Media Server can be administered through a web interface. Using a PEER WEB server through a LAN connection, the administrator can download software (INTUITY Message Manager, ASA), connect to INTUITY AUDIX, schedule a backup or look at backup results and restore from a backup.

Telnet Session

A terminal emulation access can be used via a Telnet session.

Avaya MultiVantage Software

The S8100 Media Server uses Avaya MultiVantage™ software for call processing solutions in both large and small customer environments. For more information on these solutions, refer to the Overview for Avaya MultiVantage™ Software.

Avaya MultiVantage is an open, scalable, highly reliable and secure telephony application. Avaya MultiVantage software provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking. MultiVantage offers over 500 features, organized in the following categories.

- Call Center
- Telephony Features
- Localization
- Collaboration
- Mobility
- Messaging
- Telecommuting
- System Management
- Reliability
- Security, Privacy and Safety
- Hospitality
- Attendant Features
- Networking
- Intelligent Call Routing
- Application Programming Interfaces.

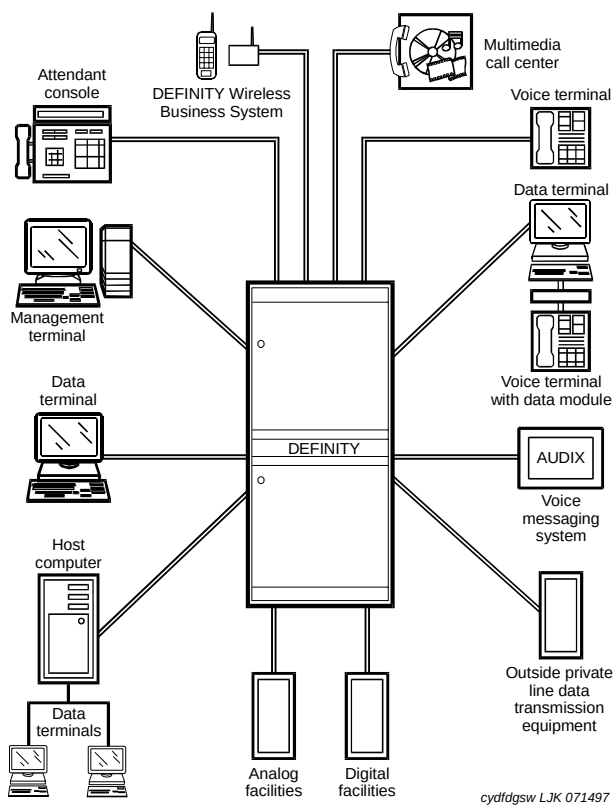
DEFINITY R

Overview

Avaya MultiVantage™ on a DEFINITY® Server R (DEFINITY R) offers an upgrade-ready processor, a high-level operating system, open interfaces, and distributed processing. It provides circuit-switched voice communications and Internet Protocol (IP) telephony including voice and data network integration, unified messaging, and multimedia conferencing and collaboration. Because it is part of Avaya's Enterprise Class IP Solutions family, it is designed to work seamlessly with other Avaya products to create a comprehensive IP communication solution.

All major hardware is housed in multi-carrier cabinets (MCC1) that hold up to five carriers or stackable single-carrier cabinets (SCC1). The solution enables high-speed connections between analog and digital trunks, data lines connected to host computers, data-entry terminals, personal computers (PC), and IP network addresses. Refer to the following diagram.

DEFINITY R



Detailed Description

DEFINITY R provides a common architecture platform across all supported line sizes. It gives large customers with multiple locations the same look and feel across their enterprise network including international locations. The configuration allows for maximizing an initial infrastructure and carrying it over into the new technologies applications.

DEFINITY R provides:

- Scalable applications for multimedia, messaging, conferencing, collaboration, call centers, mobility and distributed workforce
- Control of operational costs through the use of strong networking and management solutions
- Simple process for adding features. Add a circuit pack and use the graphical user interface (GUI) to administer it.
- Transparent multimedia calling including bandwidth on demand
- Improved use of Wide Area Network (WAN) facilities through sharing of bandwidth by voice, data, and video traffic
- Cost-effective and distributed switching arrangements in both Local Area Networks (LAN) and WANs
- Converged network environments through the use of IP and Asynchronous Transfer Mode (ATM) protocols and services
- Time Division Multiplex (TDM) transmission and switching infrastructures for both LANs and WANs. TDM supports analog, digital, and IP endpoints, both trunking and station, and an IP interface to a LAN or WAN.

Required System Components

The following are required system components:

- Avaya MultiVantage Software. For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.
- Processor Port Network (PPN) with a Switch Processing Element (SPE) and Port Network (PN)
- Control Carrier
- Cabinet
  - — MCC1 - including power supplies for AC or DC power "MCC1 Media Gateway"
  - — SCC1 - including power supplies for AC or DC power "SCC1 Media Gateway"
- Circuit Packs
  - — UN331C suffix Processor
  - — 4 TN1650 Memory

— TN1648 SYSAM

— TN1655 Packet Interface

— UN332C suffix Mass Storage System/Network Control (MSS/NET CONT)

— TN2211 Optical Drive

— TN1657 Disk Drive

— TN780, or TN2182 Tone Clock

Each of the required components is described in the paragraphs that follow.

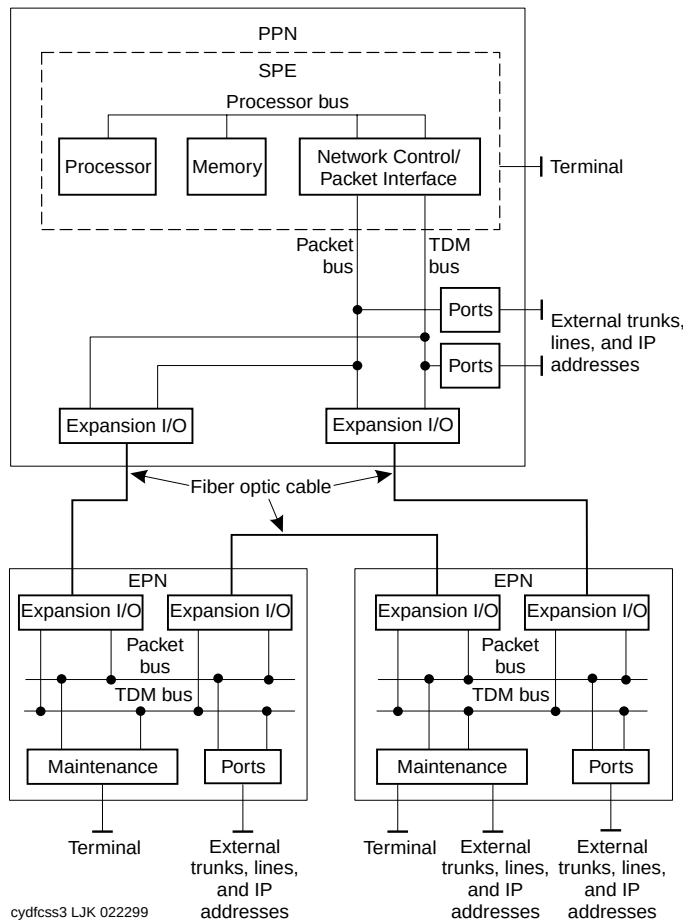
Processor Port Network and Switch Processing Element

The Processor Port Network (PPN) is a DEFINITY configuration of carriers that houses the control complex SPE of the system and port interfaces. The control complex consists of, the UN331C Processor, four TN1650 Memory packs, a TN1648 SYSAM board, a TN1655 Packet Interface, a UN332C suffix Mass Storage System/Network Control board, TN 2211 Optical Drive, TN1657 Disk Drive and a TN768, TN780 or TN2182 Tone-Clock. Additional optional circuit packs may be needed such as the UN330B for Duplication Interface in high and critical reliability systems.

When a device, such as a telephone, goes off-hook or signals call initiation, the SPE receives a signal from the port circuit connected to the device. The digits of the called number are collected and the switch is set up to make a connection between the calling and the called devices.

The following figure shows a direct-connect system with an SPE in the PPN. Buses route voice and data calls between external trunks and lines.

Components of a DEFINITY R



Port Network

The PN consists of the following components:

- *Time Division Multiplexing (TDM) bus:* Has 484 time slots, 23 B channels and 1 D channel available per bus. Runs internally throughout each PN and terminates on each end. Consists of two 8-bit parallel buses: bus A and bus B. These buses carry switched digitized voice and data signals and control signals among all port circuits and between port circuits and the SPE. The port circuits place digitized voice and data signals on a TDM bus. Bus A and bus B are normally active simultaneously.
- *Packet bus:* Runs internally throughout each PN and terminates on each end. It is an 18-bit parallel bus that carries logical links and control messages from the SPE, through port circuits, to endpoints such as terminals and adjuncts.
- The packet bus carries logical links for both on-switch and off-switch control between some specific port circuits in the system; for example, D-channels, X.25, and remote management terminals.

- *Port circuits*: Form analog/digital interfaces between the PN and external trunks and devices providing links between these devices and the TDM bus and packet bus. Incoming analog signals are converted to pulse-code modulated (PCM) digital signals and placed on the TDM bus by port circuits. Port circuits convert outgoing signals from PCM to analog for external analog devices. All port circuits connect to the TDM bus. Only specific ports connect to the packet bus.
- *Interface circuits*: Located in the PPN and in each EPN. These are types of port circuits that terminate fiber optic cables connecting TDM buses and the packet bus from the PPN cabinet to the TDM buses and packet bus of each EPN cabinet.

Carriers

Carriers hold circuit packs and connect them to power, the TDM bus, and the packet bus. There are 5 types:

- Control Carrier (PPN cabinet only)
- Optional Duplicated Control Carrier (PPN cabinet only)
- Optional Port Carrier (PPN and/or EPN cabinets)
- Optional Expansion Control Carrier (EPN cabinets only)
- Optional Switch Node Carrier (PPN and/or EPN cabinets)

Cabinets

The system cabinets house the carriers and all other components, including the power supply. A cabinet contains at least 1 carrier in an enclosed shelf with vertical slots to hold circuit packs. The circuit packs fit into connectors that attach to the rear of the slots. There are 2 cabinet types:

- Single-Carrier Cabinet (SCC1)
- Multi-Carrier Cabinet (MCC1)

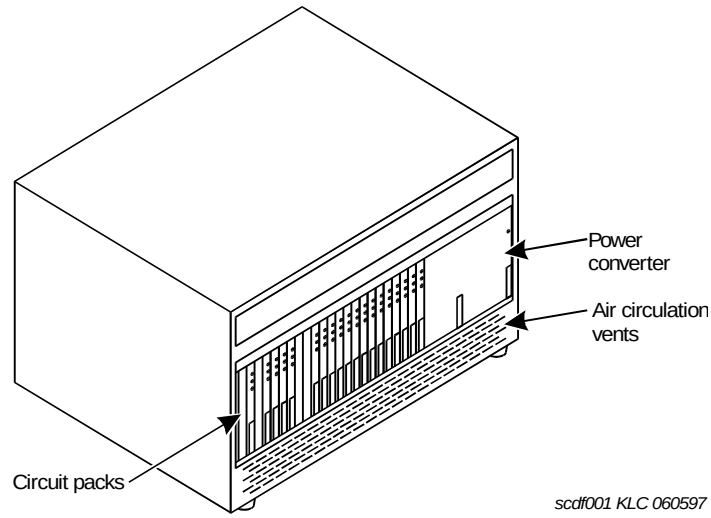
Single-Carrier Cabinets

Up to 4 Single-Carrier Cabinets (SCC1) can be stacked to form a single PN.

Single-Carrier Cabinets come in any of 4 configurations:

- An expansion control cabinet that contains additional port circuit packs, interfaces to the PPN, a maintenance interface and a power converter
- A port cabinet that contains port circuit packs and a power converter
- Refer to the following figure.

Typical Single-Carrier Cabinet (SCC1)



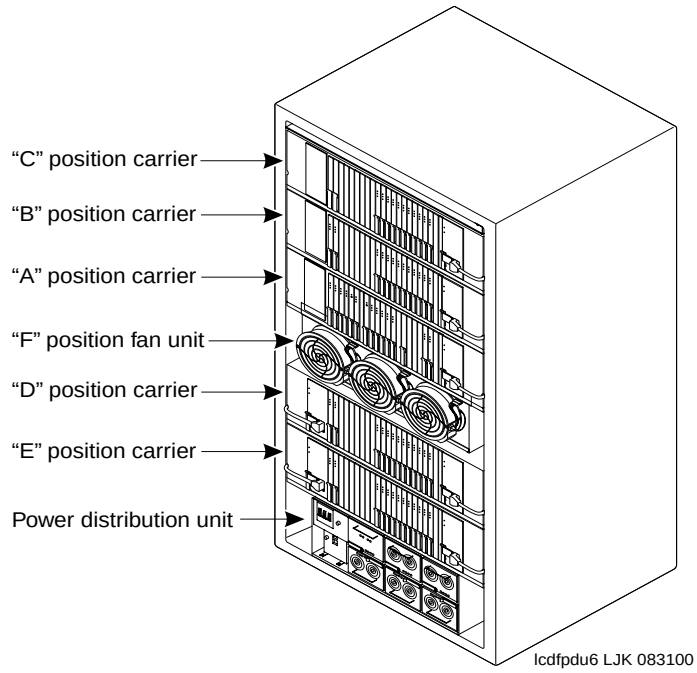
Multi-Carrier Cabinets

A Multi-Carrier Cabinet (MCC1) is a 70 in. (178 cm) cabinet that has up to 5 carriers. The 3 types of Multi-Carrier Cabinets are as follows:

- PPN cabinet contains the ports, SPE, an interface to an EPN cabinet, and/or a CSS.
- EPN cabinet contains additional ports, interfaces to the PPN and other EPN cabinets, the maintenance interface, optional interfaces to other EPN cabinets, a switch node in an SN in a CSS-connected system, or an ATM switch.
- Auxiliary cabinet contains equipment used for optional, system-related hardware, such as rack-mount equipment.

Refer to the following figure.

Typical Multi-Carrier Cabinet (MCC1)



Optional Components

Expansion Port Network

An Expansion Port Network (EPN) (optional) contains additional ports that increase the number of connections to trunks and lines. An EPN in an r model can be configured to provided service to users when the link to the main processor (fiber or T1/E1) fails or is severed, or when the processor or center stage fails. This service requires that special Standby Remote Processor equipment is colocated with the EPN to provide SPE processor capabilities to the EPN.

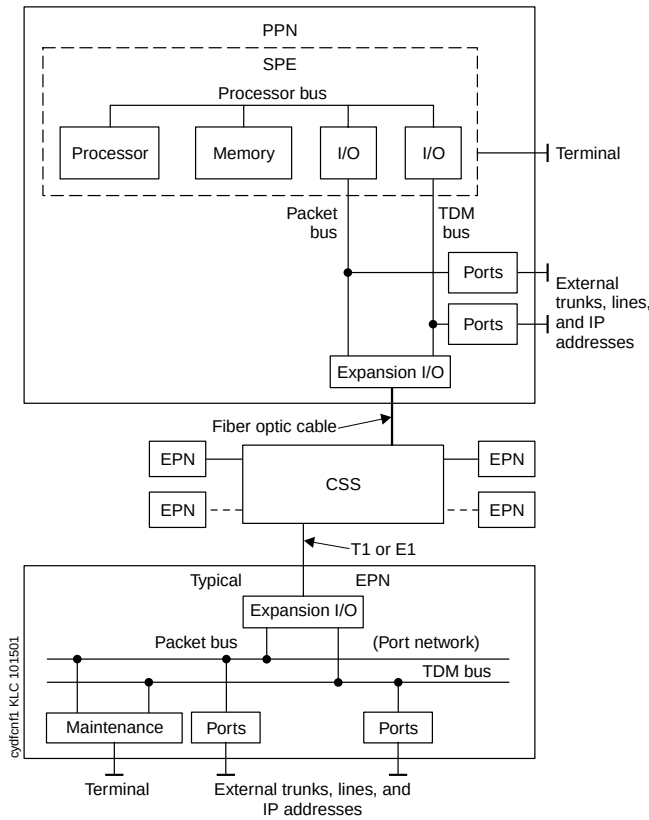
Center Stage Switch (CSS)

A CSS (optional for 3 PNs or less) in the DEFINITY R is the central interface between the PPN and the EPNs. The CSS consists of 1, 2, or 3 switch nodes (SN). One SN can expand the system from 1 EPN to up to 15 EPNs. Two SNs can expand the system to up to 29 EPNs. Three SNs can expand the system to up to 43 EPNs.

NOTE:
The number of EPNs that can be connected with 2 or 3 SNs may be less than the numbers given, depending on the internal SN-to-SN traffic.

The following figure shows a system with the added CSS to route voice and data calls between external trunks and lines.

Components of a CSS-Connected System

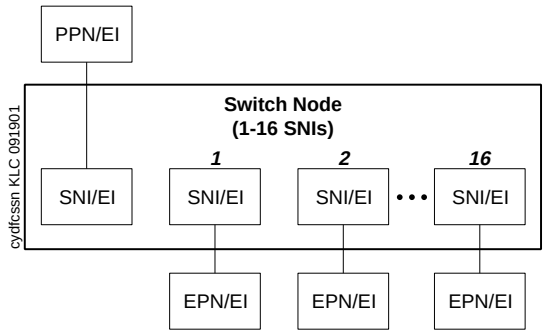


The following figure shows the CSS linking the PPN to EPNs by the SNI circuit packs in an SN carrier. An SN reduces the amount of interconnect cabling between the PPN and the EPNs by acting as a *hub* to distribute cabling.

A system using a CSS can connect from 3 to 43 PNs. The CSS can consist of up to 3 SN carriers. The CSS can also consist of 2, 4, or 6 SN carriers (duplicated SNs) in a critical-reliability system.

Each SN contains from 1 to 16 SNI circuit packs. Each interface can connect to a PN or another SN using fiber-optic cable. One interface always connects to the PPN and 1 connects to each EPN.

CSS with Switch Nodes (SNs)



In a high reliability system (with duplicated processor), 2 SNI circuit packs connect to the PPN, allowing up to 15 PNs to connect to 1 SN, up to 29 PNs to connect to 2 SNs, and up to 43 PNs to connect to 3 SNs, depending on the exact configuration chosen.

ATM Switch

The Asynchronous Transfer Mode (ATM) switch is a replacement option for the CSS. Several Avaya ATM switch types can provide switch port network connectivity. Non-Avaya ATM switches that comply with the ATM standards set by the ITU can also provide port network connectivity. In this configuration, TN2305 multi-mode or TN2306 single-mode ATM circuit packs are installed on the port networks and connected to the ATM switch with the multi- or single-mode fiber specified for the ATM switch.

NOTE:
The components of an ATM-connected system are similar to those shown in the previous figure. However, in an ATM-connected system, the CSS is replaced with an ATM switch or switches and each Expansion I/O is replaced with a TN2305 or TN2306 circuit pack.

IP Media Processor

Provides voice over internet protocol (VoIP) audio access to the switch for local stations and outside trunks. The IP Media Processor provides audio processing for between 32 and 64 voice channels and supports hairpin connections, as well as shuffling of calls between IP-IP direct connections. It can perform echo cancellation, silence suppression, fax relay service, and DTMF detection. The IP Media Processor can be updated using the firmware download feature.

IP Configurations

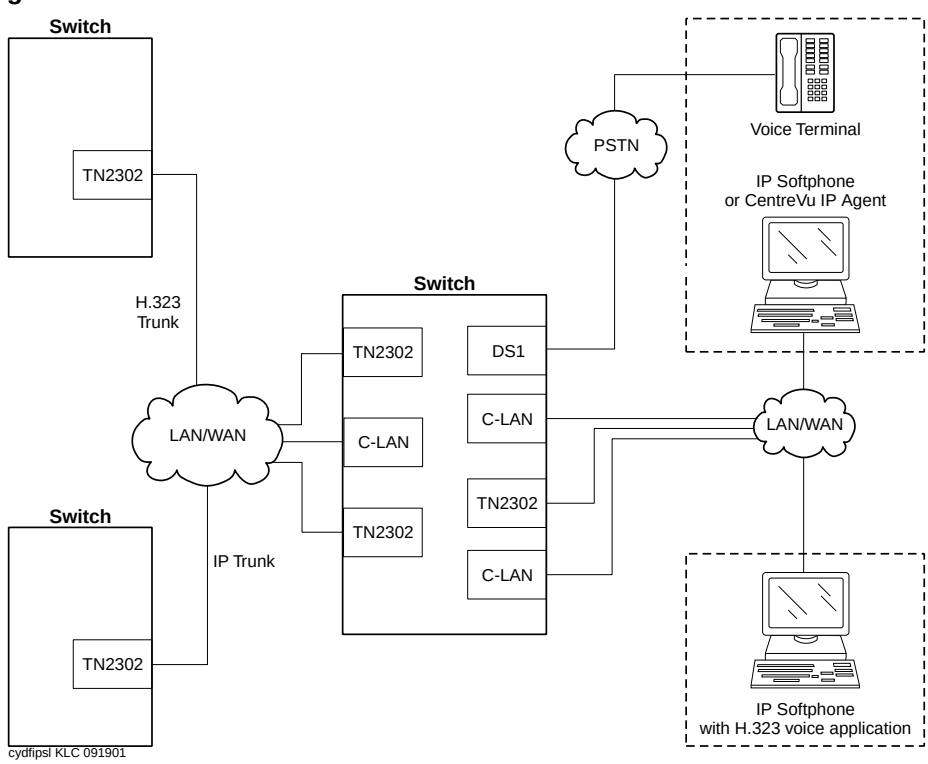
MultiVantage Solutions brings together the flexibility of IP networks with the richness of voice communication. It allows investment protection and optimization in IP, ATM, and PSTN networks. Full applications, features, and management capabilities are carried into the IP environment. Remote workers have full access to communication system features from their PCs.

MultiVantage Solutions offers featurs that enables IP administrators to specify the quality of voice communications. The Quality of Service feature enables users to administer and download the Differentiated Services Type-of-Service value to optimize voice quality. The Quality of Service feature reduces latency by implementing buffers in the audio-processing circuit pack and assists some routers in prioritizing audio traffic.

MultiVantage Solutions uses hairpin and IP-IP direct connections, two features that make voice communications more efficient. These features increase the efficiency of voice communications by reducing both per port costs and IP bandwidth usage. Hairpin connections route the voice channel connecting two IP endpoints so that the voice goes through the IP Media Processor circuit pack in IP format, thereby bypassing the TDM bus. IP-IP direct connections route the voice channel connecting two IP endpoints by sending the voice directly through the LAN or WAN between the two endpoints, instead of carrying a mixed connection of IP signaling and TDM bus signaling.

The following figure shows the trunk and line connections available with MultiVantage Solutions.

IP Configuration



As shown in the figure, MultiVantage Solutions supports connectivity for IP trunks, IP Softphones, and IP telephones.

NOTE:
The Avaya R300 Remote Office Communicator (R300) also offers full access to communication system features for remote workers.

IP is implemented using the TN2302AP, which is an IP Media Processor circuit pack inside the switch. The TN2302AP IP Media Processor provides H.323 trunk connections and H.323 voice processing for IP phones. The features that use the TN2302AP also require the TN799 C-LAN circuit pack.

NOTE:
The IP trunk used in R7 and the current TN2302AP H.323 trunks are not interoperable; that is, the TN2302AP H.323 in trunk mode cannot communicate with an R7 IP trunk. However, the TN2302AP H.323 trunk can communicate with a TN802B in MedPro mode.

Trunks

IP supports two trunk configurations:

- H.323 IP Trunk (IP Solutions mode)
- IP Trunk mode (R7)

The benefits of IP trunks include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full-function network with data and voice convergence and optimizing networks by using the available network resources.

H.323 IP Trunk (IP Solutions mode)

The IP Media Processor circuit pack (either the TN2302AP or the TN802B) supports the H.323 version 2 protocol and operates with H.323 version 2 endpoints, including stations, trunks, and gateways. An IP Media Processor circuit pack enables H.323 trunk service using IP connectivity between two DEFINITYor MultiVantage configurations. H.323 trunk groups can be configured as tie trunks supporting ISDN trunk features such as DCS+ and QSIG, or as generic tie trunks permitting interconnection with other vendors' H.323 v2-compliant switches, or as direct-inward-dial (DID) type of "public" trunks providing access to the switch for unregistered users. The TN2302AP requires the TN799 for signaling.

The TN2302AP IP Media Processor is also used for H.323 VoIP applications.

IP Trunk mode

The IP Trunk mode, using either the TN802, TN802B, or TN2302AP, will typically be chosen for interoperability with existing TN802, as opposed to the TN802B, IP Interface circuit packs.

The IP Trunk mode allows trunk groups to be defined as DS1 tie lines between DEFINITY/MultiVantage configurations over a customer's data network. Each IP Interface circuit pack in IP Trunk mode provides a basic twelve-port package that can be expanded up to a total of 30 ports.

Each TN802 or TN802B in IP Trunk mode requires a connection to a modem, an incoming line for Avaya remote access, and direct access to the NT server on the hard disk using pcANYWHERE, version 8 or later. The TN2302AP does not require connection to a modem or an incoming line. A TN802B or TN2302AP in the IP Trunk mode does not require the TN799B or later.

Avaya R300 Remote Office Communicator

The Avaya R300 offers a cost-effective method for providing full functionality at a remote site. The remote telephony available through the R300 has all the capabilities of telephony that is connected directly. Because voice and data can share the same WAN link between the DEFINITY R and the remote site, the R300 provides voice and data convergence.

The R300 terminates at the local LAN and supports local PSTN connections that use the following methods:

- 2 DS1/E1/T1/PRI
- 6 BRI ST V.35 serial port connections
- 2 analog trunks

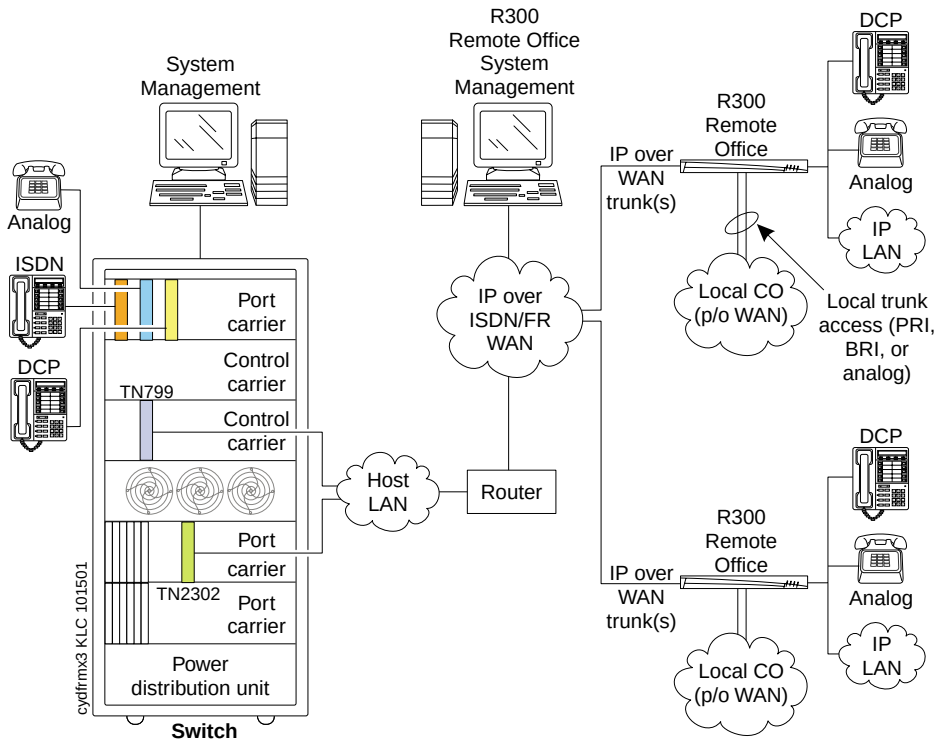
The R300 provides connections between remote stations and local access trunks. It is a rack-mounted box that features two expansion slots. One slot houses a DSP blade for VoIP support; the other slot houses a new combo blade that supports the DCP as well as the analog line and trunk connections. The R300 supports up to 24 DCP sets and 2 analog lines from a host switch. A single DEFINITY switch can support multiple R300 units, as shown in *DEFINITY Systems Capacities*, 555-233-604. This document can be viewed from the documentation Web site.

To access the documentation Web site:

1. Using a web browser, go to:
2. http://www.avaya.com/support/
3. Click Online Services
   A drop down menu is displayed.
4. Point at Support Center Applications
5. Click on Documentation.

Typically, the host DEFINITY R is linked to the R300 as shown in the following figure.

Avaya R300 Remote Office Communicator



Connectivity

The following sections describe connectivity using TCP/IP and connectivity using ATM.

Connectivity using TCP/IP

LAN Gateway

With the optional J58890MA-1List 2 LAN Gateway circuit-pack assembly installed, the switch works with PC/LAN-based communications applications that support the CallVisor Adjunct-Switch Application Interface (ASAI).

C-LAN

TCP/IP Connectivity is provided over Ethernet or Point-to-Point Protocol (PPP) to adjuncts such as CMS or INTUITY™ AUDIX®, and for DCS connectivity.

IP Asynchronous Links using C-LAN

The IP Asynchronous Links feature enables the switch to transfer existing asynchronous adjunct connectivity to an Ethernet (TCP/IP) environment. IP Asynchronous Links is a simple, session-layer, proprietary protocol that creates value for the customer in the following ways:

- Reduces the cost of connecting the switch to various adjuncts
- Allows for an open architecture to transport information and increases the speed at which data is transferred
- Allows customers to manage applications from both on-site and remote locations
- Allows for several system management applications to run on a single PC, thus reducing hardware requirements
- Provides IP Services forms to support more flexible administration
- Guarantees data delivery through a reliable session-layer protocol
- Supports customers' existing serial hardware investment through use of Network Terminal Servers

IP Asynchronous Links supports switch client applications and server applications as described in the following sections.

Switch Client Applications

Client applications with asynchronous links allow you to use TCP/IP to connect adjunct equipment to the switch via the C-LAN board.

Call detail recording (CDR) devices, property management systems (PMS) and printers can be connected using asynchronous TCP/IP links. In addition, maintenance parameters can be set to allow the switch to alarm out over a TCP/IP link.

Any device that does not support a direct TCP/IP connection, but that does support an RS232 interface, can connect to the C-LAN board through a terminal server or router.

Switch Server Applications

IP Asynchronous Links provides a telnet server to interconnect C-LAN Ethernet clients to system management applications on the switch via TCP/IP or TCP/IP and RS232 signals. IP Asynchronous Links supports the following server applications:

- System administration terminal (SAT)
- Avaya Site Administration (formerly DEFINITY Site Administration, or DSA)
- DEFINITY Network Management (DNM)
- Proxy Agent
- Enterprise Directory Gateway

Server applications send data to the switch, and the telnet server supports 80 Kbps data throughput. Current application screen interactions are supported, as are current simultaneous session limits on the switch. The telnet server satisfies all current terminal emulation modes (for example, 51x, 4410, 4425, vt220, hp262x, and pctt).

Access security for system management applications over TCP/IP is provided by the existing Access Security Gateway (ASG) feature. Through either a local or a remote node/port, users can specify the remote client IP address and port number from which the switch can accept service requests. ASG must be enabled on the system-parameters customer-options form. ASG must also be enabled for at least one customer login. The user can administer a timeout period ranging from 5 to 999 minutes, but there is currently no provision for data encryption over the LAN.

Connecting with ATM

ATM-PNC

ATM Port Network Connectivity (ATM-PNC) provides an alternative to the CSS configurations for connecting the PPN to one or more EPNs. ATM-PNC replaces the CSS in an DEFINITY R network with an ATM switch or network. ATM-PNC is available with all three reliability options—standard, high, and critical. In addition, it offers ATM-PNC duplication.

ATM-PNC integrates delivery of voice, video, and data via ATM over a converged large-bandwidth network, providing reduced infrastructure cost and improved network manageability. ATM-PNC uses standards-based open interfaces that can be provisioned with either new or existing DEFINITY or MultiVantage configurations.

ATM-CES

ATM-CES (Circuit-Emulation Service) lets the switch emulate ISDN-PRI trunks on an ATM facility. These virtual trunks can serve as integrated access, tandem, or tie trunks. ATM-CES trunk emulation maximizes port network capacities by consolidating trunking. For example, the CES interface can define up to ten virtual circuits for tie-line connectivity, consolidating onto one circuit card network connectivity that usually requires multiple circuit packs.

ATM WAN Spare Processors

An ATM WAN spare processor (WSP) provides a disaster recovery option for DEFINITY R expansion port networks deployed over an ATM WAN. An ATM WSP acts as a PPN in the event of a catastrophic failure in the network. That is, a WSP functions as a PPN if the main PPN is not functional or is not communicating to one or more of the other EPNs.

DEFINITY R is highly reliable, but when it overlays a converged network infrastructure, the switch's reliability and availability depend on the entire infrastructure, not just its own hardware and software. ATM WSPs can be placed in a switch ATM port network configuration to provide a backup arrangement of PPNs, thus maintaining the availability of the switch's features and functions.

A WSP continually monitors a connection to the main PPN to determine if the PPN is actively communicating with the EPNs. From 1 to 15 WSPs may be established in an ATM port network, and each is given a priority role to avoid its contending with other EPNs for control. Each WSP will become active if it cannot establish communications with both the main PPN and a WSP of higher priority. The WSP will wait an administered interval (from 5 to 99 minutes) and then take control within approximately 15 minutes. Although calls are not preserved while the switchover takes place, the WSP enables customers to be back in business within a reasonable amount of time.

Returning to normal operation under control of the PPN requires a manual restart sequence. When control is returned to the PPN, the WSP returns to a standby status. Calls are not preserved during the return to normal operation; you may therefore want to schedule the return to normal operation for a time when it will be least disruptive.

NOTE:
ATM WSPs cannot be used for critical reliability switches. It also cannot be used with a conventional CSS.

Reliability

Duplication is a strategy to create fully redundant systems that are highly reliable. Duplication minimizes single failure points that can interrupt call processing. Four system reliability and duplication options are available:

- Standard reliability -- does not duplicate the Tone-Clocks, the control carrier, or any inter-PN connectivity.
- High reliability -- duplicates the hardware that is associated with the SPE. The Control Carrier is duplicated, which provides duplicate SPEs and Tone-Clocks. Inter-PN connectivity and EPN Tone-Clocks are not duplicated. The strategy is to duplicate items that are associated with the SPE so that a single fault will not cause the loss of the SPE.
- Critical reliability -- requires the full duplication of the SPE, inter-PN connectivity, and the Tone-Clocks.
- ATM Network Duplication -- requires full duplication of the inter-PN connectivity and the Tone-Clocks.

As duplication increases, the maximum number of port carriers and port circuit packs per cabinet decreases.

System Capacity

The following are Busy Hour Call Completion capacities:

Type of Call	DEFINITY R
All Analog	135,000
General Business	100,000
ISDN	40,000
ACD	70,000
ICM	30,000
OCM	44,000
CTI/ASAI	70,000
Wireless	70,000
IP Telephones, Non-IP Trunks	24,000
DCP Telephones, IP Trunks	18,000
IP Telephones, IP Trunks	10,000

Adjunct Systems

Avaya provides the following adjunct systems.

- Voice messaging and response such as INTUITY AUDIX
- Call center systems such as Call Management Systems (CMS), CentreVu CMS, CentreVu Agent, CentreVu Supervisor, CentreVu Explorer
- BCMS VU Terminal and Servers
- System Printer
- Journal Printer
- Call Accounting Systems
- Call Detail Recording (CDR)
- Avaya Site Administration (ASA)
- DEFINITY Network Management (DNM)
- DEFINITY Translator ATM Manager (DTA)

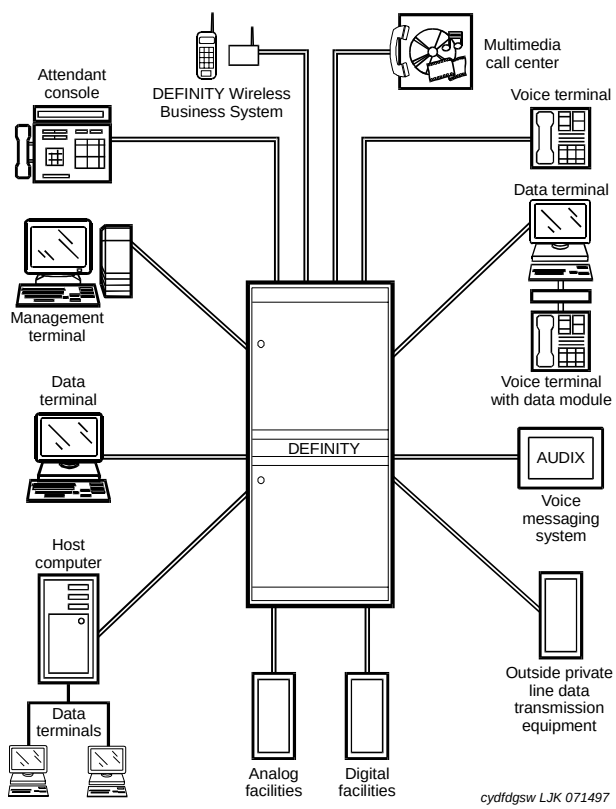
DEFINITY SI

Overview

Avaya MultiVantage™ on a DEFINITY® Server SI (DEFINITY SI) offers an upgrade-ready processor, a high-level operating system, Avaya MultiVantage Software™, open interfaces, and distributed processing. It provides circuit-switched voice communications and Internet Protocol (IP) telephony including voice and data network integration, unified messaging, and multimedia conferencing and collaboration. Because it is part of Avaya's Enterprise Class IP Solutions family, it is designed to work seamlessly with other Avaya products to create a comprehensive IP communication solution.

All major hardware is housed in multi-carrier cabinets (MCC1) that hold up to five carriers or stackable single-carrier cabinets (SCC1). The solution enables high-speed connections between analog and digital trunks, data lines connected to host computers, data-entry terminals, personal computers (PC), and IP network addresses. Refer to the following diagram.

DEFINITY SI with MultiVantage Software



Detailed Description

DEFINITY SI with MultiVantage Software provides a common architecture platform across all supported line sizes. At the heart of the DEFINITY SI is the Processor Port Network (PPN). It is the master controller of the system. The Expansion Port Network (EPN) contains line and trunk ports, and provides seamless growth capabilities. The universal port hardware allows station, trunk, and service circuit packs to be installed in any available slot.

DEFINITY SI with MultiVantage Software provides:

- Scalable applications for multimedia, messaging, conferencing, collaboration, call centers, mobility and distributed workforce
- Control of operational costs through the use of strong networking and management solutions
- Simple process for adding features. Add a circuit pack and use the graphical user interface (GUI) to administer it.
- Transparent multimedia calling including bandwidth on demand
- Improved use of Wide Area Network (WAN) facilities through sharing of bandwidth by voice, data, and video traffic
- Cost-effective and distributed switching arrangements in both Local Area Networks (LAN) and WANs
- Converged network environments through the use of IP and Asynchronous Transfer Mode (ATM) protocols and services
- Time Division Multiplex (TDM) transmission and switching infrastructures for both LANs and WANs. TDM supports analog, digital, and IP endpoints, both trunking and station, and an IP interface to a LAN or WAN.

Required System Components

The following are required system components:

- Avaya MultiVantage Software. For information about Avaya MultiVantage software, refer to the Overview for Avaya MultiVantage™ Software, 555-233-767.
- Processor Port Network (PPN) with a Switch Processing Element (SPE) and Port Network (PN)
- Control Carrier
- Cabinet
  - — SCC1 - including power supplies for AC or DC power "SCC1 Media Gateway"
  - — MCC1 - including power supplies for AC or DC power "MCC1 Media Gateway"

- Circuit Packs
  - — TN2404 Processor
  - — TN2401 Network Control/Packet Interface (NetPkt)
  - — TN780 or TN2182 Tone Clock

Each of the required components is described in the paragraphs that follow.

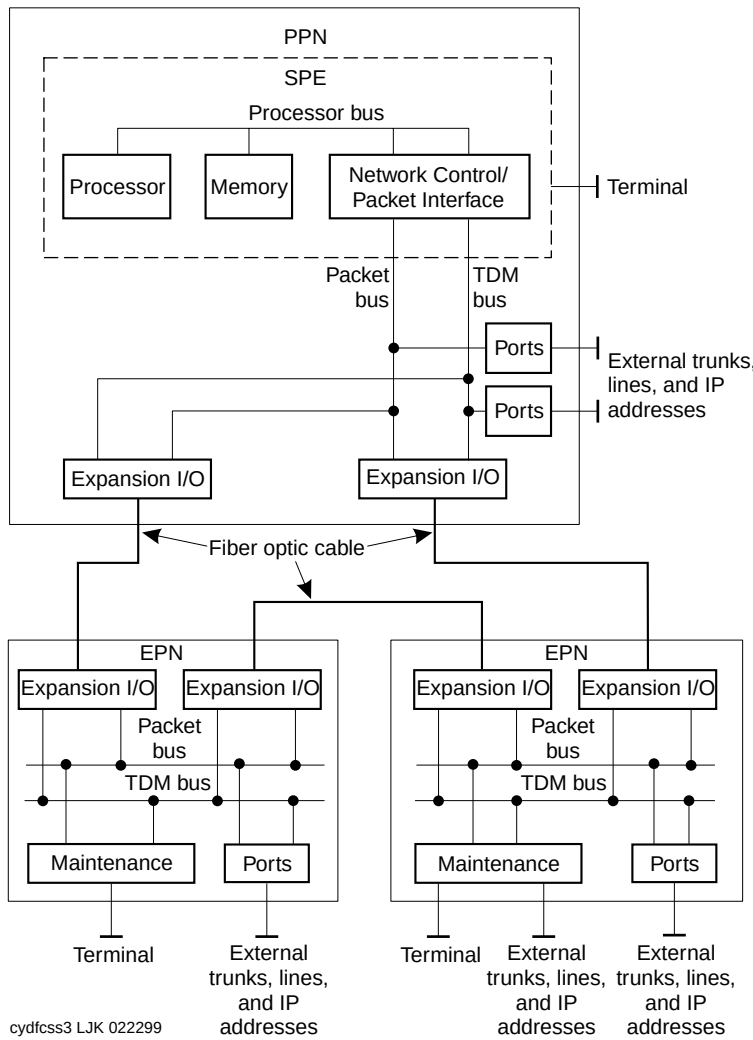
Processor Port Network and Switch Processing Element

The PPN is a DEFINITY SI configuration of carriers that houses the control complex SPE of the system and port interfaces. The control complex consists of three circuit packs, the TN2404 RISK processor, the TN2401 Network controller and the TN2182, TN780 or TN768 tone clock board. All of the control circuit packs reside in the control carrier within the PPN. Additional optional circuit packs may be needed such as the TN799DP C-LAN board which provides TCP/IP connectivity and the TN765 Processor Interface for BX.25 connectivity. For high and critical reliability systems the TN792 Duplication Interface pack is used.

When a device, such as a telephone, goes off-hook or signals call initiation, the SPE receives a signal from the port circuit connected to the device. The digits of the called number are collected and the switch is set up to make a connection between the calling and the called devices.

The following figure shows a direct-connect system with an SPE in the PPN. Buses route voice and data calls between external trunks and lines.

Components of a DEFINITYSI System



Port Network

The PN consists of the following components:

- *Time Division Multiplexing (TDM) bus:* Has 484 time slots, 23 B channels and 1 D channel available per bus. Runs internally throughout each PN and terminates on each end. Consists of two 8-bit parallel buses: bus A and bus B. These buses carry switched digitized voice and data signals and control signals among all port circuits and between port circuits and the SPE. The port circuits place digitized voice and data signals on a TDM bus. Bus A and bus B are normally active simultaneously.
- *Packet bus:* Runs internally throughout each PN and terminates on each end. It is an 18-bit parallel bus that carries logical links and control messages from the SPE, through port circuits, to endpoints such as terminals and adjuncts.

- The packet bus carries logical links for both on-switch and off-switch control between some specific port circuits in the system; for example, D-channels, X.25, and remote management terminals.
- *Port circuits*: Form analog/digital interfaces between the PN and external trunks and devices providing links between these devices and the TDM bus and packet bus. Incoming analog signals are converted to pulse-code modulated (PCM) digital signals and placed on the TDM bus by port circuits. Port circuits convert outgoing signals from PCM to analog for external analog devices. All port circuits connect to the TDM bus. Only specific ports connect to the packet bus.
- *Interface circuits*: Located in the PPN and in each EPN. These are types of port circuits that terminate fiber optic cables connecting TDM buses and the packet bus from the PPN cabinet to the TDM buses and packet bus of each EPN cabinet. These interface and cabling terminations provide a transmission path between the port circuits in different PNs.
- An Expansion Interface (EI) circuit pack also terminates each end of a cable connecting the PPN to an EPN, each end of a cable connecting an EPN to another EPN, and the PN end of a cable connected between a PN carrier and an SN carrier.
- A Switch Node Interface (SNI) circuit pack terminates the SN carrier end of a cable connected between an SN carrier and a PN.
- *DS1 converter*: Converts from a fiber interface to a DS1 interface between PNs for DS1 remoting. This reduces the maximum number of timeslots per fully-administered DS1 configuration (4 T1's) to 92.
- *Service circuits*: Connect to an external terminal to monitor, maintain, and troubleshoot the system. Also provide tone production and detection as well as call classification, recorded announcements, and speech synthesis.

Carriers

Carriers hold circuit packs and connect them to power, the TDM bus, and the packet bus. There are 5 types:

- Control Carrier (PPN cabinet only)
- Optional Duplicated Control Carrier (PPN cabinet only)
- Optional Port Carrier (PPN and/or EPN cabinets)
- Optional Expansion Control Carrier (EPN cabinets only)
- Optional Switch Node Carrier (PPN and/or EPN cabinets)

Cabinets

The system cabinets house the carriers and all other components, including the power supply. A cabinet contains at least 1 carrier in an enclosed shelf with vertical slots to hold circuit packs. The circuit packs fit into connectors that attach to the rear of the slots. There are 2 cabinet types:

- Single-Carrier Cabinet (SCC1)
- Multi-Carrier Cabinet (MCC1)

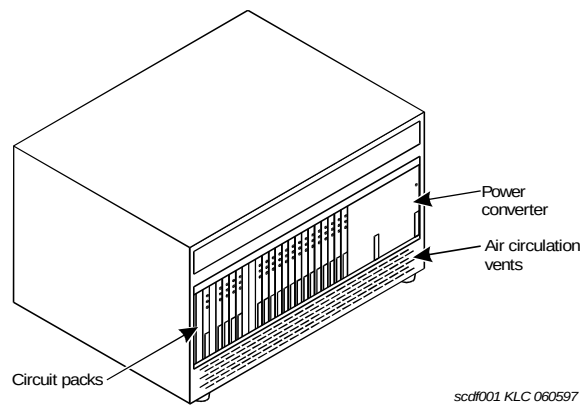
Single-Carrier Cabinets

Up to 3 Single-Carrier Cabinets (SCC1) can be stacked to form a single PN. Refer to the following figure.

Single-Carrier Cabinets come in any of 4 configurations:

- A basic control cabinet that contains a TN2404 processor, tone clock, and a power converter
- An expansion control cabinet that contains additional port circuit packs, interfaces to the PPN, a maintenance interface and a power converter
- A duplicated control cabinet that contains the same equipment as the basic control cabinet
- A port cabinet that contains port circuit packs and a power converter

Typical Single-Carrier Cabinet (SCC1)

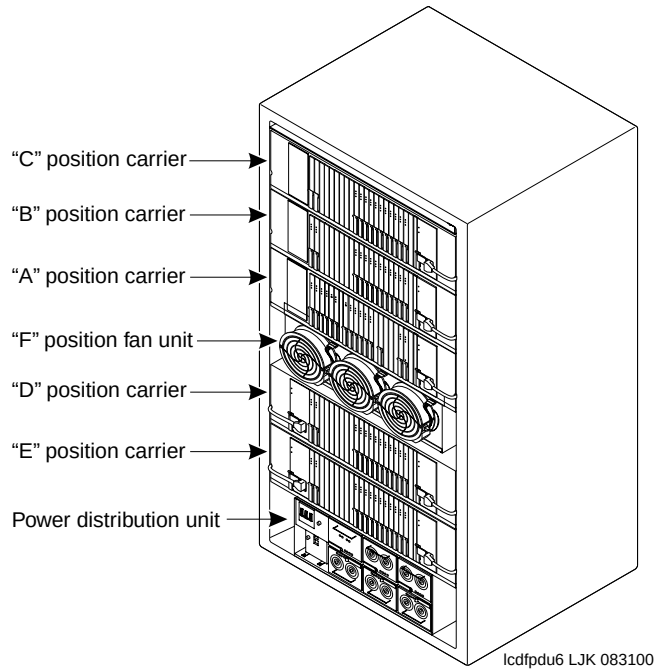


Multi-Carrier Cabinets

A Multi-Carrier Cabinet (MCC1) is a 70 in. (178 cm) cabinet that has up to 5 carriers. Refer to the following figure. The 3 types of Multi-Carrier Cabinets are as follows:

- PPN cabinet contains the ports, SPE, an interface to an EPN cabinet, and/or a CSS.
- EPN cabinet contains additional ports, interfaces to the PPN and other EPN cabinets, the maintenance interface, optional interfaces to other EPN cabinets, a switch node (in an SN in a CSS-connected system), or an ATM switch.
- Auxiliary cabinet contains equipment used for optional, system-related hardware, such as rack-mount equipment.

Typical Multi-Carrier Cabinet (MCC1)



Network Control/Packet Interface

Communicates control channel messages between the processor circuit pack and the distributed network of port circuit packs on the TDM bus. The NetPkt circuit pack (TN2401) provides eight asynchronous data channels that process and route information directly from the processor circuit pack to customer-connected equipment.

Optional Components

Expansion Port Network

An Expansion Port Network (EPN) contains additional ports that increase the number of connections to trunks and lines.

IP Media Processor

The IP Media Processor provides voice over internet protocol (VoIP) audio access to the switch for local stations and outside trunks. The IP Media Processor provides audio processing for between 32 and 64 voice channels and supports hairpin connections, as well as shuffling of calls between IP-IP direct connections. It can perform echo cancellation, silence suppression, fax relay service, and DTMF detection. The IP Media Processor can be updated using the firmware download feature.

IP Configurations

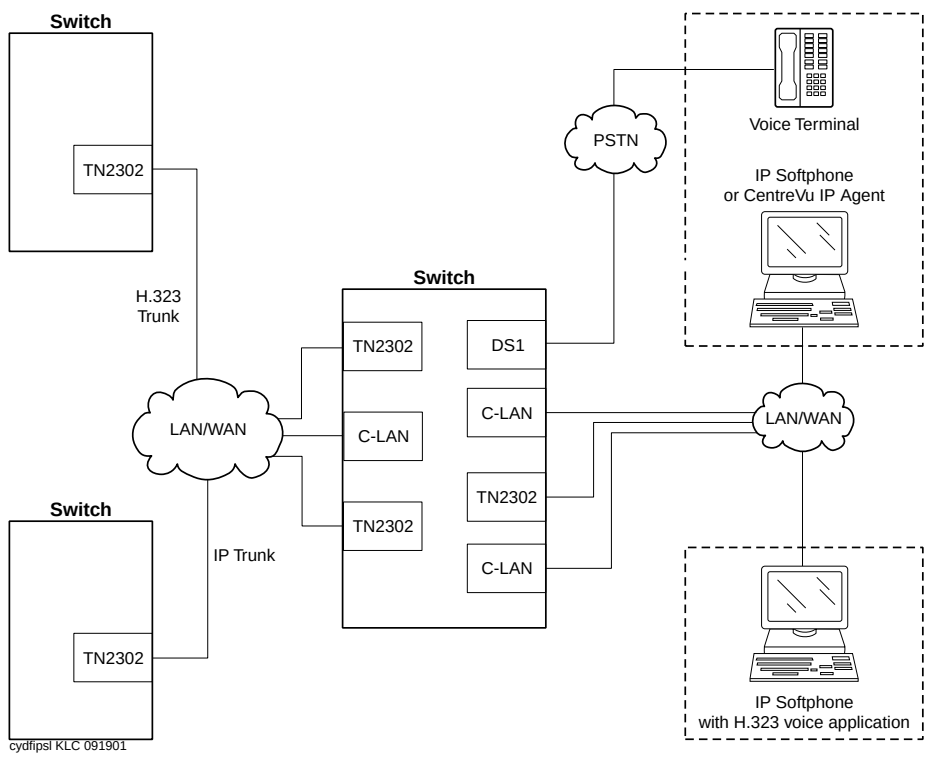
MultiVantage Solutions brings together the flexibility of IP networks with the richness of voice communication. It allows investment protection and optimization in IP and PSTN networks. Full applications, features, and management capabilities are carried into the IP environment. Remote workers have full access to communication system features from their PCs.

MultiVantage offers features that enables IP administrators to specify the quality of voice communications. The Quality of Service feature enables users to administer and download the Differentiated Services Type-of-Service value to optimize voice quality. The Quality of Service feature reduces latency by implementing buffers in the audio-processing circuit pack and assists some routers in prioritizing audio traffic.

MultiVantage uses hairpin and IP-IP direct connections, two features that make voice communications more efficient. These features increase the efficiency of voice communications by reducing both per port costs and IP bandwidth usage. Hairpin connections route the voice channel connecting two IP endpoints so that the voice goes through the IP Media Processor circuit pack in IP format, thereby bypassing the TDM bus. IP-IP direct connections route the voice channel connecting two IP endpoints by sending the voice directly through the LAN or WAN between the two endpoints, instead of carrying a mixed connection of IP signaling and TDM bus signaling.

The following figure shows the trunk and line connections available with MultiVantage.

IP Configuration



As shown in the figure, MultiVantage supports connectivity for IP trunks, IP Softphones, and IP telephones.

IP is implemented using the TN2302AP, which is an IP Media Processor circuit pack inside the switch. The TN2302AP IP Media Processor provides H.323 trunk connections and H.323 voice processing for IP phones. The features that use the TN2302AP also require the TN799 C-LAN circuit pack.

NOTE:
The IP trunk used in R7 and the current TN2302AP H.323 trunks are not interoperable; that is, the TN2302AP H.323 in trunk mode cannot communicate with an R7 IP trunk. However, the TN2302AP H.323 trunk can communicate with a TN802B in MedPro mode.

Trunks

IP supports two trunk configurations:

- H.323 IP Trunk (IP Solutions mode)
- IP Trunk mode

The benefits of IP trunks include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full-function network with data and voice convergence and optimizing networks by using the available network resources.

H.323 IP Trunk (IP Solutions mode)

The IP Media Processor circuit pack (either the TN2302AP or the TN802B) supports the H.323 version 2 protocol and operates with H.323 version 2 endpoints, including stations, trunks, and gateways. An IP Media Processor circuit pack enables H.323 trunk service using IP connectivity between two DEFINITY systems. H.323 trunk groups can be configured as DEFINITY-specific tie trunks supporting ISDN trunk features such as DCS+ and QSIG, or as generic tie trunks permitting interconnection with other vendors' H.323 v2-compliant switches, or as direct-inward-dial (DID) type of "public" trunks providing access to the switch for unregistered users. The TN2302AP requires the TN799 for signaling.

The TN2302AP IP Media Processor is also used for H.323 VoIP applications.

IP Trunk mode

The IP Trunk mode (using either the TN802, TN802B, or TN2302AP) will typically be chosen for interoperability with existing TN802 (as opposed to the TN802B) IP Interface circuit packs.

The IP Trunk mode allows trunk groups to be defined as DS1 tie lines between DEFINITY systems over a customer's data network. Each IP Interface circuit pack in IP Trunk mode provides a basic twelve-port package that can be expanded up to a total of 30 ports.

Each TN802 or TN802B in IP Trunk mode requires a connection to a modem, an incoming line for Avaya remote access, and direct access to the NT server on the hard disk using pcANYWHERE, version 8 or later. The TN2302AP does not require connection to a modem, an incoming line, or access using pcANYWHERE. A TN802B or TN2302AP in the IP Trunk mode does not require the TN799B or later.

Avaya R300 Remote Office Communicator

The Avaya R300 offers a cost-effective method for providing full functionality at a remote site. The remote telephony available through the R300 has all the capabilities of telephony that is connected directly. Because voice and data can share the same WAN link between the DEFINITY SI switch and the remote site, the R300 provides voice and data convergence.

The R300 terminates at the local LAN and supports local PSTN connections that use the following methods:

- 2 DS1/E1/T1/PRI
- 6 BRI ST V.35 serial port connections
- 2 analog trunks

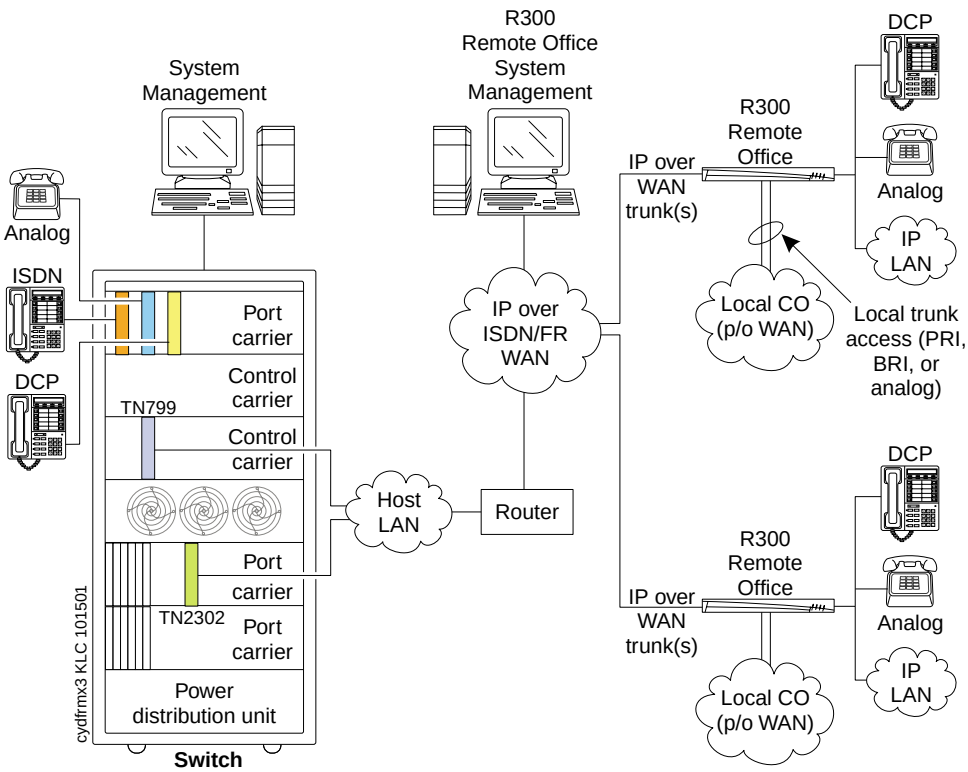
The R300 provides connections between remote stations and local access trunks. It is a rack-mounted box that features two expansion slots. One slot houses a DSP blade for VoIP support; the other slot houses a new combo blade that supports the DCP as well as the analog line and trunk connections. The R300 supports up to 24 DCP sets and 2 analog lines from a host switch. A single DEFINITY switch can support multiple R300 units, as shown in *DEFINITY Systems Capacities*, 555-233-604. This document can be viewed from the documentation Web site.

To access the documentation Web site:

6. Using a web browser, go to:
7. http://www.avaya.com/support/
8. Click Online Services
   A drop down menu is displayed.
9. Point at Support Center Applications
10. Click on Documentation.

Typically, the host DEFINITY SI is linked to the R300 as shown in the following figure.

Avaya R300 Remote Office Communicator



Connectivity

The following sections describe connectivity using TCP/IP.

Connecting with TCP/IP

LAN Gateway

With the optional J58890MA-1List 2 LAN Gateway circuit-pack assembly installed, the switch works with PC/LAN-based communications applications that support the CallVisor Adjunct-Switch Application Interface (ASAI).

C-LAN

TCP/IP Connectivity is provided over Ethernet or Point-to-Point Protocol (PPP) to adjuncts such as CMS or INTUITY™ AUDIX®, and for DCS connectivity. The C-LAN circuit pack (TN799) provides a “bridge” from the TDM bus to the packet bus on DEFINITY CSI.

IP Asynchronous Links using C-LAN

The IP Asynchronous Links feature enables the switch to transfer existing asynchronous adjunct connectivity to an Ethernet (TCP/IP) environment. IP Asynchronous Links is a simple, session-layer, proprietary protocol that creates value for the customer in the following ways:

- Reduces the cost of connecting the switch to various adjuncts
- Allows for an open architecture to transport information and increases the speed at which data is transferred
- Allows customers to manage applications from both on-site and remote locations
- Allows for several system management applications to run on a single PC, thus reducing hardware requirements
- Provides “IP Services” forms to support more flexible administration
- Guarantees data delivery through a reliable session-layer protocol
- Supports customers’ existing serial hardware investment through use of Network Terminal Servers

IP Asynchronous Links supports switch client applications and server applications as described in the following sections.

Switch Client Applications

Client applications with asynchronous links allow you to use TCP/IP to connect adjunct equipment to the switch via the C-LAN board.

Call detail recording (CDR) devices, property management systems (PMS) and printers can be connected using asynchronous TCP/IP links. In addition, maintenance parameters can be set to allow the switch to alarm out over a TCP/IP link.

Any device that does not support a direct TCP/IP connection, but that does support an RS232 interface, can connect to the C-LAN board through a terminal server or router.

Switch Server Applications

IP Asynchronous Links provides a telnet server to interconnect C-LAN Ethernet clients to system management applications on the switch via TCP/IP or TCP/IP and RS232 signals. IP Asynchronous Links supports the following server applications:

- System administration terminal (SAT)
- Avaya Site Administration (formerly DEFINITY Site Administration, or DSA)
- DEFINITY Network Management (DNM)
- Proxy Agent
- Enterprise Directory Gateway

Server applications send data to the switch, and the telnet server supports 80 Kbps data throughput. Current application screen interactions are supported, as are current simultaneous session limits on the switch. The telnet server satisfies all current terminal emulation modes (for example, 51x, 4410, 4425, vt220, hp262x, and pctt).

Access security for system management applications over TCP/IP is provided by the existing Access Security Gateway (ASG) feature. Through either a local or a remote node/port, users can specify the remote client IP address and port number from which the switch can accept service requests. ASG must be enabled on the system-parameters customer-options form. ASG must also be enabled for at least one customer login. The user can administer a timeout period ranging from 5 to 999 minutes, but there is currently no provision for data encryption over the LAN.

Reliability

Duplication is a strategy to create fully redundant systems that are highly reliable. Duplication minimizes single failure points that can interrupt call processing. Three system reliability and duplication options are available:

- Standard reliability--does not duplicate the Tone-Clocks, the control carrier, or any inter-PN connectivity.
- High reliability--duplicates the hardware that is associated with the SPE. The Control Carrier is duplicated, which provides duplicate SPEs and Tone-Clocks. Inter-PN connectivity and EPN Tone-Clocks are not duplicated. The strategy is to duplicate items that are associated with the SPE so that a single fault will not cause the loss of the SPE.
- Critical reliability -- requires the full duplication of the SPE, inter-PN connectivity, and the Tone-Clocks.

As duplication increases, the maximum number of port carriers and port circuit packs per cabinet decreases.

System Capacity

The following are Busy Hour Call Completion capacities:

Type of Call	G3si
All Analog	20,000
General Business	20,000
ISDN	20,000
ACD	20,000
ICM	20,000
OCM	20,000
CTI/ASAI	20,000
Wireless	20,000
IP Telephones, Non-IP Trunks	18,000
DCP Telephones, IP Trunks	13,500
IP Telephones, IP Trunks	7,500

Adjunct Systems

Avaya provides the following adjunct systems.

- Voice messaging and response such as INTUITY AUDIX
- Call center systems such as Call Management Systems (CMS), CentreVu CMS, CentreVu Agent, CentreVu Supervisor, CentreVu Explorer
- BCMS VU Terminal and Servers
- System Printer
- Journal Printer
- Call Accounting Systems
- Call Detail Recording (CDR)
- Avaya Site Administration (ASA)
- DEFINITY Network Management (DNM)
- DEFINITY Translator ATM Manager (DTA)

DEFINITY CSI

Overview

Avaya MultiVantage™ on a DEFINITY® Server CSI (DEFINITY CSI) is medium office solution from 50 to 900 stations. This solution uses DEFINITY TN circuit packs, Avaya MultiVantage Software, and the CMC1 Media Gateway. DEFINITY CSI has the ability to easily and cost-effectively migrate to a larger system as a mid-sized company expands. All of the DEFINITY CSI applications and almost all of the hardware can still be used, protecting the initial investment.

Depending on business needs, DEFINITY CSI can cover a single site or be networked in multiple locations. For example, DEFINITY CSI might provide a solution for a satellite office within a larger business or a branch locations around the world. Multisite companies can centrally maintain DEFINITY CSIs via remote diagnostics and alarming and can centrally administer the system using a system administration tool.

Detailed Description

Highlights of DEFINITY CSI are:

- Ideal for small single sites, multi-sites, and branch locations
- Supports up to 1300 total ports
- Supports 400 trunks and 900 stations
- Provides MultiVantage Software including switching, messaging, and administration
- Same administration over a customer's network
- Same user interface over a customer network

Required System Components

DEFINITY CSI consists of the following required components:

- TN2402 Processor Board
- TN2182 Tone Clock
- CMC1 Media Gateway

Refer to the following sections for a description of each component.

TN2402 Processor Board

The TN2402 Processor Board resides in slot one of cabinet A. This board contains 32 Mb of DRAM memory and 32 Mb of flash memory. The software is stored on flash memory and removable memory through Flash-ROM that plugs directly into the TN2402. The memory on the Processor circuit pack contains the generic program and the system translations. The cartridge contains a copy of the system translations and error log. Additional features include:

- Provides the 5-volt ATA PC-card Memory Card Interface.
- Uses an external modem to dial out alarms.
- Three external RS232 interfaces [CD1]
- SAT terminal interface
- SMDR/Printer or other DTE interface
- Connection for external Modem

Tone Clock (TN2182)

The TN2182B tone-clock integrates the tone generator, tone detection-call classifier, system clock, and synchronization functions onto one circuit pack for all system reliability configurations. The TN2182 supports 8 ports for tone detection and allows gain or loss applied to PCM signals received from the bus.

CMC1 Media Gateway

The CMC1 Media Gateway has the following characteristics:

- Size is 25.5 inches wide, 24.5 inches high, and 11.3 inches deep. It is designed for wall mounting. Where required the first cabinet can be floor or table mounted.
- Ten Universal port slots plus one power supply slot per CMC1.
- Standard reliability only.
- Is limited to one port network made up of a maximum of three CMC1s. A port network consists of a control CMC1 which is designated "A". The second and third CMC1s are optional and are designated cabinets "B" and "C" respectively. Cabinet address ID within the port network is set via a DIP switch on the backplane.
- Circuit packs are inserted and removed from the left side; cabinet I/O is from the right side.
- The CMC1s in a port network are interconnected via shielded TDM/LAN bus cables.
- The CMC1 is AC powered only, there are no internal batteries, and no DC power option.
- CMC1 Cooling

  Two 12-volt DC variable speed fans integrated into the bottom of the cabinet provides CMC1 cooling. The fans force air through an air filter and up through the cabinet. Air exits out the back of the cabinet. Speed control is provided via the 650A Global Power supply which varies the fan input voltage between 8VDC and 14VDC depending on a temperature sensor mounted in the power supply. The fan assembly includes the two fans, a frame to which they are attached, wiring, and a connector that plugs into the backplane. The assembly is easily installed and removed. The entire assembly will be replaced as a unit if a fan fails.

  A fan failure will result in the following:

  — An alarm through the power supply that will appear to the system

  — The remaining fan will go to high speed

  — The red LED on the 650A Global Power supply faceplate will light

DEFINITY CSI Reliability and Recovery

DEFINITY CSI provides the following capabilities:

- DEFINITY CSI can survive minor power surges (including lightning-induced surges up to 2500 Volts peak) without service interruption. Surge protectors can be purchased for increased coverage
- DEFINITY CSI can be placed in less-than-ideal locations since it can handle above-average temperatures and humidity.
- In case of power outages, DEFINITY CSI automatically restores the last saved version of user translations and runs them at system restart.
- DEFINITY CSI supports the remote diagnostics capability, which enables quick troubleshooting and maintenance.
- DEFINITY CSI conducts self-diagnostics and can self correct a large portion of system errors. If further technical assistance is required, DEFINITY CSI calls out to support using the external modem.
- DEFINITY CSI conducts standard maintenance routines automatically.
- By default DEFINITY CSI backs up all the user translations every day at midnight.
- DEFINITY CSI has 99.9% reliability due to the single processor configuration

DEFINITY CSI System Capacity

Type	Capacity
Maximum Trunks	400
Maximum Stations	900
Maximum Ports	1300 (limited by slots, not software
Maximum IP Endpoints	390

Avaya MultiVantage Software

DEFINITY CSI uses Avaya MultiVantage™ software for call processing solutions in both large and small customer environments. For more information on these solutions, refer to the Overview for Avaya MultiVantage™ Software.

Avaya MultiVantage is an open, scalable, highly reliable and secure telephony application. Avaya MultiVantage software provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking. MultiVantage offers over 500 features, organized in the following categories.

- Call Center
- Telephony Features
- Localization
- Collaboration
- Mobility
- Messaging
- Telecommuting
- System Management
- Reliability
- Security, Privacy and Safety
- Hospitality
- Attendant Features
- Networking
- Intelligent Call Routing
- Application Programming Interfaces

s

Required Components

Avaya MultiVantage Software

The S8700 IP Connect uses Avaya MultiVantage™ software for call processing solutions in both large and small customer environments. For more information on these solutions, refer to the Overview for Avaya MultiVantage™ Software.

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- System Management
- Reliability
- Security, Privacy and Safety
- Hospitality
- Attendant Features
- Networking
- Intelligent Call Routing
- Application Programming Interfaces

S8700 Media Server for Multi-Connect

Characteristics of the S8700 Media Server for Multi-Connect Configurations include:

- Sufficient 10/100 Ethernet ports to support IPSI network control links 1 & 2, services access, duplication, and administration/alarming.
- IDE hard disk
- IDE CD ROM
- 100 meters distance limitation between S8700 Media Servers
- Support for global power
- Storage media for OS, customer translations, and maintenance software
- Supports USB port connectivity for modem
- 128 MB Flash Card for removable media
- Provides monitoring and alarming
- Supports remote call out alarming from either server
- SNMP alarming

The S8700 Media Server must be mounted in a 19 inch rack.

S8700 Media Server for IP Connect

Characteristics of the S8700 Media Server for IP Connect Configurations include:

- Sufficient 10/100 Ethernet ports to support IPSI network control links 1 & 2, services access, duplication, and administration/alarming.
- IDE hard disk
- IDE CD ROM
- 100 meters distance limitation between S8700 Media Servers
- Support for global power
- Storage media for OS, customer translations, and maintenance software
- Supports USB port connectivity for modem
- 128 MB Flash Card for removable media
- Provides monitoring and alarming
- Supports remote call out alarming from either server
- SNMP alarming

The S8700 Media Server must be mounted in a 19 inch rack.

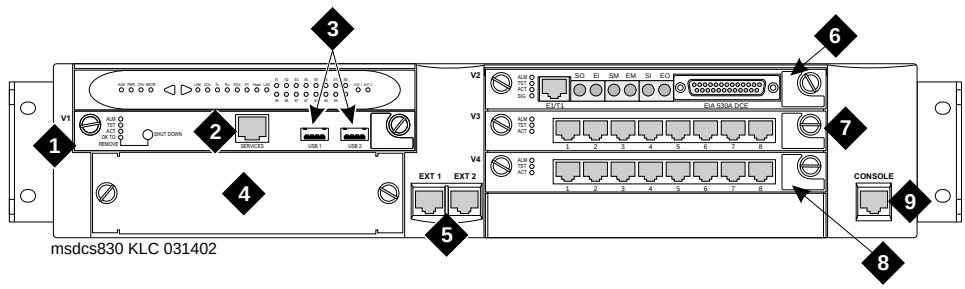
S8300 Media Server

The S8300 Media Server is a Pentium-based processor that runs on a Linux operating system. It is placed in slot V1 in the G700 Media Gateway. When installing or removing the S8300 Media Server, the S8300 Media Server and LED board must be removed or installed as a single unit.

The S8300 Media Server provides MultiVantage Software that is derived from DEFINITY® functionality. It provides a feature-rich enviornment. The S8300 Media Server does not support port networks or extended port networks.

The S8300 Media Server is **NOT** hot swappable. The following figure shows the S8300 Media Server in a G700 Media Gateway.

s8300 Media Server



1 Media Module Slot V1. In this case, it contains the S8300 Media Server

2 An S8300 Media Server or an S8300 Media Server in a Local Survivable Processor (LSP) configuration can only be used in the Media Module slot labeled V1.

3 Service Ports

4 USB Ports

5 Cajun Expansion Module Slot

6 Dual 10/100Base-T Ethernet Switch Ports

7 Media Module Slot V2

8 Media Module Slot V3

9 Media Module Slot V4

10 Console (Serves as Serial On-Site Administration Interface)

The S8300 Media Server comes standard with:

- Avaya MultiVantage Software
- Administration and maintenance provisioning software
- 20G Hard drive
- 256 MB RAM
- Web server
- Linux OS (Redhat v6.2)
- H.248 Gateway Controller (Megaco)
- H.323 Gate Keeper
- TFTP server

IDE Interfaces

The S8300 Media Server design supports IDE interfaces. These interfaces support PIO IDE transfers up to 14 MBs or Bus Master IDE transfer up to 33MBs. Ultra DMA/33 Synchronous Mode Transfers are also supported.

The S8300 Media Server's basic configuration does not support removable media for backup storage. The primary method of backup for the S8300 Media Server's configuration and application data is via a LAN-connected storage device. The address of this storage directory is maintained in flash memory.

S8300 Media Server Module Interfaces

There is no external connector support for VGA video or PS2 keyboard and mouse. The interfaces to the unit are through network connections, and all customer and services access to the S8300 Media Server is via the network connections. There are USB connectors for optional customer connectivity such as the service's modem for remote access to the S8300 Media Server as well as an Ethernet connector for service's access on the faceplate.

Backplane Interfaces

All backplane interfaces reside on one connector. They include:

- The DEFINITY TDM bus
- ENET packet interface
- All power leads
- A 10/100BaseTX Ethernet twisted pair interface
- Power and Ground
- Various other signals

Faceplate Interfaces

The following interfaces are accessible to Avaya personnel and customers via the faceplate:

- A 10/100BaseT Ethernet interface provided via an RJ45 filtered connector
- Two USB interfaces using standard USB connectors
- A shutdown switch
- LEDs for status reporting

Operating System

The hardware platform is compatible with the Red Hat Linux 6.2 or later operating systems.

IP Server Interface for S8700 Multi-Connect

In the S8700 Multi-Connect the bearer and control paths are separate. Control information for port networks travels over a dedicated LAN through the Ethernet switch and terminates on the S8700 Media Server at one end and a TN2312AP IP Server Interface (IPSI) on the other.

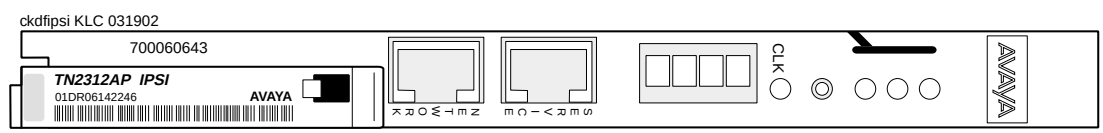
Many of the services required for a port network are available on the IPSI board such as tone generation, tone detection, call classification equivalent to a TN2182C, and clock generation. Each IPSI will have one 10/100BaseT Ethernet connection to the Ethernet switch. In the S8700 Multi-Connect, each IPSI normally controls five port networks by tunneling control messages over the bearer network to port networks without IPSIs. Further, IPSIs cannot be placed in a port network that has either a Stratum 3 clock or DS1-Converter.

Characteristics of the IP Server Interface:

- Always resides in the tone/clock slot
- Has 10/100 BaseT interface for connection to server
- Has 10/100 BaseT interface for connection to services laptop
- Supports 8 global Call Classification ports
- Supports network diagnostic capabilities
- Provides PN clock generation and synchronization (Stratum 4 type II only)
- Provides PN tone generation
- Provides PN tone detection/global call classifier/international protocols
- Provides distributed PN packet interface
- Supports IPSI firmware download
- One IPSI per configuration provides serial number support for License File Feature Activation

The following diagram is of the IPSI faceplate.

IPSI Faceplate



The IPSI connects the S8700 Media Server to the PN and carries the control network signals to the other boards in the PN. It provides network packet, tone clock, and tone detection and receiving functionality.

IPSI IP addresses are typically assigned automatically using DHCP service. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance and firmware downloads

To determine the number of IPSI-connected port networks recommended to support a Multi-Connect configuration, use the following formula:

([number of IPSI connected PNs] divided by 5) + 1

For example: You have 20 PNs, so you need 20/5 = 4. Always round up! In this example, you need 4 + 1 or a minimum of 5 IPSIs to support the 20 PNs. There is an absolute minimum of 2 IPSIs per configuration.

NOTE:
The addition of 1 IPSI provides additional fault tolerance by adding excess IPSI capacity.

The following table shows some results of the calculation:

Total Number of Port Networks	Number of IPSI-Connected Port Network
1	1
2 to 5	2
6 to 10	3
11 to 15	4
16 to 20	5

IP Server Interface (TN2312AP) for S8700 IP Connect

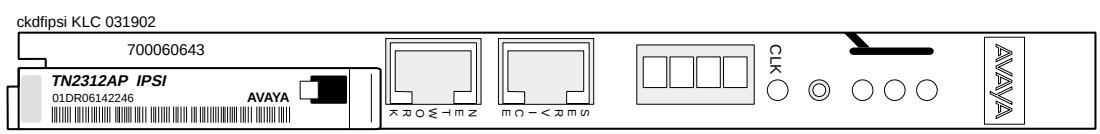
The IP Server Interface (IPSI) provides transport of CCMS messages over IP allowing the S8700 Media Server to communicate with the PNs. The IPSI is required to provide control network signaling over the customer's LAN and WAN. Tone generation, tone detection, global call classification, as well as stratum 4 type clock generation are provided on the IPSI board.

The IPSI has the following characteristics:

- Always resides in the tone/clock slot II
- Has 10/100 BaseT interface for connection to server
- Has 10/100 BaseT interface for connection to services laptop
- Supports 8 global Call Classification ports
- Supports network diagnostic capabilities
- Provides PN clock generation and synchronization (Stratum 4 type II only)
- Provides PN tone generation
- Provides PN tone detection/global call classifier/international protocols
- Provides distributed PN packet interface
- Supports IPSI firmware download
- One IPSI per configuration provides serial number support for License File Feature Activation

The IPSI has one 10/100BaseT Ethernet connection on the faceplate to connect to a laptop computer via telnet for service access. This interface is used to set and reset the IP address of the IPSI board, the sub-net mask, and the gateway address. It can be used for network debugging, such as ping, traceroute, and netstat. It can also be used to access the server. It is important to note that any computer connected to this port is expected to have its IP address to be 192.11.13.6 and subnet mask 255.255.255.252

IPSI Faceplate



Avaya Cajun P330

The Avaya P330 family of stackable Ethernet workgroup switches includes a range of modules with 10/100/1000 Mbps ports and a Layer 3 capability/ATM Expansion sub-module. The Avaya P333T switch has 24x10/100 Mbps ports and an Expansion sub-module slot. The Avaya P334T switch has 48x10/100 Mbps ports and an Expansion sub-module slot. The optional expansion sub-modules provide additional Ethernet, Fast Ethernet, and Gigabit Ethernet connectivity.

An Avaya P330 stack can contain up to 10 switches and up to 3 backup power supply units. The stacked switches are connected using the Avaya X330STK stacking sub-modules which plug into a slot in the back of the Avaya P330. They are connected using the X330SC or X330LC cable (if the stack is split between 2 racks). The Avaya X330RC cable connects the top and bottom switches in the stack and provides redundancy and hot-swappability in the same way that modules can be swapped in a modular switching chassis.

The Avaya P330 is fully compliant with IEEE standards for VLAN Tagging, Gigabit Ethernet, Spanning Tree and Flow Control. This full standards-compliance, combined with auto-negotiation for 10/100/1000 Mbps and half/full duplex facilitates the expansion of your network to match your company's growing needs.

Avaya P330 Family Features

- Up to ten P330 switches in one stack
- 802. 3 compliant ports for full/ half duplex traffic, auto- negotiation, and flow control
- LAG (Link Aggregation Group)
- LAG and Link/ Inter- module redundancy
- IEEE 802.1d Spanning Tree
- Congestion Control
- GMP snooping allows dynamic multicast switching
- RADIUS protocol for security ï 802.1Q/ p VLAN and priority
- 1K VLANs
- Multiple VLANs per port
- Port Security ï Special secure VLAN for management
- Terminal and Modem interface
- CajunView network management
- CLI (Command Line Interface)
- Modem connection via PPP
- Embedded Web management
- RFC 2613 SMON Switch Monitoring
- RMON support for groups 1, 2, 3 and 9 (Statistics, History, Alarms, and Events)
- Stack port mirroring

Interfaces

- 24 x 10/ 100Base- TX ports with RJ connectors (P333T)
- 48 x10/ 100 100Base- TX ports with RJ connectors (P#34T)
- RS- 232 for terminal setup/ modem and PPP
- One Expansion module
- One Stack module

Standard Supported

- IEEE 802.3x Flow control on all ports
- IEEE 802.1Q/ p VLAN Tagging and priority on all ports
- IEEE 802.1D Spanning Tree protocol
- IEEE 802.3z Gigabit Ethernet ports
- IETF MIB- II, Bridge MIB, RMON, SMON

Physical Characteristics

Dimensions: 2U (3. 5"/ 88 mm) x 19", (h, w, d): (482.6 mm) x 17.7" (450 mm), Weight: 16.5lb(7.5Kg)

Environmental Conditions

- Operating Temp: 23 to 122 o F (- 5 to 50 o C)
- Humidity: 5% to 95% non- condensing

Power Consumption

- Power Entry (AC): 100- 240 VAC, 2A, 50/ 60 Hz
- Power Consumption: 150W max.

Agency Approval

EMC Emission:

- US - FCC Part 15, Subpart B, Class A
- Europe - EN55022 class A and EN61000- 3- 2
- Japan - VCCI- A

Immunity: Approved according to EN55024 and EN61000- 3- 3 Safety:

- UL for US approved according to UL1950 Std.
- C- UL (UL for Canada) approved according to C22.2 No. 950 Std.
- CE for Europe approved according to EN 60950 Std

CLEI Code According to Telcordia (Bellcore) KS- 22022 standard NEBS Level 3 Certified

Avaya P330 Network Management

Comprehensive network management is a key component of today's networks. Therefore we have provided multiple ways of managing the Avaya P330 to suit your needs.

Avaya P330 Device Manager (Embedded Web)

The built-in Avaya P330 Device Manager (Embedded Web Manager) allows you to manage an Avaya P330 stack using a Web browser without purchasing additional software. This application works with the Microsoft(r) Internet Explorer and Netscape(r) Navigator web browsers and Sun Microsystems Java(tm) Plug-in.

Avaya P330 Command Line Interface (CLI)

The Avaya P330 CLI provides a terminal type configuration tool for local or remote configuration of Avaya P330 features and functions.

CajunView(tm)

When you need extra control and monitoring or wish to manage other Cajun Campus equipment, then the CajunView network management suite is the answer. This suite provides the ease-of-use and features necessary for optimal network utilization.

- CajunView is available for Windows(r) NT(r)/2000 and Solaris 8.
- CajunView can operate in Stand-Alone mode with Windows(r) NT(r)/2000.
- CajunView operates under HP OpenView for Windows(r) NT(r)/2000 and Solaris 8.

Avaya P330 Network Monitoring

RMON MIBs - RFC 1757

- RMON support for groups 1,2,3 and 9 -
    - Statistics
    - History
    - Alarms
    - Events

SMON MIBs - RFC 2613

- SMON support for groups
    - Data Source Capabilities
    - Port Copy
    - VLAN and Priority Statistics

Bridge MIB Groups - RFC 2674

- dot1dbase and dot1dStp fully implemented.
- Support for relevant MIB objects: dot1q (dot1qBase, dot1qVlanCurrent)

Port Mirroring

The Avaya P330 provides port mirroring for additional network monitoring functionality. You can filter the traffic and mirror either incoming traffic to the source port or both incoming and outgoing traffic. This allows you to monitor the network traffic you need. Ports which are members in a Link Aggregation Group (LAG) cannot also be used as Port Mirroring Destination or Source ports.

SMON

The Avaya P330 supports Avaya's ground-breaking SMON Switched Network Monitoring, which the IETF has now adopted as a standard (RFC2613). SMON provides an unprecedented top-down monitoring of switched network traffic at the following levels:

- Enterprise Monitoring
- Device Monitoring
- VLAN Monitoring
- Port-level Monitoring

This top-down approach gives you rapid troubleshooting and performance trending to keep the network running optimally.

NOTE:
CajunView is required to run SMON monitoring. Note: You need to purchase one SMON License per Avaya P330 Stack.

Avaya P130

The Avaya P133G2 and P134G2 are part of the Avaya Cajun Switch Portfolio line of cost effective Workgroup Ethernet switches, which provide all the features needed at the network edge to implement emerging applications, yet they are simple to install, operate and manage.

Installing the Avaya P133G2 and P134G2 Ethernet switches at the network edge or a small enterprise allows customers to easily build a network fully featured to meet their requirements. The switch's integrated uplinks enable connectivity to the network Backbone and you can scale up as you grow: stack up to four P130 switches to create a single logical switch.

For the P133G2 and P134G2 products, application configuring, network operation and network monitoring are straightforward tasks due to the user-friendly WEB management capabilities. The P133G2 and P134G2 products have the same management and monitoring capabilities as other Avaya Cajun switches and therefore can also be centrally managed by the widely acclaimed CajunView(tm) network management suite of applications.

Like all Avaya MultiService Network products, the new P133G2 and P134G2 products are DayOne Ready for Voice, Video and Data Networking; Day One Ready means these products offer application optimization for converged voice, video and data networks. These products are designed to support a superior data network environment with capabilities for QoS, Policy Management and redundancy support that provides superior operational reliability and network availability.

Avaya P133G2

The P133G2 is a workgroup switch with 24 Fast Ethernet and 2 SFP GBIC slots for uplink and cascading. This unit has a 8.8 Gbps wire speed switching and forwarding rate.

Avaya P134G2

The P134G2 configuration provides 48 Fast Ethernet ports and 2 SFP GBIC slots.

General Features of the P133G2 and P134G2Cajun Switches

- 802.3 compliant ports for full/half duplex traffic, auto-negotiation, and flow control
- 802.1Q VLAN tagging
- Port based VLAN
- QoS Support
- Transparent IEEE 802.1p
- Per-port basis
- Congestion control per port
- Link Redundancy

- Link Aggregate Group (LAG) Supports both load sharing and redundancy as an incremental way to scale link bandwidth
- LAG Redundancy
- IEEE 802.1d Spanning Tree
- Load-Sharing Back-up Power Supply (BUPS)
- Port Mirroring
- SMON (RFC2613) IETF SMON standard for Layer 2

Interfaces

- 24 x 10/100Base-TX ports with RJ connectors
- 2 x SFP GBIC connectors
- RS-232 for terminal setup/modem and PPP

Standards Supported

- IEEE 802.3x Flow control on all ports
- IEEE 802.1Q/p VLAN Tagging and priority on all ports
- IEEE 802.1D Spanning Tree protocol
- IEEE 802.3z Gigabit Ethernet ports
- IETF MIB-II, Bridge MIB, RMON, SMON

Physical Characteristics

- Dimensions 2U (3.5"/88 mm) x 19", (h, w, d): (482.6 mm) x 13.8" (350 mm), Weight 11.4 lb (5.2 kg)

Environmental Conditions

- Operating Temp: 23 to 122 o F (-5 to 50 o C)
- Humidity: 5% to 95% non-condensing

Power Consumption

- Power Entry (AC): 100-240 VAC, 1A, 50/60 Hz
- Power Consumption: 75W max.

Agency Approval

- EMC Emission:
  - US - FCC Part 15, Subpart B, Class A
  - Europe - EN55022 class A and EN61000-3-2
  - Japan - VCCI-A
- Immunity: Approved according to EN55024 and EN61000-3-3
- Safety:
  - UL for US approved according to UL1950 Std.
  - C-UL (UL for Canada) approved according to C22.2 No.950 Std.
  - CE for Europe approved according to EN 60950 Std
- CLEI Code According to Telcordia (Bellcore) KS-22022 standard NEBS Level 3 Certified

Avaya P330

The Avaya P330 family of stackable Ethernet workgroup switches includes a range of modules with 10/100/1000 Mbps ports and a Layer 3 capability/ATM Expansion sub-module. The Avaya P333T switch has 24x10/100 Mbps ports and an Expansion sub-module slot. The optional expansion sub-modules provide additional Ethernet, Fast Ethernet, and Gigabit Ethernet connectivity.

An Avaya P330 stack can contain up to 10 switches and up to 3 backup power supply units. The stacked switches are connected using the Avaya X330STK stacking sub-modules which plug into a slot in the back of the Avaya P330. They are connected using the X330SC or X330LC cable (if the stack is split between 2 racks). The Avaya X330RC cable connects the top and bottom switches in the stack and provides redundancy and hot-swappability in the same way that modules can be swapped in a modular switching chassis.

The Avaya P330 is fully compliant with IEEE standards for VLAN Tagging, Gigabit Ethernet, Spanning Tree and Flow Control. This full standards-compliance, combined with auto-negotiation for 10/100/1000 Mbps and half/full duplex facilitates the expansion of your network to match your company's growing needs.

Avaya P330 Family Features

- You can connect up to 10 Avaya P330 switches in a stack. Moreover, this stack can be either in one rack or split over several racks using the X330LC Long Cable, according to your requirements.
- Avaya X330STK - this stacking sub-module is used to connect Avaya P330 switches in a stack, via the Octaplane.
- Avaya P330 BUPS - this back-up power supply module supports up to four Avaya P330 switches.
- One RJ45/RS232 front panel console connector for both terminal and modem sessions.
- Two fan units in every switch, with operation sensors.
- One virtual IP address for managing the whole stack, the P330 stack is managed as a single entity.
- Hot-swapping of one switch at a time - by activation of the redundant cable:
    - — Does not disrupt the operation of other Avaya P330 switches.
    - — Does not change stack configuration.
    - — Does not require network downtime.
- Connection through Telnet from the front panel ports of any switch, with:
    - — multiple levels of password protection
    - — login and inactivity timeouts

Auto-Negotiation

Every 10/100 port on the Avaya P330 supports Auto-Negotiation which automatically detects and supports the operating mode and speed of a connected device. Auto-negotiation is also supported on the Gigabit Ethernet ports for flow control mode only. This means that you can simply connect the Avaya P330 to Ethernet or Fast Ethernet equipment at full or half duplex without configuration.

Congestion Control

Congestion control is a key element of maintaining network efficiency as it prevents resource overload. The Avaya P330 supports congestion control on all Ethernet ports, using the following:

- Back Pressure in half duplex mode
- IEEE 802.3x Flow Control in full duplex mode

VLANs

The Avaya P330 VLANs are fully IEEE 802.1Q compliant and can handle up to 1k tagged VLANs.

Multiple VLANs per Port

The Avaya P330 provides the ability to set multiple (1K) VLANs per port. The three available Port Multi-VLAN binding modes are:

- Bound to All - the port is programmed to support the entire 3K VLANs range. Traffic from any VLAN is forwarded through a port defined as Bound to All.
- Bound to Configured - the port supports all the VLANs configured in the switch/stack. These may be either Port VLAN IDs (PVID) or VLANs that were manually added to the switch.
- Statically Bound - the port supports VLANs manually configured on it.

Leaky VLAN

Leaky VLAN provides the ability to send unicast traffic between two ports on different VLANs. Leaky VLAN will function only on modules (and sub-modules) with C/S 2.0 and higher.

Port Classification

With the Avaya P330, you can classify any port as regular or valuable. Setting a port to valuable means that a link fault trap can be sent even when the port is disabled. This feature is particularly useful for the software redundancy application, where you need to be informed about a link failure on the dormant port.

Network TIME Acquiring Protocols

The Avaya P330 supports the SNTP Protocol over UDP port 123. You can choose between SNTP or TIME protocol over UDP port 37.

MAC Security

You can define a port as secure to prevent it from learning new MAC addresses. If an unknown MAC or station tries to access a secure port, the intruder request is sent to the management station.

Link Aggregation Group (LAG)

LAG provides increased bandwidth and redundancy for critical high-bandwidth applications such as inter-stack links and connections to servers. You can aggregate the bandwidth of up to eight 10/100Base-Tx ports, two 100Base-FX or 1000Base-X ports. Load sharing ensures that if one of the port connections fails, the other connections will assume the load seamlessly. Load balancing guarantees that the traffic load at any level will be evenly divided among all the LAG links.

IP Multicast Filtering

IP Multicast allows you to send a single copy of an IP packet to multiple destinations, and can be used for various applications including video streaming and video conferencing. On LANs, IP Multicast packets are transmitted in MAC Multicast frames. Traditional LAN switches flood these Multicast packets to all stations in the VLAN. Multicast filtering functions may be added to the Layer 2 switches to avoid sending Multicast packets where they are not required. Layer 2 switches capable of Multicast filtering send the Multicast packets only to ports that connect members of that Multicast group. In order for this feature to operate correctly, you need in your network a router issuing IGMP queries.

NOTE:
IP Multicast filtering will function only based on the port's VLAN ID and not based on any VLAN bound to the port.

Radius Security

The Remote Authentication Dial-In User Service (RADIUS) is an IETF standard (RFC 2138) client/server security protocol. Security and login information is stored in a central location known as the RADIUS server. RADIUS clients such as the Avaya P330, communicate with the RADIUS server to authenticate users.

All transactions between the RADIUS client and server are authenticated through the use of a "shared secret" which is not sent over the network. The shared secret is an authentication password configured on both the RADIUS client and its RADIUS servers. The shared secret is stored as clear text in the client's file on the RADIUS server, and in the non-volatile memory of the Avaya P330. In addition, user passwords are sent between the client and server are encrypted for increased security.

Port Redundancy

Redundancy can be implemented between any two ports in the same stack at the link level. You can also assign redundancy between any two LAGs in the stack or between a LAG and a port. One port or LAG is defined as the primary port, and the other as the secondary port. In case the primary port link fails, the secondary port takes over.

Intermodule Redundancy

Intermodule redundancy includes all Port Redundancy functionality, and additionally maintains port integrity even when the primary port link fails as the result of a failure of the module. If the module on which the active port in an Intermodule Port Redundancy pair is located is powered down or removed from the stack, the secondary port in the Intermodule Port Redundancy pair takes over. Only one pair per stack can be set for Intermodule Port Redundancy.

Stack Redundancy

In the unlikely event that an Avaya P330 switch or Octaplane link should fail, stack integrity is maintained if the redundant cable is connected to the stack. The broken link is bypassed and data transmission continues uninterrupted. The single management IP address for the stack is also preserved for uninterrupted management and monitoring. Hot-Swappable You can remove or replace any module within the stack without disrupting operation or performing stack-level reconfiguration. You can therefore adapt the P330 to your requirements on the fly and with a down-time which is second to none. When you remove an expansion module from the stack, all configuration definitions on expansion modules are lost.

Backup Power Supply

Each Avaya P330 module comes with a Backup Power Supply (BUPS) connector. If the internal power supply should fail, the Avaya P330 BUPS (available separately,) will automatically supply power to the switch for uninterrupted operation.

Fans

The Avaya P330 module fans have integrated sensors which provide advance warnings of fan failure via management.

Network Management Agent (NMA) Redundancy

Since each Avaya P330 module has an integral SNMP agent, any module in a stack can serve as the stack NMA while other NMAs act as redundant agents in hot standby. If the live NMA fails then a backup is activated instantaneously.

Software Download

Avaya P330 includes a safe software download procedure in which backup code is always present. You should perform a reset after downloading software to the Module.

P330 Standards Supported by Avaya

The Avaya P330 complies with the following standards.

IEEE

- 802.3x Flow Control on all ports
- 802.1Q VLAN Tagging support on all ports and 802.1p compatible
- 802.1D Bridges and STA
- 803.2z Gigabit Ethernet ports
- 803.2u Ethernet/Fast Ethernet ports

IETF

- MIB-II - RFC 1213
- Bridge MIB for Spanning Tree - RFC 1492
- RMON - RFC 1757
- SMON - RFC 2613
- Bridge MIB Groups - RFC 2674 dot1dbase and dot1dStp fully implemented. Support for relevant MIB objects: dot1q (dot1qBase, dot1qVlanCurrent)

Avaya P330 Network Management

Comprehensive network management is a key component of today's networks. Therefore we have provided multiple ways of managing the Avaya P330 to suit your needs.

Avaya P330 Device Manager (Embedded Web)

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Avaya P330 Network Monitoring

RMON MIBs - RFC 1757

- RMON support for groups 1,2,3 and 9 -
  - — Statistics
  - — History
  - — Alarms
  - — Events

SMON MIBs - RFC 2613

- SMON support for groups
  - — Data Source Capabilities
  - — Port Copy
  - — VLAN and Priority Statistics

Bridge MIB Groups - RFC 2674

- dot1dbase and dot1dStp fully implemented.
- Support for relevant MIB objects: dot1q (dot1qBase, dot1qVlanCurrent)

Port Mirroring

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- Enterprise Monitoring
- Device Monitoring
- VLAN Monitoring
- Port-level Monitoring

This top-down approach gives you rapid troubleshooting and performance trending to keep the network running optimally.

NOTE:
CajunView is required to run SMON monitoring. Note: You need to purchase one SMON License per Avaya P330 Stack.

Avaya UPS Information

700VA 120V Online UPS

The 700VA 120V Online UPS provides 700VA/490 Watts/5.8 Amps at 120 Volts AC and battery holdover of 9 minutes at full load. Two optional Extended Battery Modules (EBM24) extend the run time to 156 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 120V models:

- Extended Battery Module (EBM24)
- Bypass Distribution Module 700-1500VA 120V
- SNMP Module

Safety Compliance: UL, CSA, NOM
EMC Compliance: FCC Class B, VCCI Class II.
Input: 120 Vac, 45-65Hz, auto sensing; user-selectable 110, 127 VAC; 6-foot attached cord with 5-15P.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (2) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 15kg = 34lbs

700VA 230V Online UPS

The 700VA 230V Online UPS provides up to 490 Watts and basic battery holdover of 9 minutes at full load. Output current for 208/220/230/240 volts is 3.8/3.6/3.4/3.3 Amps. Two optional Extended Battery Modules (EBM24) extend the run time to 156 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 230V models:

- Extended Battery Module (EBM24)
- Bypass Distribution Module 700-2000VA 230V
- SNMP Module

Safety Compliance: UL, CSA, NOM, CE
EMC Compliance: FCC Class B, EN 50091-2, VCCI Class II, IECS-003
Input: 230Vac, 50/60 Hz, auto sensing, user-selectable 220, 240; 2-meter cord with unterminated plug-end.

Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (2) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 15kg = 34lbs

700VA 100V Online UPS Japan

The 700VA 100V Online UPS provides 700VA/490 Watts/6.0 Amps at 100 Volts AC and battery holdover of 9 minutes at full load. Two optional Extended Battery Modules (EBM24) extend the run time to 156 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 100V models:

- Extended Battery Module (EBM24)
- Bypass Distribution Module 700-1500VA 120V
- SNMP Module

Safety Compliance: UL, CSA, NOM
EMC Compliance: FCC Class B, VCCI Class II.
Input: 100 Vac, 45-65Hz, auto sensing; user-selectable 110, 127 VAC; 6-foot attached cord with 5-15P.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (2) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 15kg = 34lbs

700VA 200V UPS Japan

The 700VA 200V Online UPS provides 700VA/490 Watts/4.0 Amps at 200 Volts AC and battery holdover of 9 minutes at full load. Two optional Extended Battery Modules (EBM24) extend the run time to 156 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 230V models:

- Extended Battery Module (EBM24)
- Bypass Distribution Module 700-2000VA 230V
- SNMP Module

Safety Compliance: UL, CSA, NOM, CE
EMC Compliance: FCC Class B, EN 50091-2, VCCI Class II, IECS-003
Input: 200Vac, 50/60 Hz, auto sensing, user-selectable 220, 240; 2-meter cord with unterminated plug-end.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (2) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 15kg = 34lbs

1500VA 120V Online UPS

The 1500VA 120V Online UPS provides 1500VA/1050 Watts/12.5 Amps at 120 Volts AC and battery holdover of 8 minutes at full load. Four optional Extended Battery Modules-EBM48 extend the run time to 144 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 1500VA model:

- Extended Battery Module (EBM48)
- Bypass Distribution Module 700-1500VA 120V
- SNMP Module

Safety Compliance: UL, CSA, NOM
EMC Compliance: FCC Class B, VCCI Class II.
Input: 120 Vac, 45-65Hz, auto sensing; user-selectable 110, 127 VAC; 6-foot attached cord with 5-15P.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (4) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 23kg = 50lbs

1500VA 230V Online UPS

The 1500VA 230V Online UPS provides up to 1050 Watts and basic battery holdover of 8 minutes at full load. Output current for 208/220/230/240 volts is 7.2/6.8/6.5/6.2 Amps. Four optional Extended Battery Modules (EBM24) extend the run time to 144 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 230V models:

- Extended Battery Module (EBM48)
- Bypass Distribution Module 700-2000VA 230V
- SNMP Module

Safety Compliance: UL, CSA, NOM, CE
EMC Compliance: FCC Class B, EN 50091-2, VCCI Class II, IECS-003
Input: 230Vac, 50/60 Hz, auto sensing, user-selectable 220, 240; 2-meter cord with unterminated plug-end.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (4) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 23kg = 50lbs

1500VA 100V Online UPS Japan

The 1500VA 100V Online UPS provides 1500VA/1050 Watts/12.7 Amps at 100 Volts AC and battery holdover of 8 minutes at full load. Four optional Extended Battery Modules-EBM48 extend the run time to 144 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included. The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 1500VA model:

- Extended Battery Module (EBM48)
- Bypass Distribution Module 700-1500VA 120V
- SNMP Module

Safety Compliance: UL, CSA, NOM
EMC Compliance: FCC Class B, VCCI Class II.
Input: 120 Vac, 45-65Hz, auto sensing; user-selectable 110, 127 VAC; 6-foot attached cord with 5-15P.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (4) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 23kg = 50lbs

1500VA 200V Online UPS Japan

The 1500VA 120V Online UPS provides 1500VA/1050 Watts/12.5 Amps at 120 Volts AC and battery holdover of 8 minutes at full load. Four optional Extended Battery Modules (EBM24) extend the run time to 144 minutes at full load. By grouping the six available 5-15R receptacles into two groups of three, the UPS allows users to shutdown one set of loads to allow longer run times for more critical loads during a power failure. Power management software is included.

The UPS chassis is suitable for tower and rack-mounted installations in a compact 2U height. Serial interface capabilities and Definity alarm contacts are standard.

Accessories for 700-1000VA 230V models:

- Extended Battery Module (EBM48)
- Bypass Distribution Module 700-2000VA 230V
- SNMP Module

Safety Compliance: UL, CSA, NOM, CE
EMC Compliance: FCC Class B, EN 50091-2, VCCI Class II, IECS-003
Input: 230Vac, 50/60 Hz, auto sensing, user-selectable 220, 240; 2-meter cord with unterminated plug-end.
Output: Six 5-15R receptacles in two load segments; sine wave; voltage +/- 3% of nominal.
Battery: (4) 12V 9 A/H sealed, lead-acid; maintenance free
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 23kg = 50lbs

Extended Battery Module-EBM24 700-1000VA

The Extended Battery Module-EBM24 supports the 700VA and 1000VA Online UPS with a 24v battery string to extend the run time (see table for run times). The EBM chassis is suitable for tower and rack-mounted installations in a compact 2U height. No other battery module or cabinet is compatible with the 700 or 1000VA Online UPS.

Maximum: Up to two Extended Battery Module (EBM24) per 700 or 1000VA Online UPS.
Safety Compliance: UL, CSA, NOM.
EMC Compliance: FCC Class B, VCCI Class II.
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 29.5kg = 65lbs

UPS Extended Battery Module-EBM48 1500-2000VA

The UPS Extended Battery Module-EBM48 supports the 1500VA and 2000VA online UPS with a 48v battery string to extend the run time up to 144 minutes aty full load. The EBM chassis is suitable for tower and rack-mounted installations in a compact 2U height.

Maximum: Up to four Extended Battery Module-EBM48 per 1500 or 2000VA Online UPS.
Safety Compliance: UL, CSA, NOM.
EMC Compliance: FCC Part 15 (Class B) and VCCI Class II.
Dimensions: (H x W x D) 89 mm x 432 mm x 494 mm = 3.5” x 17” x 19.4”
Weight: 65kg = 29.5lbs

SNMP MODULE 700-2000VA

The SNMP module adds direct control and monitoring capabilities in SNMP-based networks. The module is hot-swappable with the standard serial interface module.

Dimensions: (H x W X D) 4.5” x 4” x 1”
Weight: 0.2lbs

BYPASS DISTRIBUTION MODULE 120V 700-1500VA

The bypass distribution module (BDM) allows for replacement of the UPS or internal batteries without shutting down the connected load.

Dimensions: (H x W X D) 12” x 5” x 4”
Weight: 5.0lbs

PWR UPS BYPASS DISTR MOD S1 700VA-2KVA

The bypass distribution module (BDM) allows for replacement of the UPS or internal batteries without shutting down the connected load.

Dimensions: (H x W X D) 12” x 5” x 4”
Weight: 5.0lbs

Media Gateways

MCC1 Media Gateway

A MCC1 Media Gateway can be used as a Port Network (PN) cabinet. Doors on the front and rear of the MCC1 protect the internal equipment and allow easy access to the circuit packs. Each Media Gateway contains casters. Leveling feet keep the cabinet from rolling. Each corner of a Media Gateway can bolt to the floor, if required.

Typical MCC1 Media Gateway Layout

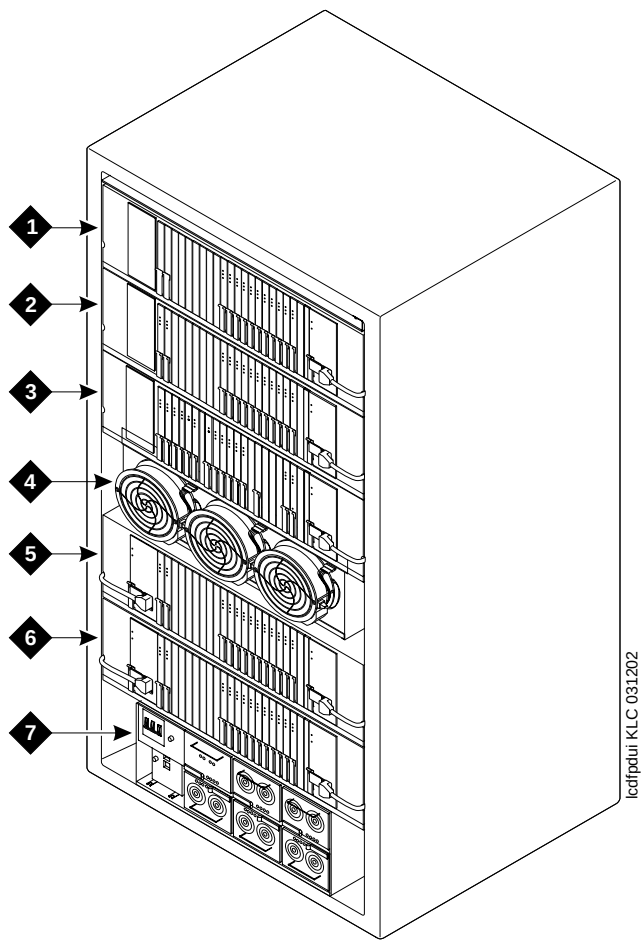


Figure notes

1	C Position Carrier	**5**	D Position Carrier
2	B Position Carrier	**6**	E Position Carrier
3	A Position Carrier	**7**	Power Distribution Unit
4	F Position Carrier		

AUX Connector Capacity

The AUX (auxiliary) connector is on the rear of the carrier A in the Port Cabinet. Up to 3 attendant consoles or telephone adjuncts can be powered by the AUX connector in the A position in the MCC1. In addition, up to 7 emergency transfer panels can be powered by the AUX connector in the A position in the MCC1.

Auxiliary Cabinet

The Auxiliary Cabinet contains the hardware to install optional equipment. The cabinet allows carrier, rack (width: 23 in.; 58.4 cm), and panel types of mounting. An Auxiliary Cabinet contains the following:

- Fuse panel (J58889AB) distributes –48 VDC to fused cabinet circuits
- AC-power receptacle strip provides switched and non-switched 120 VAC receptacles
- DC connector block is required when the cabinet is powered by an external DC source, or an AC to DC power supply that converts AC-power provided by the AC power strip switched-outlet to the required DC-power

MCC1 Media Gateway

- Port carrier (J58890BB) — 1 to 4
- In solutions with Asynchronous Transfer Mode (ATM), the ATM interface card is placed in a port carrier.
- Switch Node (SN) carrier (J58890SA) in solutions with a Center Stage Switch (CSS) — minimum of 1 in standard and high reliability solutions or minimum of 2 in critical reliability solutions

Typical PN Cabinet

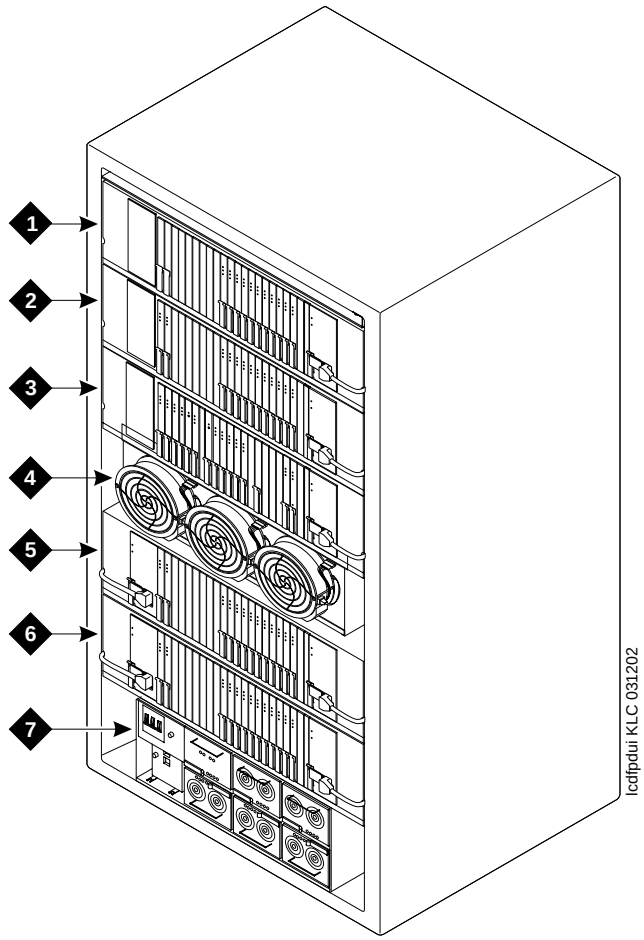


Figure notes

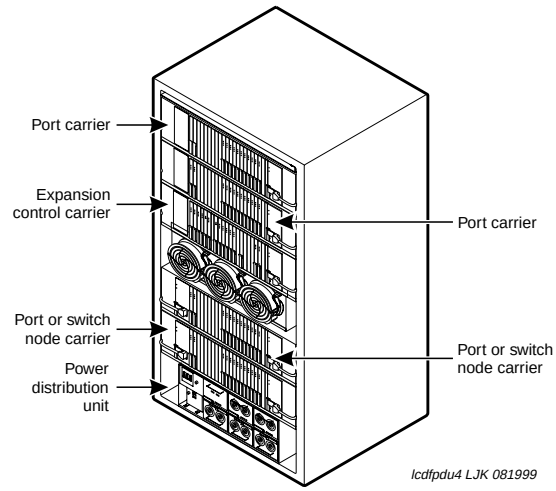
1	Port Carrier	**5**	Port or Switch Node Carrier
2	Port or Control Carrier	**6**	Port or Switch Node Carrier
3	Control Carrier	**7**	Power Distribution Unit
4	Fan Unit	**8**	

Port Network Cabinet

An Port Network (PN) cabinet contains the following carriers:

- Port carrier (J58890BB) — 1 to 4
- Expansion control carrier (J58890AF) — 1
- SN carrier (J58890SA) in CSS-connected r model — 0, 1, or 2 when required

MCC1 Media Gateway



In a minimal Dual PN cabinet configuration, the A, B, and C carrier positions are intended for the first port network in the cabinet. The D and E carrier positions are intended for the second port network in the cabinet. When a cabinet has two Port Networks, carrier position E must be used and populated first and carrier position D added and populated second.

MCC1 Carriers

The types of carriers that can install in the MCC1 Media Gateways are described in the following table.

Description	Cabinet
Port Carrier. Contains port, service, tone/clock, and EI circuit packs.	PN
Switch Node Carrier. Contains SNI circuit packs composing the Center Stage Switch (CSS).	PN
Expansion Control Carrier. Contains extra port circuit Packs, maintenance interface, and IPSI.	PN

Carrier Circuit-Pack Slots

There are 2 types of circuit pack slots in the carriers: Port and Service.

The purple-colored and white-colored circuit packs and slots are being replaced by circuit packs and slots labeled with gray and white rectangles, respectively. A label with a solid gray rectangle indicates a port slot/circuit pack. A label with an outlined white rectangle indicates a control slot/circuit pack.

- Port – colored purple or labeled with a gray rectangle and can accept any purple or gray-labeled circuit pack
- Service – colored purple or labeled with a gray rectangle; is a special type of circuit pack that does not have an I/O connector

Each port slot attaches to a 50-pin (25-pair) connector on the carrier's rear panel. A cable attaches to each connector and routes to the cross-connect field. Each slot containing a fiber optic interface circuit pack (EI or SNI) uses a fiber optic transceiver on the carrier's rear panel.

A current limiter board (CFY1B) plugs into the backplane of the carrier located in the A position only. The board supplies emergency transfer logic, current-limited power, 5 VDC to trip the main circuit breaker in an over-temperature condition, and the ringing transfer relay. Terminators on the backplane terminate each end of the processor expansion bus.

The following apparatus blank faceplates (with widths) cover unused circuit pack slots in the carriers to maintain proper air flow:

- Z100A1 (0.75 inches/1.9 cm)
- Z100C (0.5 inches/1.27 cm)
- Z100D (0.25 inches/0.64 cm)

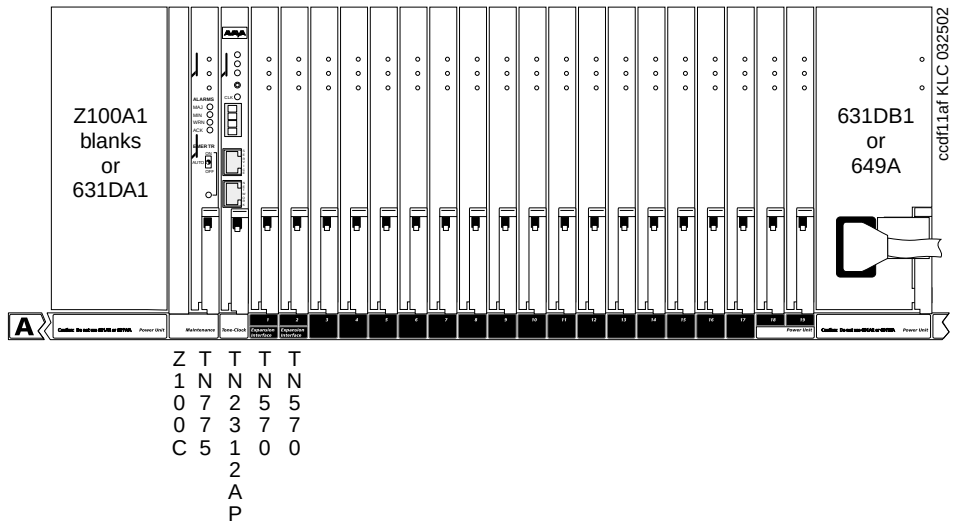
NOTE:
In the following illustrations, a balanced ring generator (BRG) is shown below the power unit slot in certain carriers. This means the power unit slot can include a 50 Hz BRG when optioned for France.

Expansion Control Carrier

The Expansion Control Carrier contains an EI or ATM Interface circuit pack in port slots 1 and 2. It is used in a fiber optic cabling path to another cabinet or the CSS in the same cabinet. These slots may contain optional port circuit packs.

The Expansion Control Carrier also contains port slots 3 to 19 and the AC or DC power units. The maintenance and tone-clock circuit packs are also shown. In the case of an IPSI-contolled port network the tone-clock is not needed. An optional neon power unit can be in slots 18 and 19.

Expansion Control Carrier (Front)



The following table describes Expansion Control Carrier connectors and functions.

Connector	Function
1 and 2 (A1 and A2)	Provides a fiber-optic cable interface to an expansion interface (EI) circuit pack in slot 1[1] or a copper cable interface for a DS1 Converter
1 to 19 (A1 to A19)	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
AUX	Provides interfaces for customer alarms, attendant console power, and emergency power transfer panels
TERMINAL	Connects a management terminal to the maintenance circuit pack in an expansion control carrier
P1	Provides position indicator of the carrier and access to alarm and control circuits
P2	Connects ringing voltage from the ring generator to the carrier and produces control signals

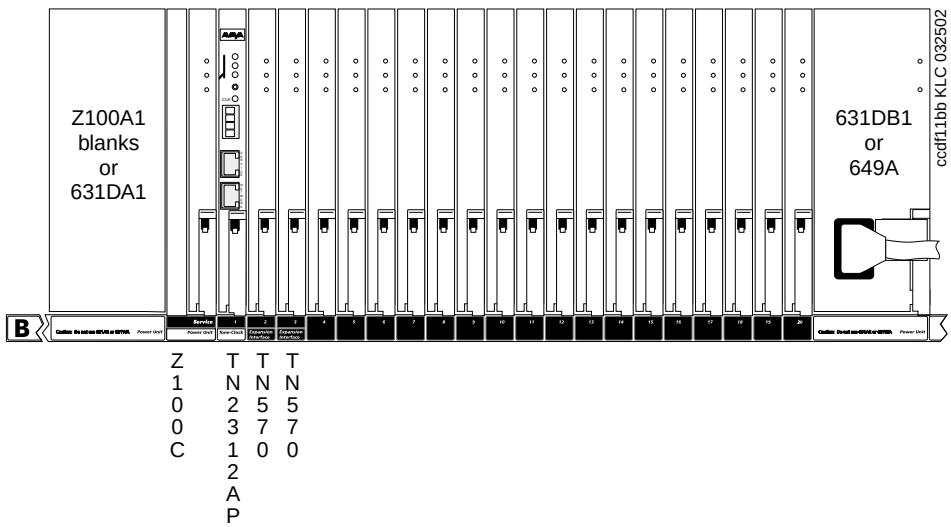
1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

Port Carrier

A Port Carrier contains the following circuit packs:

- Port slot locations 1 to 20 for the port circuit packs. Slot 1 may contain a tone-clock circuit pack when the port carrier is in the B position of an PN cabinet in a critical reliability system. Slot 2 contains an optional EI or ATM Interface circuit pack for a critical reliability system.
- Power unit service slot in which a power unit circuit pack or service circuit pack can be installed.
- AC or DC power units located at each end of the carrier.

Port Carrier (Front)



The following table describes Processor Carrier connectors and functions.

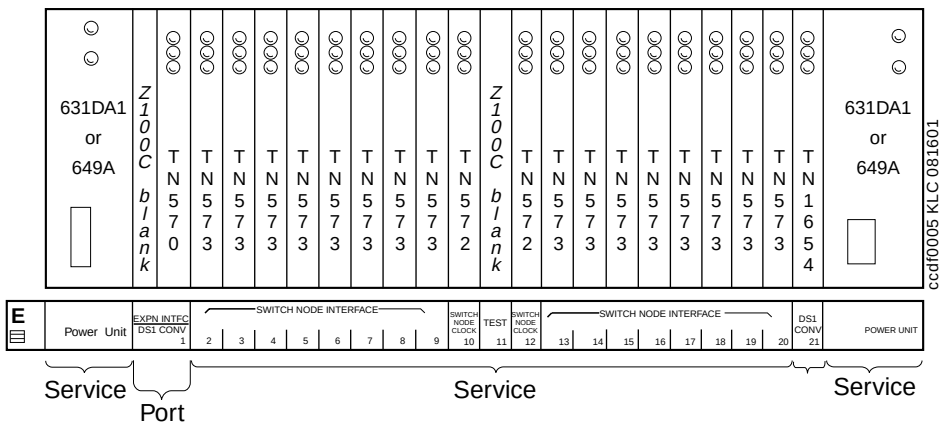
Connector	Function
1 to 20	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
P1	Provides position indicator of the carrier and access to alarm and control circuits

Switch Node Carrier

The Switch Node Carrier (SN) (J58890SA) can contain one or two switch node clocks, up to 16 Switch Node Interface (SNI) circuit packs, 1 or 2 DS1 converter circuit packs, one EI circuit pack, and 2 AC or DC power units.

The AC or DC power units are located at each end of the SN. The SN can be used when connecting 3 or more EPNs.

Switch Node Carrier (Front)



The following table describes Switch Node Carrier connectors and functions.

Connector	Function
1 (E1)	EI connector for the cable between the EI circuit pack in slot 1 and the Switch Node Interface (SNI) circuit pack in slot 2 for a duplicated PN only. Also used for a DS1 Converter circuit pack in slot 1.
2-9 and 13-20 (E2-E19 and E13-E20)	SN ports that are fiber optic cabling interfaces to the SNI circuit packs and other circuit packs connected to SN ports or circuit packs in expansion PNs
21 (E21)	Interface to connect the DS1 Converter circuit pack to the cross-connect field and an SNI circuit pack
P1	Provides the position indicator of the SN carrier and provides access to alarm and control circuits

SCC1 Media Gateway

This section describes the SCC1 Media Gateway. Each SCC1 has vertical slots that hold circuit packs. A blank faceplate covers each unused slot. Refer to the figure for an example of a typical SCC1.

Typical SCC1

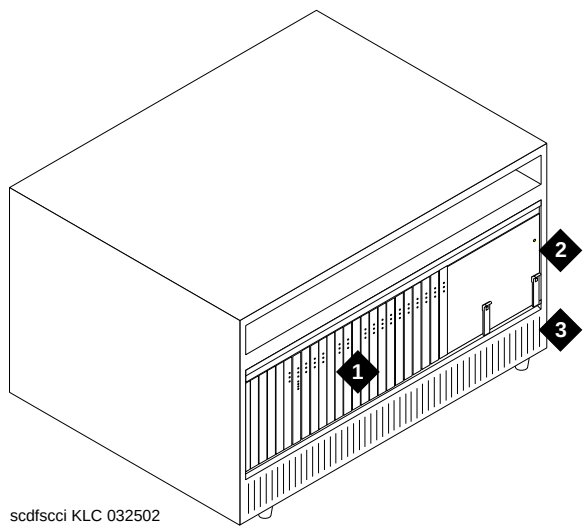


Figure notes

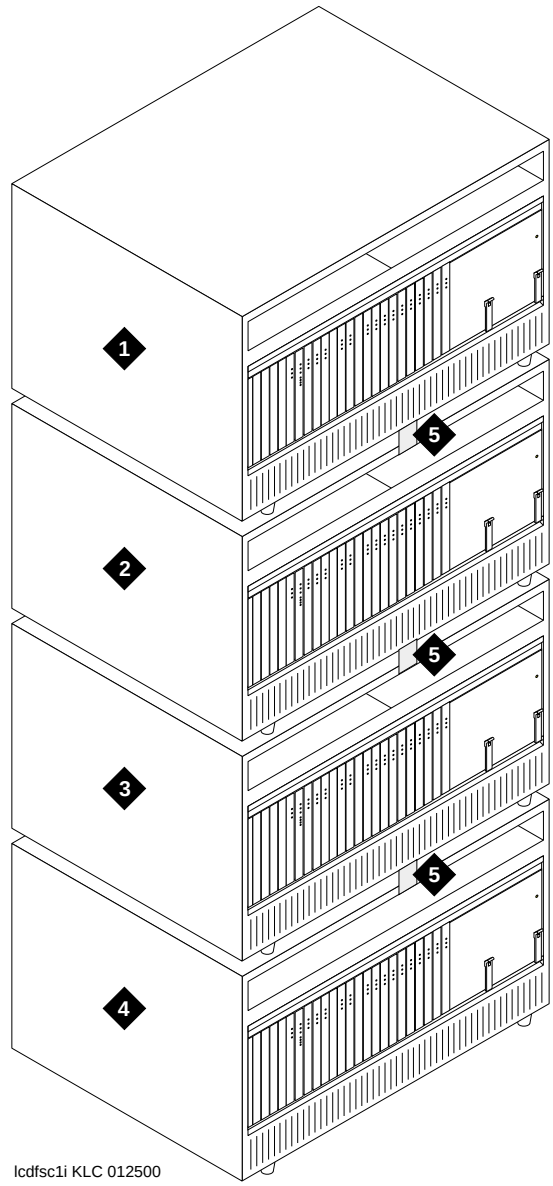
1 Circuit Packs

2 Power Converter

3 Air Circulation Vents

A maximum of 4 SCC1s can stack on top of each other. The cabinet positions are labeled A through D. The position of the basic control cabinet or expansion control cabinet is always labeled A. Additional port cabinet positions are labeled B, C, and D, sequentially.

Typical SCC1 Stack



lcdfsc1i KLC 012500

Figure notes

1	Port Cabinet D Position	**4**	Basic Control Cabinet or Expansion Control Cabinet
2	Port Cabinet C Position	**5**	Cabinet Clips
3	Port Cabinet or Duplicated Control Cabinet B Position		

The maximum number of cabinet stacks or port networks is 3.

Cabinet clips connect the cabinets together. At the rear of the cabinets, a ground plate connects between cabinets for ground integrity.

Carrier Circuit Pack Slots

There are 2 primary types of circuit pack slots in the carriers: Port and Service.

The purple-colored and white-colored circuit packs and slots are being replaced by circuit packs and slots labeled with gray and white rectangles, respectively. A label with a solid gray rectangle indicates a port slot/circuit pack.

- Port – colored purple or labeled with a gray rectangle and can accept any purple or gray-labeled circuit pack
- Service – colored purple or labeled with a gray rectangle; is a special type of circuit pack that does not have an I/O connector

Each port slot in a port carrier, an expansion control carrier, and a control carrier attaches to a 25-pair connector on the carrier's rear panel. A cable attaches to each connector and routes to the cross-connect field.

Blank faceplates cover empty carrier slots, as follows:

- 158J (4 in/9.2 cm) covers the area left of slot 1 in port cabinets
- 158P (0.75 in/1.9 cm) covers any unused slot.
- 158N (0.50 in/1.27 cm) is used with the LAN gateway in DEFINITY AUDIX R3 and CallVisor ASAI installations.
- 158G (0.25 in/0.63 cm) is used with the TN755 or TN2202.

In the following illustrations, a balanced ring generator (BRG) is shown below the power unit slot in certain carriers. This means the power unit slot can include a 50 Hz BRG when optioned for France.

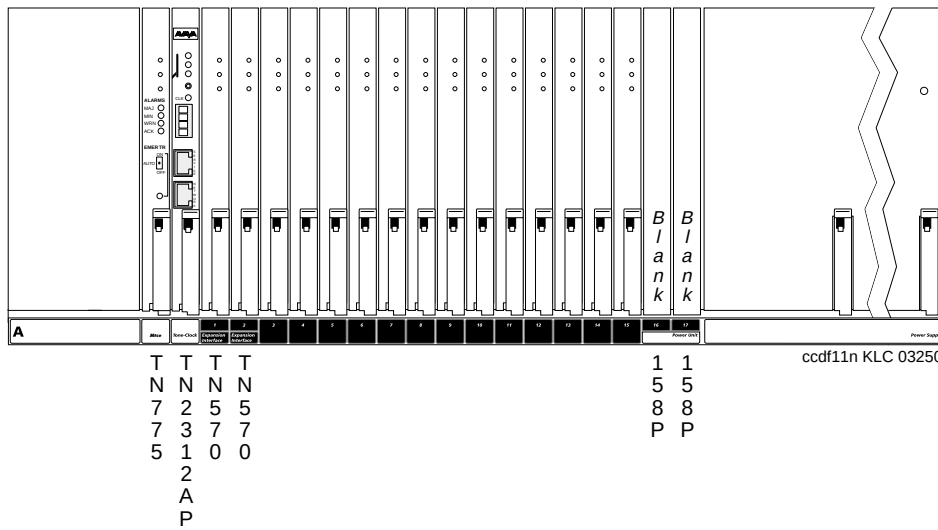
SCC1 Carriers

Expansion Control Cabinet

The Expansion Control Cabinet contains ports, an interface to a port cabinet, a maintenance interface, and a tone-clock. In the case of an IPSI-contolled port network, the tone-clock is not needed and is replaced by the IPSI.

The Expansion Control Cabinet is the first in an expansion PN stack of SCC1s. It has optional port circuit packs in port slots 2 to 17. The AC or DC power supply, located at the right side of the cabinet, supplies power.

Expansion Control Cabinet



The following table lists Expansion Control Cabinet connectors and their function.

Connector	Function
1 (A1)	Provides a fiber-optic cable interface to an expansion interface (EI) circuit pack in slot 1[1] or a copper cable interface for a DS1 Converter
2 to 17 (A2 to A17)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver
AUX (auxiliary)	Provides interface for customer alarms, attendant console power, and emergency power transfer panels
TERM (terminal)	Connects an administration terminal to the maintenance circuit pack

1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

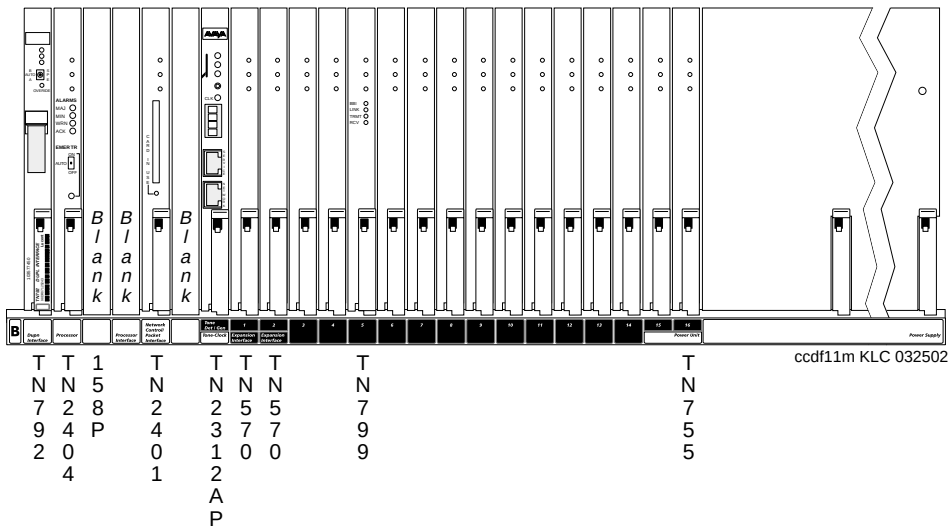
Duplicated Control Cabinet

The Duplicated Control Cabinet is optional and is only in the PN. This cabinet contains ports and a duplicated control complex.

The Duplicated Control Cabinet has dedicated white circuit pack slots for designated control circuit packs. The port circuit pack slots can contain any port circuit packs.

An AC or DC power supply, at the right side of the cabinet, supplies power to the cabinet. The cabinet contains a duplication interface circuit pack in slot DUPN INTFC. Refer to the figure that follows.

Duplicated Control Cabinet (Front)



The following table lists Duplicated Control Cabinet connectors and their function.

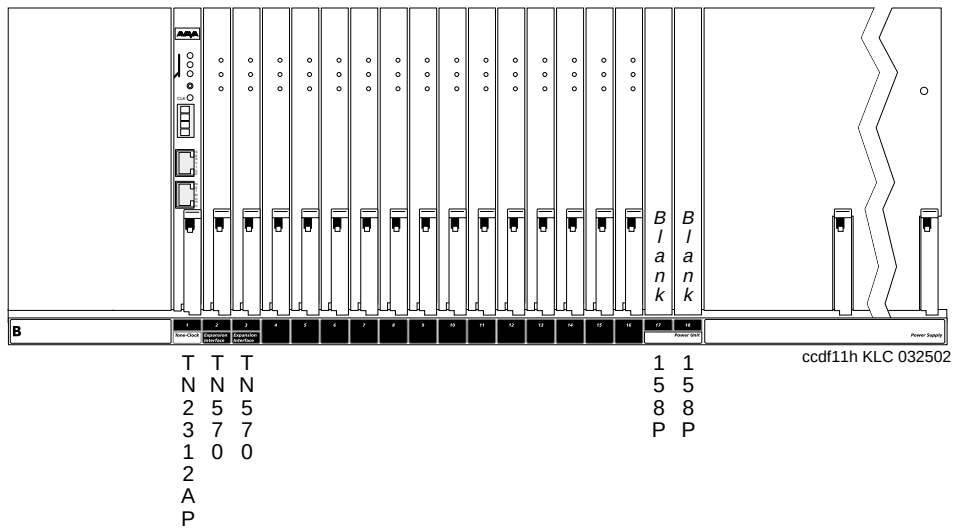
Connector	Function
01 to 16 (A01 to A16)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver
TERM (Terminal)	Connects an administration terminal to the processor circuit pack in the Duplicated Control Cabinet if the duplication interface circuit pack fails in the control carrier

Port Cabinet

The Port Cabinet is located in the PN and in expansion PNs. It contains ports and an interface to an Expansion Control Cabinet. The Port Cabinet has optional port circuit packs in port slots 1 to 18. In a critical-reliability system, these can include a tone-clock circuit pack in slot 1 and expansion interface circuit packs in slots 2 and 3. A neon power unit can be installed in slots 17 and 18.

The AC or DC power supply, located at the right side of the cabinet, supplies power to the cabinet. Refer to the following figure for an example of the Port Cabinet.

Port Cabinet



The following table lists Port Cabinet connectors and their function.

Connector	Function
2 to 3 (B2 to B3)	EI port that provides an interface for the fiber optic cable between an EI circuit pack in slot 2[1] or 3 in another PN or ATM system, or to an SNI
1 to 18 (B1 to B18)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver

1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

G700 Media Gateway in S8700 Multi-Connect

This section describes the G700 Media Gateway. It also briefly addresses other components within the G700 Media Gateway.

The basic architecture of the G700 Media Gateway is as follows:

- Contains four Media Module slots
- Allows the insertion and functioning of modules that traditionally fit in the Cajun Expansion/Personality slot and the Cajun Stacking/Octaplane slot
- Fits in a standard EIA-310-D 19 inch, 23 inch, and 600mm cabinet/rack with applicable brackets
- Usable on a table top
- Interfaces are standard connection types. A wall field or breakout panel is not required.
- Emissions are in the Class B level in some specified configurations
- Includes front-loaded media models
- Developed using ISO-compliant processes
- Uses materials that meet or exceed UL 94-VO requirements
- All interfaces with the exception of power, earth ground, and Octaplane are taken off the front of the unit. This facilitates movement and changes with the unit after initial installation.
- All cables that are plugged into the front of the G700 Media Gateway, whether in a table-top or rack-mount environment, are routed off to the side of the unit. A specified package of screws is included as part of this kit. One exception is the Octaplane, which is installed in the back of the box.

The G700 Media Gateway contains:

- VoIP resources
- A layer 2 switch
- Four Media Module slots
- A single, standard Cajun Expansion Module interface slot
- Provides for audio services on the LAN, such as conferencing, tones, and announcements

The G700 Media Gateway is designed to offer options and scalability. A customer can mix and match Media Modules as their business grows in size, as well as stack and add additional G700 Media Gateways if using the S8700 Media Server in Multi-Connect.

19 Inch G700 Media Gateway

G700 Media Gateways can be stackable, with up to 5 per stack using the S8700 Media Server with Multi-Connect. The G700 Media Gateway is functional on its own, with other G700 Media

Gateways, or in a stack that is mixed with G700 Media Gateways and Cajun P330 family devices such as the P333T, P333R, and P334. No more than 10 can be in a stack, of which 5 can be G700 Media Gateways.

To provide power to IP telephones without additional cabling, stack the G700 Media Gateways with the Avaya P333T-PWR power over Ethernet switch.

Media Modules are optional components that can be mixed and matched inside the G700 Media Gateway. The G700 Media Gateway uses Media Modules for traditional interfacing of service provider network access solutions such as T1/E1, ISDN-PRI, and Trunk Lines.

CAUTION:
All Media Modules are field replaceable. All Media Modules are hot swappable and use universal slots. The S8300 Media Server in a LSP configuration, Cajun Expansion Module, and the Cascade Stacking modules have special consideration for inserting and shutting down.

Cajun Expansion Module

The G700 Media Gateway is architecturally based on the Cajun P330 family of data products. Therefore, customers have the flexibility of using any of the Cajun Expansion Modules with the G700 Media Gateway. This means that customers can add additional Cajun LAN/WAN expansion modules directly to the G700 Media Gateway without requiring additional hardware.

CAUTION:
Cajun modules -both Expansion Modules and Octaplane Stacking Modules - are NOT hot-swappable. They are service-disrupting and can reset the entire G700 Media Gateway upon insertion or removal. Power down the system prior to any insertion or removal of Cajun modules.

Cajun Stacking Module (Octaplane)

Each Cajun P330 switch and G700 has an expansion slot at the rear for the Octaplane stacking fabric and is 4 gbps in each direction. The use of a separate expansion slot for the Octaplane Stacking Module means that all of the front panel ports are available for network connections.

The design of the Octaplane Stackable Switching System enables the removal of a failed P330 or G700 Media Gateway from the stack without disrupting the integrity of the stack. Special considerations apply. The broken link is bypassed and the transmission continues within the stack.

The following are the major highlights of Octaplane Module Stacking:

- Octaplane
- Stacking module inserted in the back
- Connection: up & down or ring close
- Immediate close of ring in case of link or box failure

Cabling for the Cajun Stack

Cable	Description	Length	Length (metric)
X330SC Short Octaplane Cable (30 cm)	Short Octaplane cable - light- colored, used to connect adjacent switches or switches separated by one Backup Universal Power Supply (BUPS) unit.	12 in	30 cm
X330LC Long Octaplane Cable (2 m)	Long Octaplane cable - light- colored, used to connect switches from two different physical stacks	6 ft	2 m
X330RC Redundant Octaplane Cable (2 m)	Redundant cable - black, used to connect the top and bottom switches of a stack.	6ft	2 m
X330L-LC Extra Long Octaplane Cable (8 m)	Extra-Long Octaplane cable - light-colored, used to connect switches from two different physical stacks	24 ft	8 m
X330L-RC Long Redundant Octaplane Cable (8 m)	Long Redundant cable - black, used to connect the top and bottom switches of a stack.	24 ft	8 m
Cajun Stacking Module X330STK	Stacking Module provides two backplane links		

Power Supply

The G700 Media Gateway uses an AC power supply. A power supply located in the G700 Media Gateway converts AC input power to voltages needed by the system.

NOTE:
Standard, off-the-shelf wiring and cables are used externally to the box. The AC power cord is selected based on the appropriate per-country outlet design and shipped with the unit. The cables between the Octaplane Stacking modules are pre-existing, and the I/O connectivity on the front is standard and does not require custom cables. These cables are very specialized; use only the prescribed cables.

Motherboard

The Motherboard resides internally within the G700 Media Gateway. This board is responsible for the following:

- The VoIP Engine that performs IP/UDP/RTP processing, echo cancellation, G.711 A/mu, G.729 and G723.1 encode/decode, fax relay, silence suppression, jitter buffer management, packet loss concealment, etc. The VoIP Engine will support 64 channels. If more then 64 channels are needed then a VoIP Media Module will be required.
- The Gateway Processor complex which controls all the resources inside the Gateway. The Gateway Processor functions include the Media Module Manager, Tone/Clock and H.248 signaling to the Gateway Controller.
- An Avaya P330 processor complex which is based on the Cajun P330 data switch architecture. It provides an eight port Layer 2-switch function and manages the Expansion and Cascade modules.
- Provides the electrical and physical connectivity for the four media module slots.

NOTE:
The Motherboard is not replaceable in the field.

Fans

The G700 Media Gateway contains four 12-volt fans. These are monitored and can be reported by SNMP to a management station.

LEDs

The G700Media Gateway uses two types of LEDs:

- Media Module LEDs
- System-Level LEDs

Media Module LEDs

Media Module LEDs have the following characteristics:

- The MM710 Media Modules have additional LEDs that function differently than the traditional LEDs.
- Each Media Module has at least three LEDs to indicate module/port status or maintenance and administration modes.
- The location, spacing, and labeling is fixed for all LEDs on every Media Module.
- The LEDs are mounted on the Media Module printed wiring board and are placed so that they show through an opening.

System-Level LEDs

The LED board provides visual indication of system and data port status, but also allows the customer to switch between status indication modes. In addition, the System Level-LEDs have the following characteristics:

- An LED board is located in the upper left front area of each G700 Media Gateway. The front of the LED board displays the LEDs in an oblong fascia panel.
- The LED board is intended to provide visual indication of system and Ethernet port status and to allow the customer to switch between status indication modes.

The LED panel needs to be removed when installing or removing the S8300 Media Server in a LSP configuration. The two components should be installed or removed as a unit.

NOTE:
The LED panel is not the same size as standard Media Module slots. Trying to insert a Media Module into the LED slot, or vice versa, will not work.

Gateway Software

Gateway software is responsible for individual Media Gateway operations, terminating H.248 on the G700 Media Gateway, and for interacting with maintenance operations.

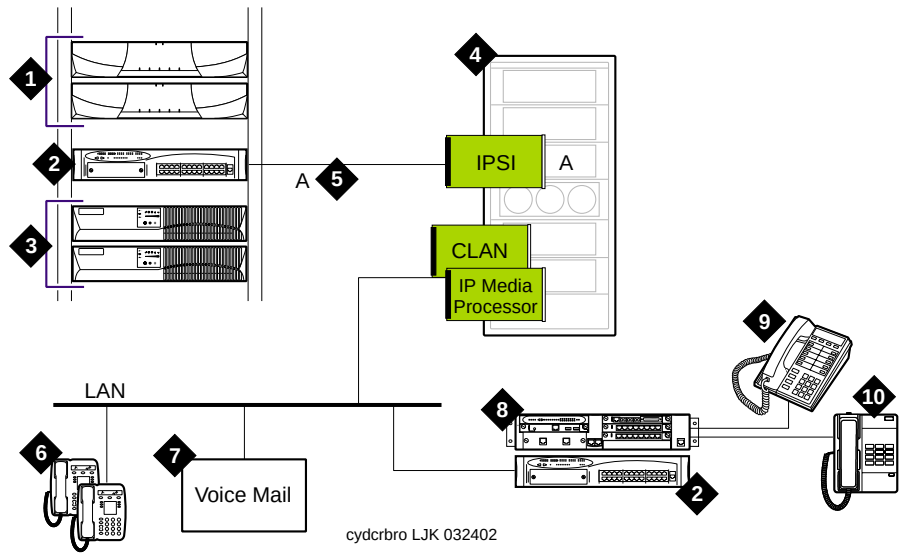
Maintenance Software

The Media Server with the G700 Media Gateway has a dual maintenance strategy in that there is maintenance software that runs on both the G700 Media Gateway platform and the Media Server for the subsystems on the platform. This platform software performs initialization and motherboard maintenance, along with internal environmental monitoring.

The media modules, on the other hand, are tested and brought into service by the Media Server maintenance software after the G700 Media Gateway registers with the Media Server. While the G700 Media Gateway maintenance software is aware of the media modules, the modules and associated ports are controlled by the Media Server. Error logs are maintained on the Media Server.

Refer to the following figure for an example of how S8700 Multi-Connect connects with a G700 Media Gateway

S8700 Multi-Connect with a G700 Media Gateway



Item #	Description
1	S8700 Media Servers in a Multi-Connect Configuration
2	Etherenet Switch: AVAYA provided.
3	UPS units: Two UPS units one for each server.
4	MCC1 Media Gateway
5	Dedicated LAN connectivity to the IPSI board in the MCC1 Media Gateway.
6	IP Phones off of the customer's LAN
7	Voice Mail: Intuity AUDIX shown connected via IP.
8	G700 Media Gateway connected via the LAN to the CLAN board located in the MCC1 Media Gateway. The S8300 Media Server in an LSP configuration is located in the G700 Media Gateway. In the event of a loss in communication between the S8700 and the G700 the LSP will provide a backup for the endpoints that register with it.
9	DCP Phones: AVAYA Multi-Function Digital Phones.
10	Analog Connectivity: Analog phones, lines and trunks.

G600 Media Gateway

The G600 Media Gateway has the following characteristics:

- 19 inches wide, 13 inches high and 21 inches deep. It is designed for rack mounting only.
- 10 universal slots plus one power supply slot are available
- Maximum of 64 port networks.
- Maximum of 4 G600 Media Gateways per PN. Must be collocated in the 19 inch data rack due to TDM cable length
- A PN consists of a control G600 Media Gateway that is designated A, and second, third, and fourth optional G600 Media Gateways designated as B, C, and D respectively.
- Cabinet address ID within the port network is set via a DIP switch on the back plane.
- Circuit packs are insert and removed from the front of the cabinet. Cabinet I/O is through the back and through a front cable pass-through slot.
- G600 Media Gateway is AC powered only. There are no internal batteries and DC power is not an option.
- A RJ45 patch panel is recommended for cross-connect to LAN or wall field.

The following circuit packs can be used in the G600 Media Gateway based port network.:

IP Server Interface (TN2312AP)

The IP Server Interface (IPSI) provides transport of CCMS messages over IP allowing the S8700 Media Server to communicate with the PNs. The IPSI is required to provide control network signaling over the customer's LAN and WAN. Tone generation, tone detection, global call classification, as well as stratum 4 type clock generation are provided on the IPSI board.

C-LAN (TN799DP)

The C-LAN circuit pack, TN799DP provides call control for all IP endpoints connected to the S8700 Media Server for IP Connect. There is a maximum number of 64 C-LANs per configuration. The number of C-LANs required depends on the number of devices connected as well as the options being utilized by the endpoint. It may be advantageous to segregate IP voice control traffic from device control traffic as a safety measure.

The default value for C-LAN socket usage of H.323 Tie Trunks is determined by dividing the total number of H.323 Tie Trunk utilizing sharing by 31. Each IP endpoint requires the use of some number of C-LAN sockets, which is the software object used to connect a TN799 board to the IP Network. The TN799DP board supports up to 500 sockets.

The C-LAN differs from an IP Media Processor in that the former controls the call and the latter provides the codecs used for the call's audio.

To take advantage of downloadable firmware capability there must be at least one TN799DP C-LAN and access to the public Internet for firmware downloads to other downloadable circuit packs. Downloads and instructions have been posted to:

http://www.avaya.com/support/
Click on Online Services and then Download Software needed.

IP Media Processor (TN2302AP)

The S8700 Media Server for IP Connect requires resources on an IP Media Processor (TN2302AP) circuit pack for inter-port network bearer communications. It is also used for bearer communications with IP endpoints on both systems. The TN2302AP Media Processor includes a 10/100 BaseT Ethernet interface to support H.323 endpoints for IP trunks and H.323 end-points. The TN2302AP can perform echo cancellation, silence suppression, dual-tone multi-frequency (DTMF) detection, and conferencing.

The TN2302AP supports the following codecs, fax detection for them beginning with vintage 32, and conversion between them:

- G.711 (mu-law or a-law, 64Kbps)
- G.723.1 (6.3Kbps or 5.3Kbps audio)
- G.729A (8Kbps audio)
- G.729, G.729B, G.729AB

G600 Media Gateway Cooling

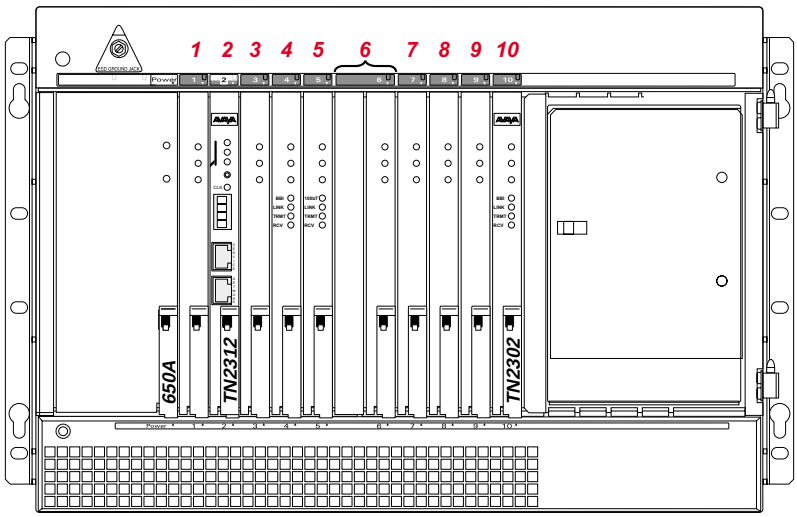
G600 Media Gateway cooling is provided by three 12-volt DC variable speed fans integrated into the back of the cabinet. The fans pull air through the front and left sides and up through the cabinet. Air exits out the back of the cabinet. Speed control is provided by the 650A Global Power supply. It varies the fan input voltage between 8VDC and 14VDC depending on a temperature sensor mounted in the power supply.

The fan assembly includes the three fans, a base plate to which they are attached, wiring, and an AMP connector that plugs into a cable that connects to the backplane. The assembly is easily installed and removed. The entire assembly will be replaced as a unit if a fan fails. A fan failure will result in the following:

- An alarm through the power supply is detected as a power alarm
- Remaining fans go to high speed
- The red LED on the 650A Global Power supply faceplate light

Refer to the following figure for an example of the G600 Media Gateway.

G600 Media Gateway



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G700 Media Gateway

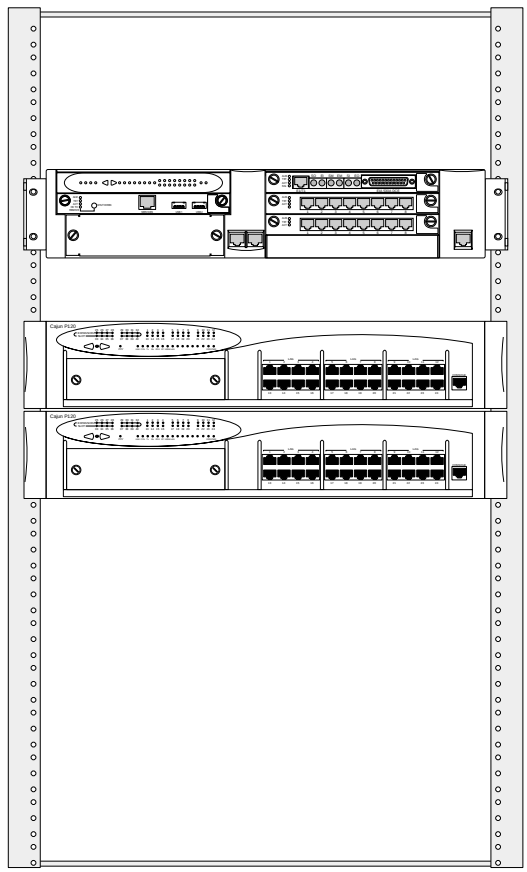
This section describes the G700 Media Gateway. It also briefly addresses other components within the G700 Media Gateway.

The basic architecture of the G700 Media Gateway is as follows:

- Contains four Media Module slots
- Allows the insertion and functioning of modules that traditionally fit in the Cajun Expansion/Personality slot and the Cajun Stacking/Octaplane slot
- Fits in a EIA-310-D standard 19 inch, 23 inch, and 600mm cabinet/rack with applicable brackets
- Usable on a table top
- Interfaces are standard connection types. A wall field or breakout panel is not required.
- Emissions are in the Class B level in some specified configurations
- Includes front-loaded media models
- Developed using ISO-compliant processes
- Uses materials that meet or exceed UL 94-VO requirements
- All interfaces with the exception of power, earth ground, and Octaplane are taken off the front of the unit. This facilitates movement and changes with the unit after initial installation.
- All cables that are plugged into the front of the G700 Media Gateway, whether in a table-top or rack-mount environment, are routed off to the side of the unit. A specified package of screws is included as part of this kit. One exception is the Octaplane, which is installed in the back of the box.

Refer to the following figure.

G700 Media Gateway



scdcrck1 KLC 031902

Faceplate Interfaces on S8300 Media Server

The following interfaces are accessible to Avaya personnel and customers via the faceplate:

- A 10/100BaseT Ethernet interface provided via an RJ45 filtered connector
- Two USB interfaces using standard USB connectors
- A shutdown switch
- LEDs for status reporting

Backplane Interfaces

All connections to the backplane within the G700 Media Gateway are accomplished through a 90-pin connector. These interfaces are present in all media modules. They include:

- The TDM bus
- ENET packet interface
- A 10/100BaseTX Ethernet twisted pair interface
- Power and ground
- Other control signals

The G700 contains:

- VoIP resources
- A layer 2 switch
- Four Media Module slots
- A single, standard Cajun Expansion Module interface slot
- Provides for audio services on the LAN, such as conferencing, tones, and announcements

The G700 Media Gateway is designed to offer options and scalability. A customer can mix and match Media Modules as their business grows in size, as well as stack and add additional G700 Media Gateways if using the S8700 Media Server with Multi-Connect.

19 Inch G700 Media Gateway

The G700 Media Gateway is functional on its own, with other G700 Media Gateways, or in a stack that is mixed with G700 Media Gateways and Cajun P330 family devices such as the P333T, P333R, and P334.

To provide power to IP telephones without additional cabling, stack the G700 Media Gateways with the Avaya P333T-PWR.

Media Modules are optional components that can be mixed and matched inside the G700 Media Gateway. The S8300 Media Server with the G700 Media Gateway uses Media Modules for traditional interfacing of service provider network access solutions such as T1/E1, ISDN-PRI, and Trunk Lines.

All Media Modules are field replaceable. All Media Modules are hot swappable and use universal slots. The S8300 Media Server, Cajun Expansion Module, and the Cascade Stacking modules have special consideration for inserting and shutting down.

Cajun Expansion Module

The G700 Media Gateway is architecturally based on the Cajun P330 family of data products. Therefore, customers have the flexibility of using any of the Cajun Expansion Modules with the G700 Media Gateway. This means that customers can add additional Cajun LAN/WAN expansion modules directly to the G700 Media Gateway without requiring additional hardware.

CAUTION:
Cajun modules -both Expansion Modules and Octaplane Stacking Modules - are NOT hot-swappable. They are service-disrupting and can reset the entire G700 Media Gateway upon insertion or removal. Power down the system prior to any insertion or removal of Cajun modules.

Cajun Stacking Module (Octaplane)

Each Cajun P330 switch and G700 has an expansion slot at the rear for the Octaplane stacking fabric and is 4 gbps in each direction. The use of a separate expansion slot for the Octaplane Stacking Module means that all of the front panel ports are available for network connections.

The design of the Octaplane Stackable Switching System enables the removal of a failed P330 or G700 Media Gateway from the stack without disrupting the integrity of the stack. Special considerations apply. The broken link is bypassed and the transmission continues within the stack.

The following are the major highlights of Octaplane Module Stacking:

- Octaplane
- Stacking module inserted in the back
- Connection: up & down or ring close
- Immediate close of ring in case of link or box failure

Cabling for the Cajun Stack

Cable	Description	Length	Length (metric)
X330SC Short Octaplane Cable (30 cm)	Short Octaplane cable - light- colored, used to connect adjacent switches or switches separated by one Backup Universal Power Supply (BUPS) unit.	12 in	30 cm
X330LC Long Octaplane Cable (2 m)	Long Octaplane cable - light- colored, used to connect switches from two different physical stacks	6 ft	2 m
X330RC Redundant Octaplane Cable (2 m)	Redundant cable - black, used to connect the top and bottom switches of a stack.	6ft	2 m
X330L-LC Extra Long Octaplane Cable (8 m)	Extra-Long Octaplane cable - light-colored, used to connect switches from two different physical stacks	24 ft	8 m
X330L-RC Long Redundant Octaplane Cable (8 m)	Long Redundant cable - black, used to connect the top and bottom switches of a stack.	24 ft	8 m
Cajun Stacking Module X330STK	Stacking Module provides two backplane links		

Power Supply

The G700 Media Gateway uses an AC power supply. A power supply located in the G700 Media Gateway converts AC input power to voltages needed by the system.

NOTE:
Standard, off-the-shelf wiring and cables are used externally to the box. The AC power cord is selected based on the appropriate per-country outlet design and shipped with the unit. The cables between the Octaplane Stacking modules are pre-existing, and the I/O connectivity on the front is standard and does not require custom cables. These cables are very specialized; use only the prescribed cables.

Motherboard

The Motherboard resides internally within the G700 Media Gateway. This board is responsible for the following:

- The VoIP Engine that performs IP/UDP/RTP processing, echo cancellation, G.711 A/mu, G.729 and G723.1 encode/decode, fax relay, silence suppression, jitter buffer management, packet loss concealment, etc. The VoIP Engine will support 64 channels. If more then 64 channels are needed then a VoIP Media Module will be required.
- The Gateway Processor complex which controls all the resources inside the Gateway. The Gateway Processor functions include the Media Module Manager, Tone/Clock and H.248 signaling to the Gateway Controller.
- An Avaya P330 processor complex which is based on the Cajun P330 data switch architecture. It provides an eight port Layer 2-switch function and manages the Expansion and Cascade modules.
- Provides the electrical and physical connectivity for the four media module slots.

NOTE:
The Motherboard is not replaceable in the field.

Fans

The G700 Media Gateway contains four 12-volt fans. These are monitored and can be reported by SNMP to a management station.

LEDs

The S8300 Media Server with the G700Media Gateway uses two types of LEDs:

- Media Module LEDs
- System-Level LEDs

Media Module LEDs

Media Module LEDs have the following characteristics:

- The MM710 Media Module and S8300 Media Server have additional LEDs that function differently than the traditional LEDs.
- Each Media Module has at least three LEDs to indicate module/port status or maintenance and administration modes.
- The location, spacing, and labeling is fixed for all LEDs on every Media Module.
- The LEDs are mounted on the Media Module printed wiring board and are placed so that they show through an opening.

System-Level LEDs

The LED board provides visual indication of system and data port status, but also allows the customer to switch between status indication modes. In addition, the System Level-LEDs have the following characteristics:

- An LED board is located in the upper left front area of each G700 Media Gateway. The front of the LED board displays the LEDs in an oblong fascia panel.
- The LED board is intended to provide visual indication of system and Ethernet port status and to allow the customer to switch between status indication modes.

The LED panel needs to be removed when installing or removing the S8300 Media Server or Standby S8300 Media Server. The two components should be installed or removed as a unit.

NOTE:
The LED panel is not the same size as standard Media Module slots. Trying to insert a Media Module into the LED slot, or vice versa, will not work.

Gateway Software

Gateway software is responsible for individual Media Gateway operations, terminating H.248 on the G700 Media Gateway, and for interacting with maintenance operations.

Maintenance Software

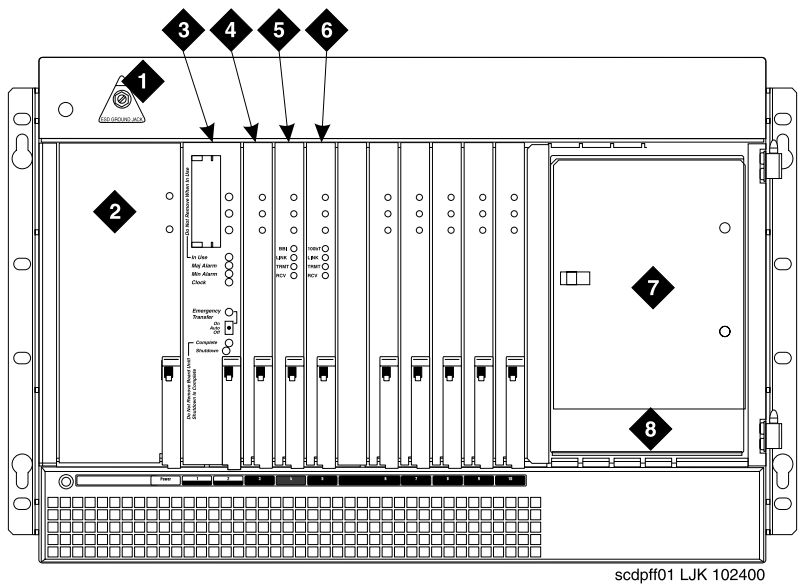
The Media Server with the G700 Media Gateway has a dual maintenance strategy in that there is maintenance software that runs on both the G700 Media Gateway platform and the Media Server for the subsystems on the platform. This platform software performs initialization and motherboard maintenance, along with internal environmental monitoring.

The media modules, on the other hand, are tested and brought into service by the Media Server maintenance software after the G700 Media Gateway registers with the Media Server. While the G700 Media Gateway maintenance software is aware of the media modules, the modules and associated ports are controlled by the Media Server. Error logs are maintained on the Media Server.

G600 Media Gateway for S8100 Media Server

The G600 Media Gateway weighs 40-50 pounds (18-22.5 kilograms) fully loaded and is about 12 x 19 x 22 inches (30 x 48 x 55 centimeters). The first cabinet includes 7 slots for circuit packs and a power supply and supports up to 618 ports (may require a second cabinet). The second and third cabinets include 10 slots each for circuit packs. The media gateway is designed for rack mounting but can be floor mounted.

G600 Media Gateway



1 Grounding Receptacle

2 650A Power Supply

3 TN2314 Processor

4 TN744D Call-Classifier-Detector

5 TN799B/C C-LAN

6 TN2302AP IP Media Processor

7 Storage Area (containing grounding wrist strap, backup PCMIA flashcard, and documentation library CD)

8 Fiber pass-through area

NOTE:
Slots 1 and 2 are reserved for the TN795 circuit pack. Slot 3 is recommended for the TN744D circuit pack, but any other slot is acceptable.

MCC1 Media Gateway

A MCC1 Media Gateway can be used as a Port Network (PN) cabinet. Doors on the front and rear of the MCC1 protect the internal equipment and allow easy access to the circuit packs. Each cabinet contains casters. Leveling feet keep the cabinet from rolling. Each corner of a cabinet can bolt to the floor, if required.

Typical MCC1 Media Gateway Layout

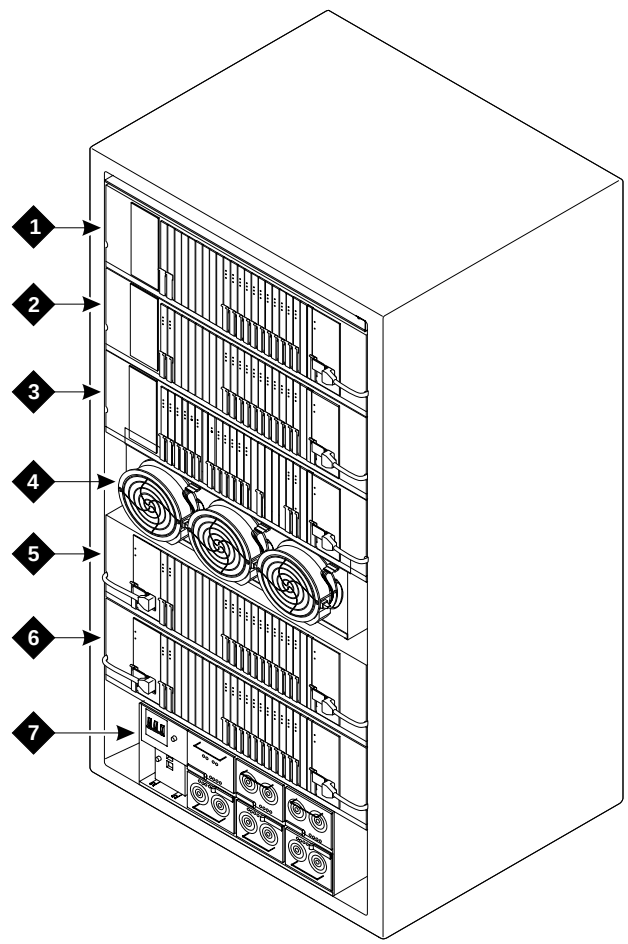


Figure notes

1	C Position Carrier	**5**	D Position Carrier
2	B Position Carrier	**6**	E Position Carrier
3	A Position Carrier	**7**	Power Distribution Unit
4	F Position Carrier		

AUX Connector Capacity

The AUX (auxiliary) connector is on the rear of the control carrier. Up to 3 attendant consoles or telephone adjuncts can be powered by the AUX connector in the A position in the MCC1. In addition, up to 7 emergency transfer panels can be powered by the AUX connector in the A position in the MCC1.

Auxiliary Cabinet

The Auxiliary Cabinet contains the hardware to install optional equipment. The cabinet allows carrier, rack (width: 23 in.; 58.4 cm), and panel types of mounting. An Auxiliary Cabinet contains the following:

- Fuse panel (J58889AB) distributes –48 VDC to fused cabinet circuits
- AC-power receptacle strip provides switched and non-switched 120 VAC receptacles
- DC connector block is required when the cabinet is powered by an external DC source or an AC to DC power supply that converts AC-power provided by the AC power strip switched-outlet to the required DC-power

Port Network Cabinet

A PN cabinet contains the following carriers:

- Port carrier (J58890BB) — 1 to 4
- In solutions with Asynchronous Transfer Mode (ATM), the ATM interface card is placed in a port carrier.
- Control carrier (J58890AH) in solutions — 1
- Duplicated control carrier (J58890AJ) in solutions — 1 in high or critical reliability configurations
- Processor carrier (J58890AP) in solutions — 1 in all solutions, 2 in high reliability and critical reliability solutions
- Switch Node (SN) carrier (J58890SA) in solutions with a Center Stage Switch (CSS) — minimum of 1 in standard and high reliability solutions or minimum of 2 in critical reliability solutions

Typical PN Cabinet

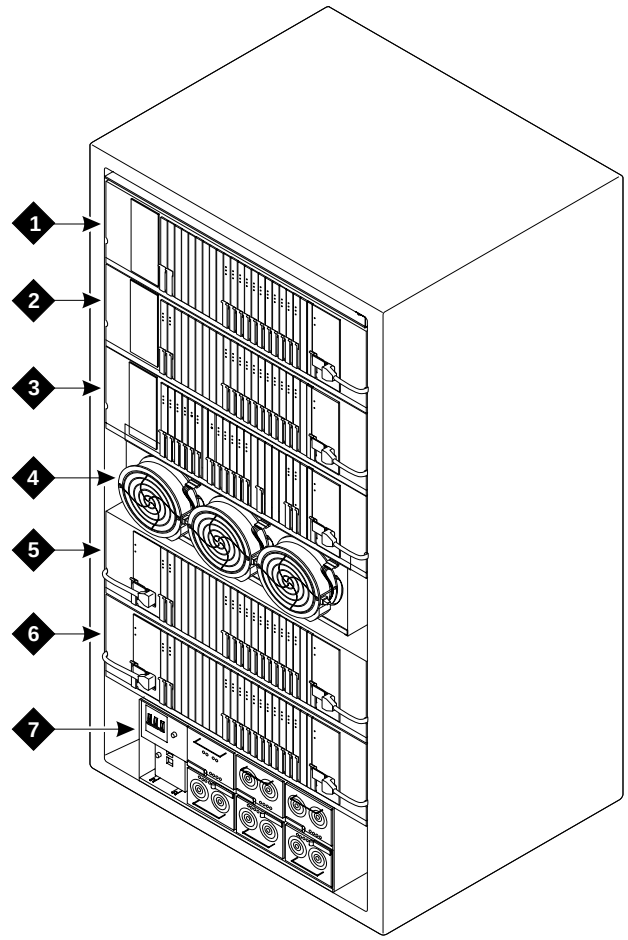


Figure notes

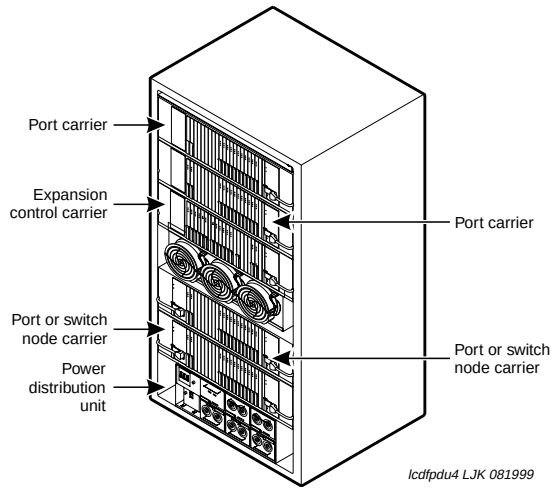
1	Port Carrier	**5**	Port or Switch Node Carrier
2	Port or Control Carrier	**6**	Port or Switch Node Carrier
3	Control Carrier	**7**	Power Distribution Unit
4	Fan Unit	**8**	

Expansion Port Network Cabinet

An Expansion Port Network (EPN) cabinet contains the following carriers:

- Port carrier (J58890BB) — 1 to 4
- Expansion control carrier (J58890AF) — 1
- SN carrier (J58890SA) in CSS-connected r model — 0, 1, or 2 when required

MCC1 EPN Cabinet



In a minimal Dual EPN cabinet configuration, the A, B, and C carrier positions are intended for the first port network in the cabinet. The D and E carrier positions are intended for the second port network in the cabinet. When a cabinet has two Port Networks, carrier position E must be used and populated first and carrier position D added and populated second.

MCC1 Carriers

The types of carriers that can install in the MCC1 Media Gateways are described in the following table.

Description	Cabinet
Control Carrier. Contains processor circuit packs to perform call processing, maintenance, and administration. This carrier can also contain port circuit packs	PN
Duplicated Control Carrier(optional). Contains duplicate processor circuit packs to perform call processing, maintenance, and administration identical to the Control Carrier. This carrier can also contain port circuit packs.	PN
Processor Carrier. Contains processing circuit packs to perform call processing, maintenance, and administration. These carriers do not contain port circuit packs. Two J58890AP carriers are in the PN for high and critical reliability (duplicate processor) systems.	PN
Port Carrier. (optional). Contains port, service, tone/clock, and EI circuit packs.	PN or expanded PN

Description	Cabinet
Expansion Control Carrier. Contains extra port circuit packs, tone-clock, maintenance interface, and EI circuit packs.	expanded PN
Switch Node Carrier (optional). Contains SNI circuit packs composing the CSS.	PPN or expanded PN

Carrier Circuit-Pack Slots

There are 3 types of circuit pack slots in the carriers: Control, Port, and Service.

The purple-colored and white-colored circuit packs and slots are being replaced by circuit packs and slots labeled with gray and white rectangles, respectively. A label with a solid gray rectangle indicates a port slot/circuit pack. A label with an outlined white rectangle indicates a control slot/circuit pack.

- Port – colored purple or labeled with a gray rectangle and can accept any purple or gray-labeled circuit pack
- Control – colored white or labeled with an outlined white rectangle and can accept only a circuit pack assigned to that slot
- Service – colored purple or labeled with a gray rectangle; is a special type of circuit pack that does not have an I/O connector

Each port slot attaches to a 50-pin (25-pair) connector on the carrier's rear panel. A cable attaches to each connector and routes to the cross-connect field. Each slot containing a fiber optic interface circuit pack (EI or SNI) uses a fiber optic transceiver on the carrier's rear panel.

A current limiter board (CFY1B) plugs into the backplane of the control carrier located in the A position only. The board supplies emergency transfer logic, current-limited power, 5 VDC to trip the main circuit breaker in an over-temperature condition, and the ringing transfer relay. Terminators on the backplane terminate each end of the processor expansion bus.

The following apparatus blank faceplates (with widths) cover unused circuit pack slots in the carriers to maintain proper air flow:

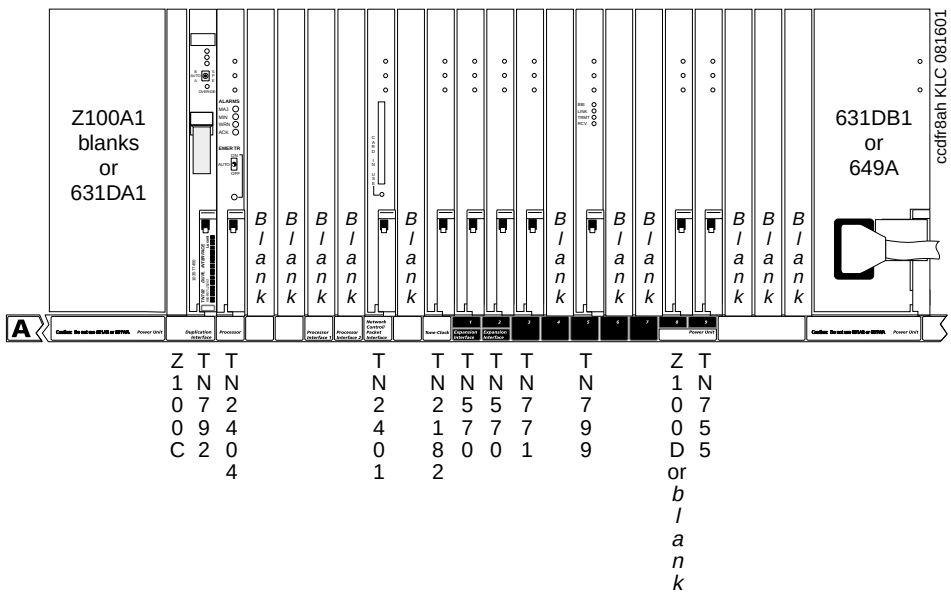
- Z100A1 (0.75 inches/1.9 cm)
- Z100C (0.5 inches/1.27 cm)
- Z100D (0.25 inches/0.64 cm)

NOTE:
In the following illustrations, a balanced ring generator (BRG) is shown below the power unit slot in certain carriers. This means the power unit slot can include a 50 Hz BRG when optioned for France.

Control Carrier

A representative J58890AH control carrier is shown below.

Control Carrier (Front)



This carrier has dedicated white-colored circuit pack slots that always contain specific control circuit packs. Dual-color slots may contain any port circuit pack or the designated white circuit packs (such as an EI or power unit). AC or DC power units supply power to the carrier.

The following table describes the Control Carrier connectors and functions.

Connector	Function
1 to 9 (A1 to A9)	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
AUX (Auxiliary)	Interface for alarms, attendant console power, emergency power transfer panels, and an internal modem (for remote maintenance)
Processor interface (standard reliability only)	Connects directly to the PI circuit pack. Provides a BX.25 protocol interface for communication between the circuit pack and external DCE equipment.
Data communications equipment	Connects the processor to CDR equipment, a system printer, or an external modem (for remote maintenance). This connector can be used with any reliability option.

Connector	Function
Terminal	Connects a management terminal to the processor in standard reliability systems. In critical reliability systems, connects a terminal to the processor in its control carrier.
Duplication option terminal	Used in high and critical reliability systems to connect an administration terminal to the active processor from the duplication interface slot position
P1	Provides position indicator of the carrier, power to fans, and access to alarm and control circuits
P2	Provides control signals to the carrier

Duplicated Control Carrier

The Duplicated Control Carrier (J58890AJ) has dedicated white circuit pack slots that always contain specific control circuit packs, gray/purple slots that can be populated with port circuit packs, and dual-colored slots (white and gray/purple) that may be populated with port circuit packs or designated white circuit packs (such as an expansion interface or power unit).

AC or DC power units located at each end of the Duplicated Control Carrier supply power to the carrier.

The following table describes the Duplicated Control Carrier connectors and functions.

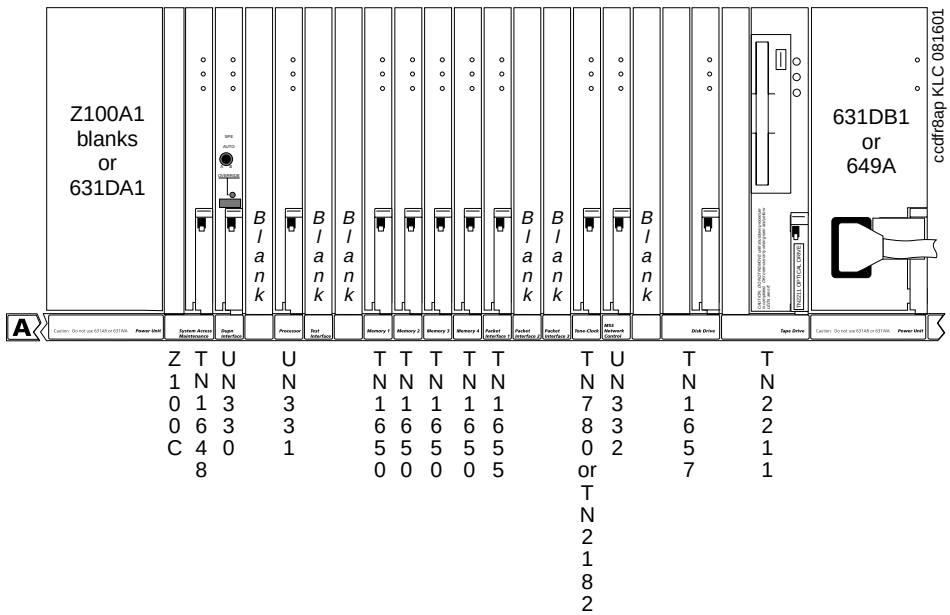
Connector	Function
1 to 9 (B1 to B9)	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
Terminal	Connects a management terminal to the processor in its Duplicated Control Carrier
P1	Provides position indicator of the carrier and access to alarm and control circuits

Processor Carrier

The Processor Carrier contains only dedicated slots for control circuit packs. It does not contain port circuit pack slots. See the following figure for a representative example.

AC or DC power units located at each end of the Processor Carrier supply the power to the carrier. The Processor Carrier always contains 4 memory circuit packs and 1 packet interface circuit pack.

Processor Carrier



The following table describes Processor Carrier connectors and functions.

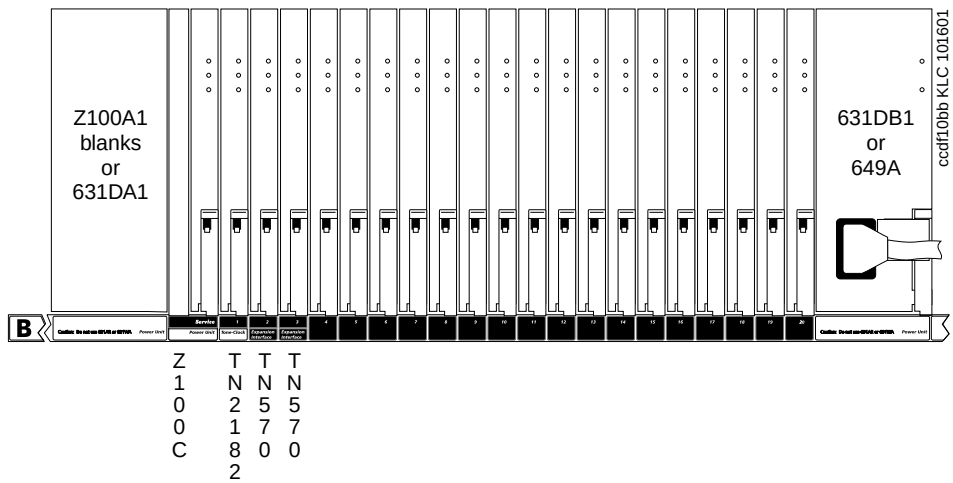
Connector	Function
Clock (stratum-3)	Provides an interface to a stratum-3 clock for digital frame timing. This is not a time-of-day clock
AUX (Auxiliary)	Provides an interface for customer alarms, attendant console power, emergency power-transfer panels, and an internal-modem interface for remote maintenance
Terminal, active	Connects a management terminal to the system access and maintenance (SYSAM) circuit pack in the active Processor Carrier
Terminal, standby	Used only in Duplicated Processors to connect a management terminal to the standby processor carrier
P1	Provides position indicator of the carrier and access to alarm and control circuits
P2	Provides control signals to the carrier

Port Carrier

A Port Carrier contains the following circuit packs

- Port slot locations 1 to 20 for the port circuit packs. Slot 1 may contain a tone-clock circuit pack when the port carrier is in the B position of an EPN cabinet in a critical reliability system. Slot 2 contains an optional EI or ATM Interface circuit pack for a critical reliability system.
- Power unit service slot in which a power unit circuit pack or service circuit pack can be installed.
- AC or DC power units located at each end of the carrier.

Port Carrier (Front)



The following table describes Processor Carrier connectors and functions.

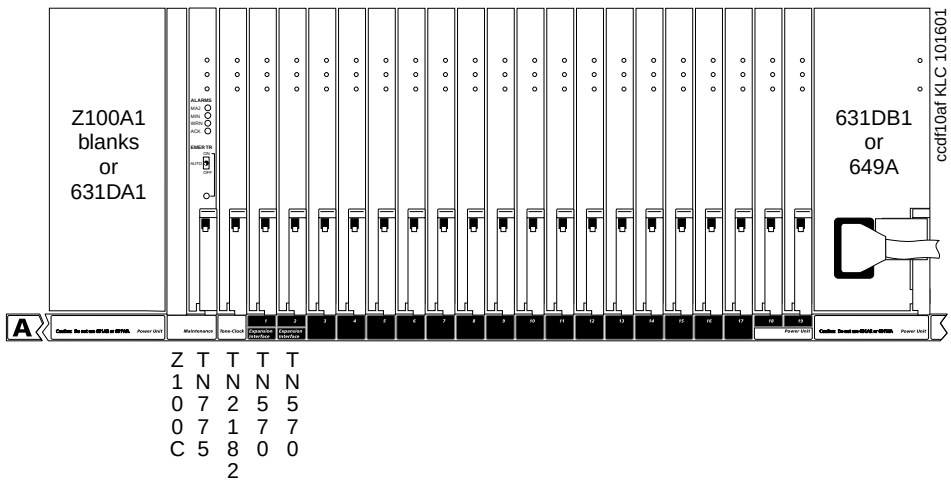
Connector	Function
1 to 20	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
P1	Provides position indicator of the carrier and access to alarm and control circuits

Expansion Control Carrier

The Expansion Control Carrier contains an EI or ATM Interface circuit pack in port slots 1 and 2. It is used in a fiber optic cabling path to another cabinet or the CSS in the same cabinet. These slots may contain optional port circuit packs.

The Expansion Control Carrier also contains port slots 3 to 19 and the AC or DC power units. The maintenance and tone-clock circuit packs are also shown. An optional neon power unit can be in slots 18 and 19.

Expansion Control Carrier (Front)



The following table describes Expansion Control Carrier connectors and functions.

Connector	Function
1 and 2 (A1 and A2)	Provides a fiber-optic cable interface to an expansion interface (EI) circuit pack in slot 1[1] or a copper cable interface for a DS1 Converter
1 to 19 (A1 to A19)	25-pair connectors provide interfaces between port circuit packs and the cross-connect field or fiber transceiver
AUX	Provides interfaces for customer alarms, attendant console power, and emergency power transfer panels
TERMINAL	Connects a management terminal to the maintenance circuit pack in an expansion control carrier
P1	Provides position indicator of the carrier and access to alarm and control circuits
P2	Connects ringing voltage from the ring generator to the carrier and produces control signals

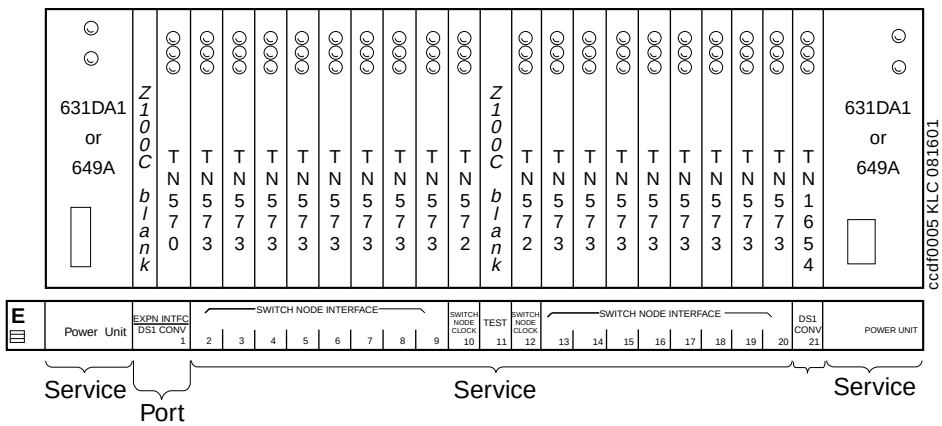
1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

Switch Node Carrier

TheSwitch Node Carrier (SN) (J58890SA) can contain one or two switch node clocks, up to 16 Switch Node Interface (SNI) circuit packs, 1 or 2 DS1 converter circuit packs, one EI circuit pack, and 2 AC or DC power units.

The AC or DC power units are located at each end of the SN. The SN can be used when connecting 3 or more EPNs.

Switch Node Carrier (Front)



The following table describes Switch Node Carrier connectors and functions.

Connector	Function
1 (E1)	EI connector for the cable between the EI circuit pack in slot 1 and the Switch Node Interface (SNI) circuit pack in slot 2 for a duplicated PN only. Also used for a DS1 Converter circuit pack in slot 1.
2-9 and 13-20 (E2-E19 and E13-E20)	SN ports that are fiber optic cabling interfaces to the SNI circuit packs and other circuit packs connected to SN ports or circuit packs in expansion PNs
21 (E21)	Interface to connect the DS1 Converter circuit pack to the cross-connect field and an SNI circuit pack
P1	Provides the position indicator of the SN carrier and provides access to alarm and control circuits

SCC1 Media Gateway

This section describes the SCC1 Media Gateway. Each SCC1 has vertical slots that hold circuit packs. A blank faceplate covers each unused slot. Refer to the figure for an example of a typical SCC1.

Typical SCC1

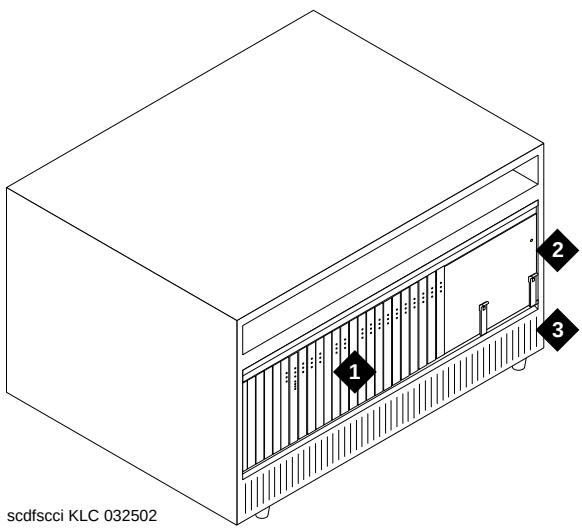


Figure notes

1 Circuit Packs

2 Power Converter

3 Air Circulation Vents

A maximum of 4 SCC1s can stack on top of each other. The cabinet positions are labeled A through D. The position of the basic control cabinet or expansion control cabinet is always labeled A. Additional port cabinet positions are labeled B, C, and D, sequentially. The Duplicated Control Cabinet is labeled B. Refer to the following figure for an example of a stack of SCC1s.

Typical SCC1 Stack

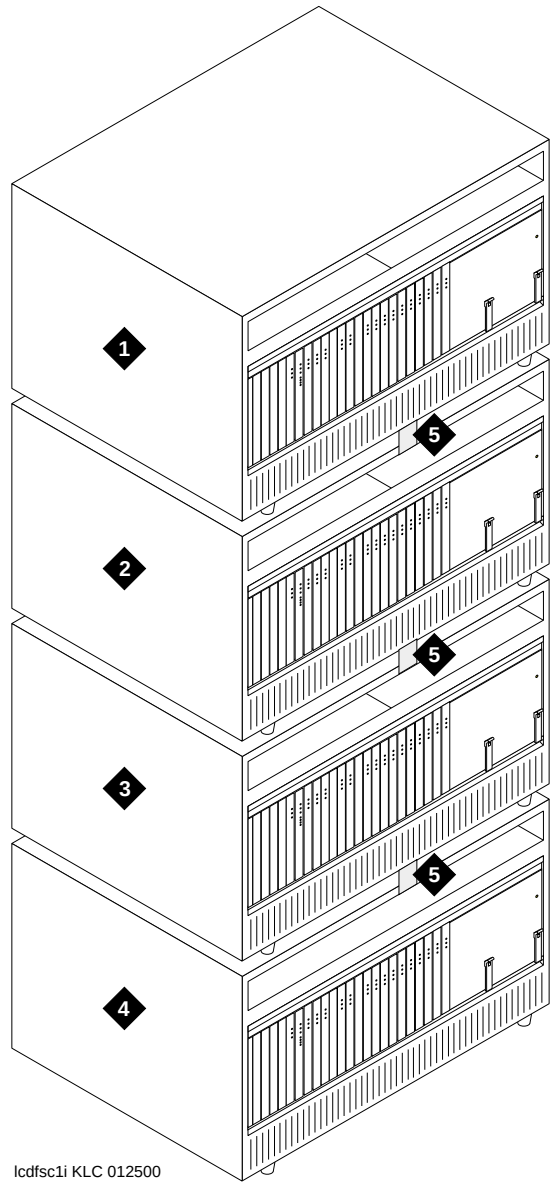


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Figure notes

1	Port Cabinet D Position	**4**	Basic Control Cabinet or Expansion Control Cabinet
2	Port Cabinet C Position	**5**	Cabinet Clips
3	Port Cabinet or Duplicated Control Cabinet B Position		

Each stack of SCC1s requires 1 basic or expansion control cabinet at the bottom of the stack. The maximum number of cabinet stacks or port networks is 3.

Cabinet clips connect the cabinets together. At the rear of the cabinets, a ground plate connects between cabinets for ground integrity.

Carrier Circuit Pack Slots

There are 3 primary types of circuit pack slots in the carriers:

The purple-colored and white-colored circuit packs and slots are being replaced by circuit packs and slots labeled with gray and white rectangles, respectively. A label with a solid gray rectangle indicates a port slot/circuit pack. A label with an outlined white rectangle indicates a control slot for a circuit pack.

- Port – colored purple or labeled with a gray rectangle and can accept any purple or gray-labeled circuit pack
- Control – colored white or labeled with an outlined white rectangle and can accept only a circuit pack assigned to that slot
- Service – colored purple or labeled with a gray rectangle; is a special type of circuit pack that does not have an I/O connector

Each port slot in a port carrier, an expansion control carrier, and a control carrier in R7si or later attaches to a 25-pair connector on the carrier's rear panel. A cable attaches to each connector and routes to the cross-connect field.

Blank faceplates cover empty carrier slots, as follows:

- 158J (4 in/9.2 cm) covers the area left of slot 1 in port cabinets
- 158P (0.75 in/1.9 cm) covers any unused slot.
- 158N (0.50 in/1.27 cm) is used with the LAN gateway in DEFINITY AUDIX R3 and CallVisor ASAI installations.
- 158G (0.25 in/0.63 cm) is used with the TN755 or TN2202.

In the following illustrations, a balanced ring generator (BRG) is shown below the power unit slot in certain carriers. This means the power unit slot can include a 50 Hz BRG when optioned for France.

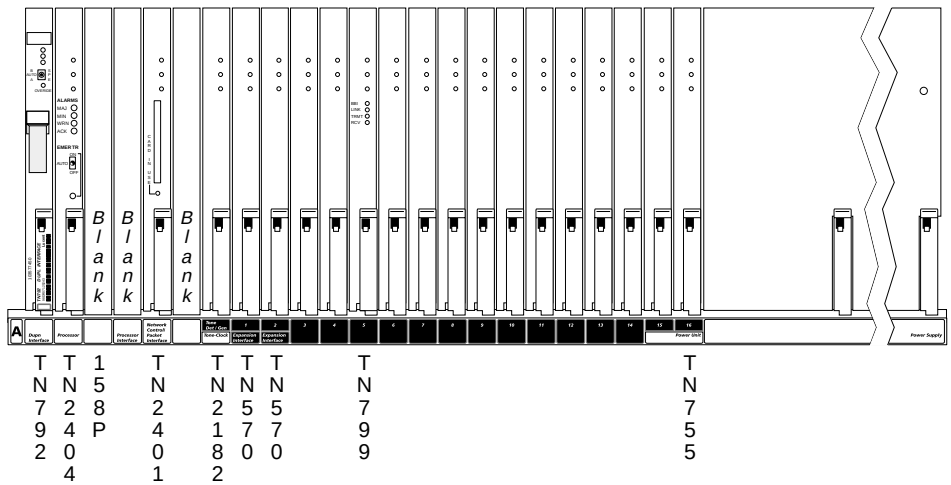
SCC1 Carriers

Basic Control Cabinet

The Basic Control Cabinet is in the PN only. It contains ports, a control complex to perform call processing, and an interface to an optional duplicated control cabinet. It also interfaces to the optional STRATUM 3 Clock.

It has dedicated white circuit pack slots that house specific control circuit packs. Dual-colored slots may contain any port circuit pack or the designated white circuit packs (such as an expansion interface or power unit). AC or DC power units supply power to the carrier. Refer to the following figure for an example of the the Basic Control Cabinet.

SCC1 Basic Control Cabinet (Front)



The following table lists Basic Control Carrier connectors and their function.

Connector	Function
1 to 16 (A1 to A16)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver
AUX (auxiliary)	Provides interfaces for customer alarms, attendant console power, emergency power-transfer panels, and an internal modem (for remote maintenance)
PI (processor interface)	Provides a BX.25 protocol interface for communication between the circuit pack and external DCE equipment. This connection is only used with the standard reliability option.
DCE	Connects the processor to CDR equipment, a system printer, or an external modem (for remote maintenance). This connector can be used with any reliability option.
TERM (terminal)	Connects an administration terminal to the processor circuit pack in standard reliability systems. The TERM connector always connects to the processor in its own carrier.
DOT (duplication option terminal)	Used in high reliability and critical reliability systems to connect an administration terminal to the active processor via the duplication interface slot. The DOT connector can be used to connect to the processor in another carrier.

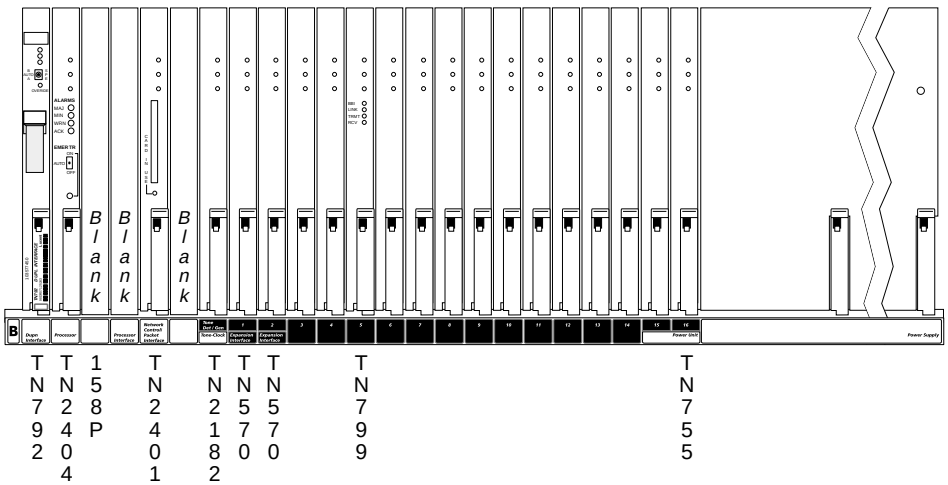
Duplicated Control Cabinet

The Duplicated Control Cabinet is optional and is only in the PN. This cabinet contains ports and a duplicated control complex.

The Duplicated Control Cabinet has dedicated white circuit pack slots for designated control circuit packs. The port circuit pack slots can contain any port circuit packs.

An AC or DC power supply, at the right side of the cabinet, supplies power to the cabinet. The cabinet contains a duplication interface circuit pack in slot DUPN INTFC. Refer to the figure that follows.

Duplicated Control Cabinet (Front)



The following table lists Duplicated Control Cabinet connectors and their function.

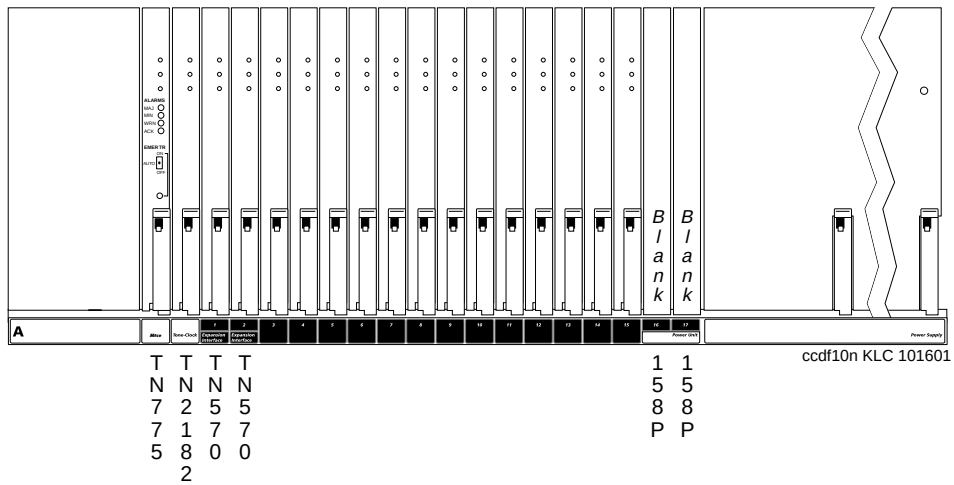
Connector	Function
01 to 16 (A01 to A16)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver
TERM (Terminal)	Connects an administration terminal to the processor circuit pack in the Duplicated Control Cabinet if the duplication interface circuit pack fails in the control carrier

Expansion Control Cabinet

The Expansion Control Cabinet contains ports, a tone-clock, an interface to a port cabinet, and a maintenance interface. It is only in an expansion PN.

The Expansion Control Cabinet is the first in an expansion PN stack of SCCs. It has optional port circuit packs in port slots 2 to 17. The AC or DC power supply, located at the right side of the cabinet, supplies power.

Expansion Control Cabinet



The following table lists Expansion Control Cabinet connectors and their function.

Connector	Function
1 (A1)	Provides a fiber-optic cable interface to an expansion interface (EI) circuit pack in slot 1[1] or a copper cable interface for a DS1 Converter
2 to 17 (A2 to A17)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver
AUX (auxiliary)	Provides interface for customer alarms, attendant console power, and emergency power transfer panels
TERM (terminal)	Connects an administration terminal to the maintenance circuit pack

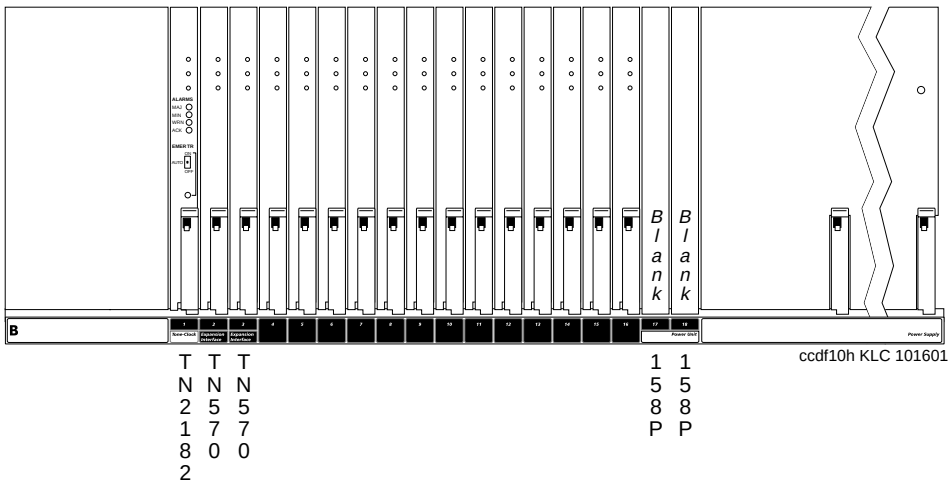
1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

Port Cabinet

The Port Cabinet is located in the PN and in expansion PNs. It contains ports and an interface to an Expansion Control Cabinet. The Port Cabinet has optional port circuit packs in port slots 1 to 18. In a critical-reliability system, these can include a tone-clock circuit pack in slot 1 and expansion interface circuit packs in slots 2 and 3. A neon power unit can be installed in slots 17 and 18.

The AC or DC power supply, located at the right side of the cabinet, supplies power to the cabinet. Refer to the following figure for an example of the Port Cabinet.

Port Cabinet



The following table lists Port Cabinet connectors and their function.

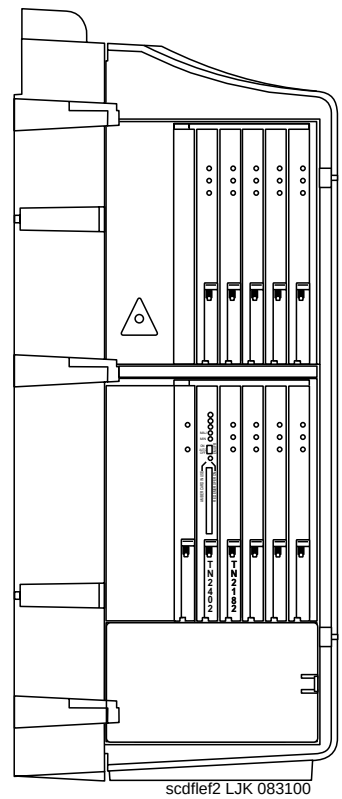
Connector	Function
2 to 3 (B2 to B3)	EI port that provides an interface for the fiber optic cable between an EI circuit pack in slot 2[1] or 3 in another PN or ATM system, or to an SNI
1 to 18 (B1 to B18)	25-pair connectors connect port circuit packs to the cross-connect field or a fiber transceiver

1. In systems using ATM-PNC, the fiber connectors for the OC-3/STM-1 interfaces to the ATM switches are located on the faceplates of the TN2305/TN2306 circuit packs.

CMC1 Media Gateway

The CMC1 Media Gateway is an economical, small-footprint alternative to a SCC1 Media Gateway. The CMC1 Media Gateway is used to support the DEFINITY CSI and S8100 Media Servers. It can be mounted on a wall or on the floor, and uses an AC-only power supply. The control carrier contains 2 control slots, one for the processor and the other for the tone clock. Slots 3 to 10 can contain optional port and service circuit packs. Refer to the following figure for an example of a CMC1.

CMC1 Left Side



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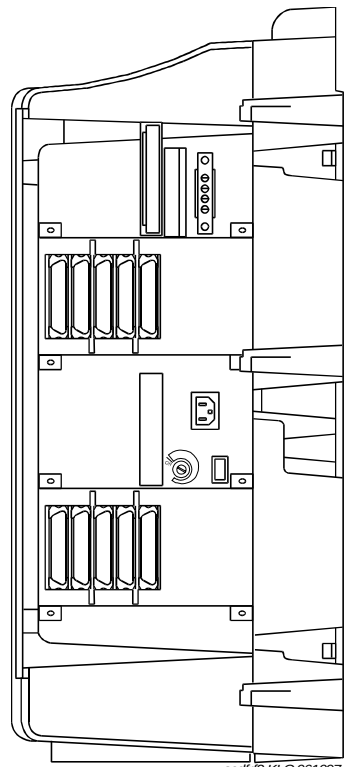
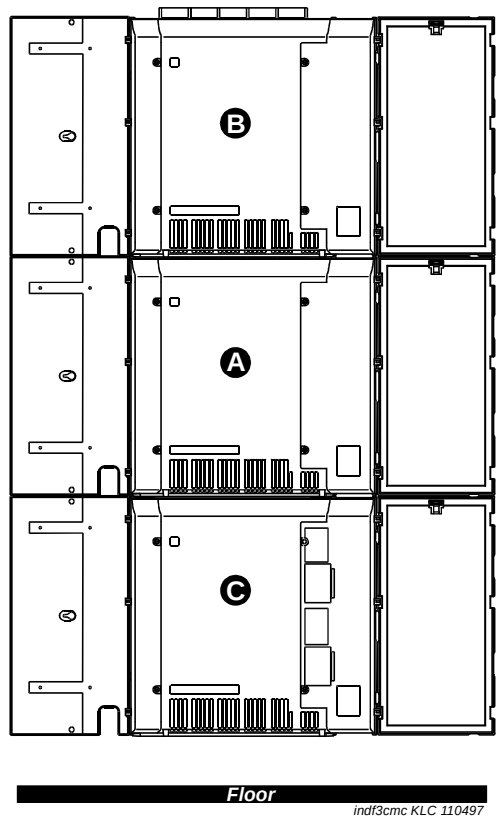
The connectors on the right side of the cabinet are shown in the following figure. One to ten 25-pair connectors interface between port circuit packs and the cross-connect field or a cable access panel.

CMC1 Right Side



Up to three CMC1s can be combined in a single installation. Port and service circuit packs fill all ten slots in the second and third cabinets. In a vertical lineup, the processor cabinet (A) installs in the middle position, the second cabinet (B) installs on the top, and the third cabinet (C) installs on the bottom. Cabinet A must have the processor pack and the tone clock. Cabinets B and C can have any packs in slots 1 and 2. Refer to the following figure for a typical vertical installation

Typical Vertical Installation Front View



Minimum Cabinet Configurations

Minimum cabinet configurations in MCC1 Media Gateways and in stacks of SCC1s serve as the foundations on which to build cabinets in direct-connect systems and Center Stage Switch (CSS)-connected systems.

The CMC1 cannot be duplicated but may be connected to 2 other CMC1 cabinets in a PN.

Site Requirements

Temperature and Humidity

The system equipment should be installed in a well-ventilated area. Maximum equipment performance is obtained at an ambient temperature between 40 to 110 degrees Fahrenheit (4 and 43 degrees Celsius) for continuous operation and between 40 and 120 degrees Fahrenheit (4 and 49 degrees Celsius) for short term operation. The system equipment can operate at the maximum short-term operational limits for a period not to exceed 72 consecutive hours or a total of more than 15 days in a year. The relative humidity range is 10 to 95 percent up to 84 degrees Fahrenheit (29 degrees Celsius). Above 84 degrees Fahrenheit (29 degrees Celsius), maximum relative humidity decreases from 95 percent down to 32 percent at 120 degrees Fahrenheit (49 degrees Celsius). Installations outside these limits may reduce system life or impede operation.

The following table correlates room temperature with allowable relative humidity. In addition these maximum ranges, the recommended temperature and humidity range for the S8700 Media Server for IP Connect and Multi-Connect is:

Recommended Temperature and Humidity

Temperature Range °F	Recommended Temperature Range °C	Humidity Range
65-85 °F	18-29 °C	20-60%
40 °F to 84 °F	4.4 to 28.8 °C	10% to 95%
86 °F	30.0 °C	10% to 89%
88 °F	31.1 °C	10% to 83%
90 °F	32.2 °C	10% to 78%
92 °F	33.3 °C	10% to 73%
94 °F	34.4 °C	10% to 69%
96 °F	35.6 °C	10% to 65%
98 °F	36.7 °C	10% to 61%
100 °F	37.8 °C	10% to 58%
102 °F	38.9 °C	10% to 54%
104 °F	40.0 °C	10% to 51%
106 °F	41.1 °C	10% to 48%
108 °F	42.2 °C	10% to 45%
110 °F	43.3 °C	10% to 43%
112 °F	44.4 °C	10% to 40%

Recommended Temperature and Humidity — *Continued*

Temperature Range °F	Recommended Temperature Range °C	Humidity Range
114 °F	45.6 °C	10% to 38%
116 °F	46.7 °C	10% to 36%
118 °F	47.8 °C	10% to 34%
120 °F	48.9 °C	10% to 32%

Altitude and Air Pressure/Purity for S8700 Media Server

The normal operating air pressure is the range from 9.4 to 15.2 psi (648 to 1048 millibars). For altitudes above 5,000 feet, reduce the maximum short-term temperature limit by 1 ^{O}F or each 1,000 feet of elevation above 5,000 feet. At 10,000 feet, for example, the maximum short-term temperature limit is 115 ^{O}F.

Air Purity

The cabinet should not be installed in an area where the air may be contaminated with any of the following:

- Excessive dust, lint, carbon particles, paper fiber contaminants, or metallic contaminants
- Corrosive gases, such as sulfur and chlorine

Above the following levels: 3

Contaminant	Average
Particulate matter	185 microgram/m 3
Nitrate in particulate matter	12 microgram/m 3
Total hydrocarbons equivalent to methane	10 ppm
Sulfur Dioxide	0.20 ppm
Oxides of Nitrogen	0.30 ppm
Total oxidants equivalent to ozone	0.05 ppm
Hydrogen Sulfide	0.10 ppm

NOTE:
It is recommended that the S8700 Media Server and related equipment not be collocated with copiers, printers, or plain paper faxes because of the fine dust they produce.

Floor and Spacing for S8700 Media Server with Multi-Connect

Components of the S8700 Media Processor with Multi-Connect are the S8700 Media Server, an Ethernet Switch and the UPS mounted in a 19 inch rack. Optional Media Gateways are also listed: MCC1, SCC1, and G700.

19 Inch Rack

The customer premise 19 inch data rack is used generically to house equipment not necessarily specified or provided by Avaya Inc. The footprint is 19 inches by 21 inches.

S8700 Media Server

The S8700 Media Server is designed for standard 19 inch data rack mounting. S8700 Media Servers weigh approximately 25 pounds.

UPS

AVAYA 700VA or 1500VA

For 28-410 minutes holdover: AVAYA 700VA UPS (700VA 120 V for US and Canada, 700 VA 230 V Online for International, 700 VA 100V and 200V for Japan).

Physical Characteristics

- Width: 17” (43.2 cm)
- Depth: 19” (48.2 cm)
- Height: 3.5” (8.9 cm)
- Weight: 34 lbs. (15 kg)

For 411-480 minutes holdover: AVAYA 1500VA UPS (1500VA 120 V for US and Canada, 1500 VA 230 V Online for International, 1500 VA 100V and 200V for Japan).

Physical Characteristics

- Width: 17” (43.2 cm)
- Depth: 24”
- Height: 3.5” (8.9 cm)
- Weight: 50 lbs.

Ethernet Switch

AVAYA Cajun

P133G2/P134G2 (with CCS PNC)

- (h, w, d): 2U (3.5”/88 mm) x 19” (482.6 mm) x 13.8" (350 mm)
- P133G2 Weight 11.4 lbs (5.2 kg)
- P134G2 13.2 lbs. (6.0 kg)

P333T/334T (with ATM PNC)

- (h, w, d): 2U (3.5”/88 mm) x 19” (482.6 mm) x 17.7" (450 mm)
- P333G2/P334G2 Weight 16.5 lbs (7.5 kg)

Media Gateways

The equipment room floor must meet the commercial floor loading code of at least 50 lbs. per square foot (242 kg per square meter). Floor plans typically allocate space around the front, ends, and rear (if necessary) of the cabinets, for maintenance access. Additional equipment room floor support may be required if the floor load is greater than 50 lbs. Per square foot (242 kg per square meter).

Media Gateway	Weight	Floor Loading
SCC1	125 lb. (56 kg.)	31 lb./sq. ft. (148.9 kg./m 2)
MCC1	200 - 800 lb. (90 - 363 kg.)	130 lb./sq. ft. (624.2 kg./m 2)
100-Amp Battery	400 lb. (181 kg.) max.	180 lb./sq. ft. (871.2 kg./m 2)
200-Amp Battery	815 lb. (370 kg.) max.	328 lb./sq. ft. (1587.5 kg./m 2)
300-Amp Battery	1480 lb. (671 kg.) max.	476 lb./sq. ft. (2303.8 kg./m 2)
400-Amp Battery	1580 lb. (717 kg.)	625 lb./sq. ft. (3025 kg./m 2)

Floor and Spacing for S8700 Media Server with IP Connect

Components of the S8700 Media Processor with IP Connect are the S8700 Media Processor, an Ethernet Switch, the G600 Media Gateway, modems, and the UPS mounted in a 19 inch rack.

19 Inch Rack

The customer premise 19 inch data rack is used generically to house equipment not necessarily specified or provided by Avaya Inc. The footprint is 19 inches by 21 inches. The rack should be sturdy enough to support two 34lb. UPSs or a pull-out rack provided by the customer.

S8700 Media Server

The S8700 Media Server is designed for standard 19" data rack mounting. S8700 Media Servers weigh approximately 25 pounds.

G600 Media Gateway

The G600 Media Gateway is designed for standard 19 inch data rack mounting. Each fully loaded G600 Media Gateway chassis weighs approximately 50 pounds.

UPS

AVAYA 700VA or 1500VA

For 28-410 minutes holdover: AVAYA 700VA UPS (700VA 120 V for US and Canada, 700 VA 230 V Online for International, 700 VA 100V and 200V for Japan).

Physical Characteristics

- Width: 17” (43.2 cm)
- Depth: 19” (48.2 cm)
- Height: 3.5” (8.9 cm)
- Weight: 34 lbs. (15 kg)

For 411-480 minutes holdover: AVAYA 1500VA UPS (1500VA 120 V for US and Canada, 1500 VA 230 V Online for International, 1500 VA 100V and 200V for Japan).

Physical Characteristics

- Width: 17” (43.2 cm)
- Depth: 24”
- Height: 3.5” (8.9 cm)
- Weight: 50 lbs.

Ethernet Switch

AVAYA Cajun

P133G2/P134G2 or P333T/334T

- Dimensions 2U (3.5”/88 mm) x 19”
- (h, w, d): (482.6 mm) x 13.8” (350 mm)
- Weight 11.4 lbs (5.2 kg)

Media Gateways

The equipment room floor must meet the commercial floor loading code of at least 50 lbs. per square foot (242 kg per square meter). Floor plans typically allocate space around the front, ends, and rear (if necessary) of the cabinets, for maintenance access. Additional equipment room floor support may be required if the floor load is greater than 50 lbs. Per square foot (242 kg per square meter).

Heat Dissipation and Power Required for S8700 Media Server

The amount of heat dissipated and power required by a S8700 Media Server for IP Connect configuration is described below. These estimates are highly dependent on the inclusion of terminals with the Media Gateways as well as the amount of calling (call rate) or terminal off-hook time.

NOTE:
Typical systems assume typical trunking, sparing, call rate and mixture of high and low function terminals (analog and 74xx digital sets). Values are given for the G600 Media Gateway (equipment room) and G600 Media Gateway dissipation plus terminal dissipation (analog and 74xx digital sets). Worst case systems assumes typical trunking and call rate, but no spare slots and all ports configured with high functioning terminals (the model used the worst case of 16 port analog circuit packs fully populated with analog sets).

Typical G600 Media Gateway Heat Dissipation Specifications

BTU Per Hour	# of G600 Media Gateways Only	G600 with Terminals
1	400	1000
2	900	2200
3	1400	3475
4	1900	4700
Worst Case	3200	5150

G700 Media Gateway Environmental Considerations

The equipment room floor must meet the commercial floor loading code of at least 50 pounds per square foot or 242 kilograms per square meter.

The following table lists the environmental considerations for the S8300 Media Server with the G700 Media Gateway..

Environmental Considerations

Consideration	Description
Heat Dissipation	The unit uses global AC, 100-240 VAC, 50/60 Hz, 1.5 to 4.9 A, which translates to 360 to 400 Watts. However, some is passed out the front, via -48VDC (up to 32 ports at 1.5 Watts each is 48 Watts).
Altitude and Air Pressure	The unit functions at altitudes of minus 197 feet to 10,000 feet. Air pressure is not specified.
Temperature and Humidity	Long term operation at +5 to +40 C at 5 to 85% humidity. Short term operation at -5 to 50 C, at 5 to 90% humidity, non-condensing.
Air Purity	Indoor environments suitable for continuous human occupancy.
Lightning	The user is protected under the UL codes against over voltage in the system. However, the system itself is susceptible to over voltage (i.e. lightning) depending on the configuration. The loss of service due to an over voltage condition can result in one or more of the following: Loss of terminal Loss of port Loss of Media Module Loss of power supply within the G700
Acoustic Noise Generated by Unit	50dBA max
Electromagnetic Compatibility Standards	S8300 Media Server conforms to the electromagnetic compatibility standards for the countries in which it operates.
European Union Standards	Approved to Safety Standard EN60950.
Air Flow with a Single Fan Failure	Nominal. In front of the backplane is 264 linear feet per minute average. If a fan fails in front of the backplane, 174 lfpm average, with a range from 42 to 340 1fpm.
Air Flow with the Power Supply Fan Failed	Minimal air flow at power supply if power supply fan fails.

G700 Media Gateway Power Requirements

This section describes G700 Media Gateway power source requirements.

The power supply complies with FCC Part 15, Subpart B Class B and EN55022 Class B requirements for conducted and radiated electromagnetic interferences (EMI). When used in a single to multiple G700 Media Gateway system, the power supply must allow the system to comply with Class B requirements with +6 dB of margin.

This power unit can be a single power supply or multiple modules sized and scalable for the load. Cajun has a power unit that meets the 802.3 AF standard and provides remote power for the phone. The power supply meets all applicable global safety, immunity, and emissions standards, and is verified by in-country testing.

Thermal Protection

Thermal protection shuts down the power supply if the internal temperature exceeds the power supply's maximum rated safe operating temperature. The minimum thermal shutdown point is determined at an ambient temperature of 50 degrees C at 10,000 ft elevation or 60 degrees C at sea level over all input and load conditions. Consider the effects of component tolerances when defining the shutdown point. This consideration ensures that the supply will not shut down at ambient temperatures less than those specified above, with forced air flowing from input to output at a nominal rate of 46 CFM (300 LFM).

Manual Reset

The power supply requires manual reset when it shuts down because of overvoltage or overheating. Reset is accomplished by recycling AC input power.

AC and Load Center Circuit Breakers

For AC power, each of the G700 Media Gateways has a detachable AC power cord that plugs into a wall socket or power strip on the rack. This circuit must be protected by a circuit breaker for the panel serving the outlet.

As a result, the G700 Media Gateway itself does not have circuit breakers or on/off switches. However, any customer AC load center must have circuit breakers protecting the power feeds to the G700 Media Gateways as required by electric codes.

AC Power Distribution

AC power distribution is plugged into an outlet or power strip and can be backed up by an optional UPS that customers can purchase from Avaya.

AC Grounding/Protective Grounds

The G700 Media Gatewaycontains a grounding screw on the back of the chassis. A reliable ground connection must be maintained whether connecting the G700 Media Gateway directly to the branch circuit or to a power distribution strip. The G700 Media Gateway also requires a cabinet ground connection directly to an approved ground.

S8100 Media Server with G600 Media Gateway Site Requirements

S8100 Media server can be mounted in a standard 19-inch (48 cm) data rack that has been properly pre-installed and secured as per the EIA 464 (or equivalent) standards. The G600 Media Gateway can be front mounted (as shipped) or mounted at its midpoint.

The customer is responsible for providing the rack and having it installed and secured prior to S8100 Media Server installation. This also applies to providing AC power to the rack. The technicians trained to install the S8100 Media Server do not typically have the tools or proper training for media gateway installation.

Installation requires 1 foot (30 cm) of clearance in the rear, and 18 inches (45 cm) of clearance in the front, which is consistent with the EIA 310 data rack standards. In a two-cabinet configuration, the dimensions of the TDM/LAN cable require that the B cabinet be mounted directly over the A cabinet (flush).

The S8100 Media Server should be installed in a well-ventilated area. Maximum equipment performance is achieved at an ambient temperature between 40 and 120 °F (4 and 49 °C) for a short-term operation (not more than 72 consecutive hours or 15 days in a year) and up to 110 °F (43 °C) for a continuous operation.

The relative humidity range is 10% to 95% at up to 84 °F (29 °C). Above this, maximum relative humidity decreases from 95% down to 32% at 120 °F (49 °C). Installations outside these limits may reduce system life or affect operation. The recommended temperature and humidity range is 65 to 85 °F (18 to 29 °C) at 20% to 60% relative humidity.

Cabinet Power Source Information

Cabinet Style and Power Distribution Unit	Power Sources	Power Input Receptacles
Rack Mount Cabinet. AC power supply (650A integrated power supply)	Single phase 120 VAC with neutral Single phase 240 VAC with neutral	120 VAC, 60 Hz NEMA 5-15R 240 VAC, 50 Hz IEC 320 Japan installations use country specific receptacles for 100 and 200 VAC, 50/60 Hz

[1]There is no integrated DC power supply. DC rectifiers can be used if desired; follow manufacturer’s instructions.

Circuit Breakers for AC-Powered Cabinets

Cabinet Type	Circuit Breaker Size
Rack Mount Cabinet (120 VAC) 60 Hz	15 A
Rack Mount Cabinet (240 VAC) 50 Hz	10 A

Air Information

Altitude and Air Pressure

At altitudes above 5,000 feet (1,525 meters), the maximum short-term temperature limit reduces by 1^{o} Fahrenheit for each 1,000 feet (305 meters) of elevation above 5,000 feet (1,525 meters). For example: at sea level, the maximum short-term temperature limit is 120^{o} F (49^{o} Celsius). At 10,000 feet (3,050 meters), the maximum short-term temperature limit is 115^{o} F (46^{o} C).

The normal operating air pressure range is: 9.4 to 15.2 psi (lbs. per sq. in.) (648 to 1,048 millibars).

Air Purity

The CMC1, SCC1, and MC1C contain an air filter to reduce particulates flowing through the equipment. Do not install the cabinet where the air may be contaminated by excessive dust, lint, carbon particles, paper fiber contaminants, or metallic contaminants. For example, do not install the cabinet near paper handling equipment such as copiers and high-speed printers, which introduce paper dust and print particles into the environment. Corrosive gases above the levels in the following table must be avoided.

Allowable Concentrations for Atmospheric Contaminants

Contaminant	Average Concentration Not to Exceed
All particulate matter	185 micrograms/cubic meter
Nitrate	12 micrograms/cubic meter
Total hydrocarbons equivalent to methane	10 ppm (parts per million)
Sulphur dioxide	0.20 ppm (parts per million)
Oxides of nitrogen	0.30 ppm (parts per million)
Total oxidants equivalent to ozone	0.05 ppm (parts per million)
Hydrogen sulfide	0.10 ppm (parts per million)

Floor Load Requirements

The equipment room floor must meet the commercial floor loading code of at least 50 lbs. per square foot (242 kg per square meter). Floor plans typically allocate space around the front, ends, and rear (if necessary) of the cabinets, for maintenance access. Additional equipment room floor support may be required if the floor load is greater than 50 lbs. per square foot (242 kg per square meter). See the table below.

Cabinet Weights and Floor Loadings

Type	Weight	Floor Loading	Remarks
Single-Carrier	125 lb. (56 kg)	31 lb./sq. ft. (148.9 kg/m^2)	
Multi-Carrier	200-800 lb. (90-363 kg)	130 lb./sq. ft. (624.2 kg/m^2)	Includes Auxiliary, Global AC and Global DC cabinets
100-Amp battery	400 lb. (181 kg) max.	180 lb./sq. ft. (871.2 kg/m^2)	
200-Amp battery	815lb. (370 kg) max.	328 lb./sq. ft. (1587.5 kg/m^2)	
300-Amp battery	1480 lb. (671 kg) max.	476 lb./sq. ft. (2303.8 kg/m^2)	
400-Amp battery	1580 lb. (717 kg) max.	625 lb./sq. ft. (3025 kg/m^2)	

Power Requirements

Cabinet Power Requirements

This section describes cabinet AC- and DC-power source requirements.

Global AC MCC Power Supply

The Global MCC (GMCC) power supply is a global offering that applies to both US and International systems. It replaces the existing AC MCC1 cabinet configuration used in the US, thereby reducing the number of power supplies and distribution units associated with the MCC1 platform.

The GMCC automatically accepts 200- to 240-volt AC power at 50 or 60 Hz. It complies with all emissions and safety requirements for customer applications worldwide. The power system consists of NP850 Rectifiers, 649A Power Supplies, one Battery Interface/Alarm Unit, battery connections, and alarm outputs. The NP850 Rectifiers are cabinet-level power units located in the power distribution unit at the bottom of the MCC1. The 649A Power Supplies are DC/DC converters that provide carrier-level power. The cabinet input cord is NEMA 6-30P in the US.

The GMCC power architecture offers both short- and long-term power backup. Batteries internal to the MCC1 provide short-term backup. External battery cabinets provide long-term backup. Because of its long-term power backup capabilities, the GMCC significantly reduces or eliminates the need for UPS and DC Battery Plants for most customer applications.

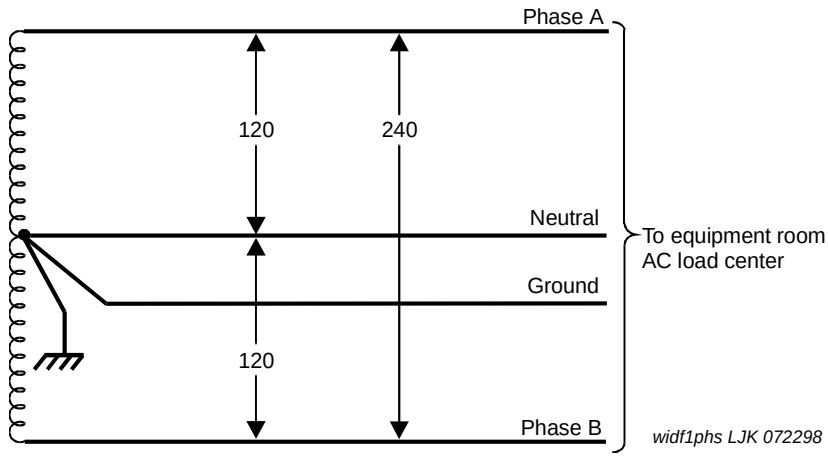
AC Power

Power feeders from a dedicated AC-power source (usually located outside the building) connect to an AC-load center. These feeders do not power other equipment. The AC-load center distributes the power to receptacles. The power cord from the AC-power distribution unit in each multicarrier cabinet and AC-power supply in each single-carrier cabinet plugs into a receptacle.

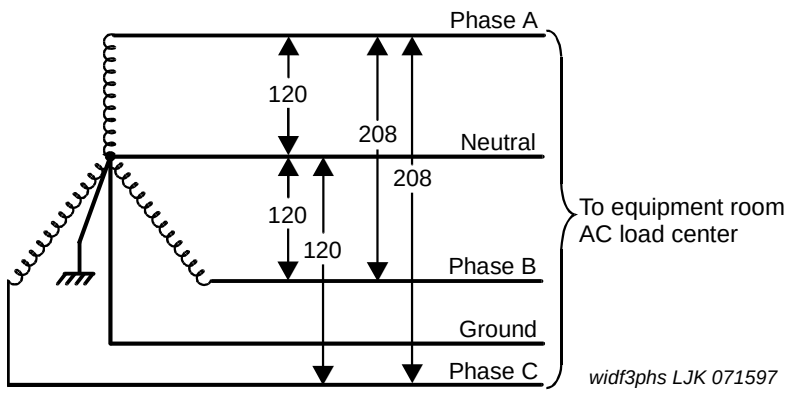
60 Hz Power Sources in G3 Systems

Each of the following power sources can supply 60-Hz power to the AC load in R7 and later systems. Refer to the following figures.

Single-Phase, 120/240 VAC, 60 Hz Source



Three-Phase, 120/208 VAC, 60 Hz Source

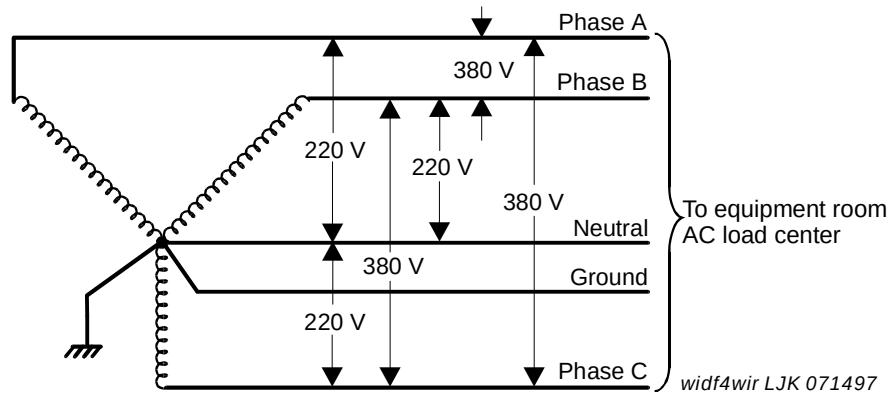


50 Hz Power Sources in G3 Systems

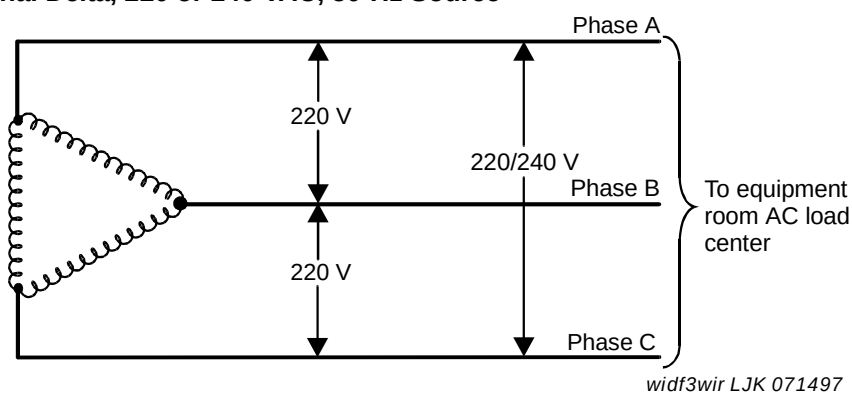
Either of the following power sources can supply 50-Hz power to the AC- load in R7 and later systems. Refer to the following figures.

NOTE:
The type of power for a MCC1 is shown on the cabinet's rear door, a SCC1 is shown on the cabinet's rear cover, and a CMC1 is shown on the right door.

International, Three Phase, 220/380 VAC, 50-Hz Source



International Delta, 220 or 240 VAC, 50-Hz Source



The following table lists the AC-power sources that can supply power to an AC- load in a cabinet. A NEMA receptacle (or equivalent) connects to the wires from the unit. The AC power cord from the power input of each unit plugs into a receptacle.

Contact your Avaya representative for the list number of each power source application.

Cabinet AC Power Sources

Cabinet Style and Power Distribution Unit	Power Sources	Power Input Receptacles
CMC AC power supply (650A)	Single phase 120 VAC with neutral	120 VAC, 60 Hz NEMA 5-15R
	Single phase 240 VAC with neutral	240 VAC, 50 Hz IEC 320 Japan installs use country specific receptacles for 100 and 200 VAC, 50/60 Hz
MCC1 AC power distribution (J58890CE-1 and J58890CE-2)	Single phase 120 VAC with neutral	120 VAC, 60 Hz NEMA 5-50R
	Single phase 240 VAC with neutral, or single phase of 3-phase, 208 VAC with neutral	208/240 VAC, 60 Hz NEMA L14-30R
MCC1 AC Power distribution (J58890CH-1)	Single phase 176-264 VAC	200-240 Volts, 50-60 Hz NEMA L6-30R. Installations outside the United States require a receptacle suitable for use in the country of installation.
SCC1 AC power supply (1217A)	Single phase 120 VAC with neutral	120 VAC, 60 Hz NEMA 5-20R or 5-15R
	Single phase of 220 VAC or Single phase of 240 VAC	220/240 VAC at country-specific receptacle

DC Power

DC-powered cabinets containing a J58890CF power distribution unit require a -42.5 to -56 VDC source at up to 75 A.

AC and DC Load Center Circuit Breakers

The circuit breaker sizes for all AC- and DC-powered cabinets are listed in the following two tables.

Circuit Breakers for AC-Powered Cabinets

Cabinet Type	Circuit Breaker Size
CMC1 (120 VAC) 60 Hz	15 A
CMC1 (240 VAC) 50 Hz	10 A
MCC1 (120 VAC) 60 Hz	50 A
MCC1 (208 VAC) 60 Hz	30 A
MCC1 (240 VAC) 60 Hz	30 A
MCC1 (200-240 VAC) 50-60 Hz	30 A
SCC1 (120 VAC)	15 or 20 A
Auxiliary cabinet (120 VAC)	20 A

Circuit Breakers for DC-Powered Cabinets

Cabinet Type (–48 VDC)	Circuit Breaker Size
MCC1	75 A
SCC1	25 A
Auxiliary cabinet	20 A

MCC1 Power System

These power systems consist of an AC- or DC-power distribution unit in the bottom of each cabinet and cabling to distribute output voltages to power unit circuit packs in the carriers. These power systems also consist of power-converter circuit packs in the carriers supplying DC- power to the circuit pack slots. The following table lists the input and output voltages of power unit circuit packs in the carriers of MCC1s.

Power Units in MCC

Model	Inputs			Outputs		
	120 VAC	144V DC	-48 VDC	+5 VDC 60 A	–5 VDC 6 A	–48 VDC 8 A
AC 631DA1	yes	yes	no	yes	no	no
AC 631DB1	yes	yes	no	no	yes	yes
DC 644A	no	no	yes	yes	no	no
DC 645B	no	no	yes	no	yes	yes
DC 649A	no	no	yes	yes	yes	10 A

AC and DC Power Distribution

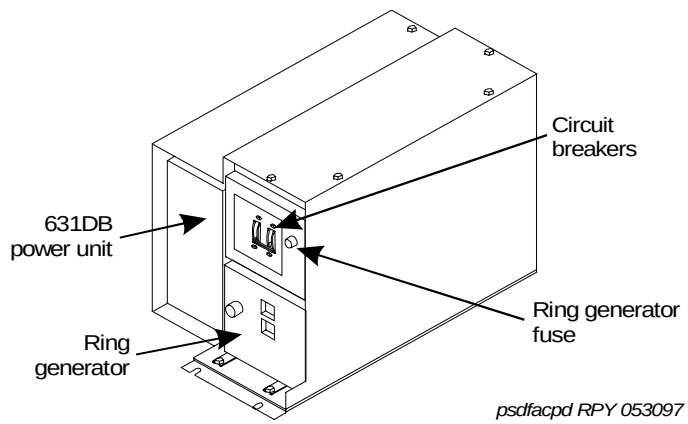
A typical AC-power distribution unit for an MCC1 contains the circuit breakers, ring generator, optional batteries, and optional battery charger. The power distribution cables carry 120 VAC during normal operation and 144 VDC from optional batteries if AC power fails. Another cable connects 120 VAC to the battery charger.

DC-powered cabinets require a -42.5 to -56 VDC source at up to 75 A.

AC Power Distribution

The following figure shows an AC Power Distribution Unit and Battery Charger (J58890CE-2 List 15 or later). This unit sits at the bottom of some MCC1s.

AC Power Distribution Unit (J58890CE-2) (Front)

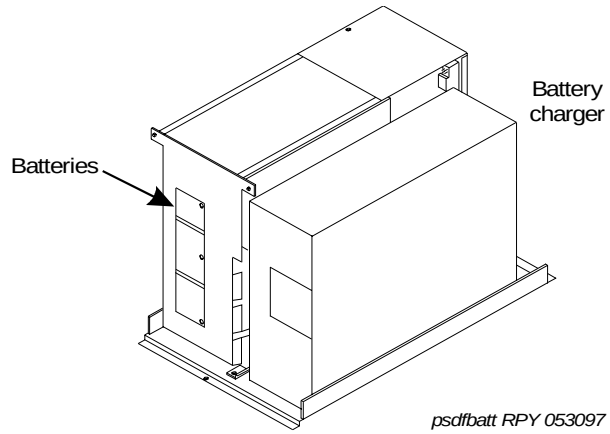


The AC-Power Distribution Unit contains the following additional components not shown in the figure:

- Electromagnetic Interference (EMI) filter
- AC input fuse
- 5 circuit breakers (1 for each carrier)
- 20-amp fuses
- Signal connector
- **-**48 VDC fan power

The optional battery charger (List 11) sits at the bottom of some MCC1s.

Battery Charger (Optional Part of J58890CE-2) (Front)



The battery charger is used only without an Uninterruptible Power Supply (UPS). The charger contains:

- Three 48-VDC batteries for backup power to the cabinet
- A DC-power relay to switch the batteries into the power circuit if a main power failure is detected

Circuit Breaker

The circuit breaker protects the AC input power to the cabinet and serves as the main AC input disconnect switch. The circuit breaker has 2 poles for 120 VAC or 3 poles for 208/240 VAC. If a problem develops, the circuit breaker automatically trips (opens) and removes the AC power input.

48-VDC Batteries

The 3 series-connected 48-VDC batteries produce a nominal 144 VDC, fused at 20 A. The batteries trickle-charge from the battery charger.

Battery Charger

When AC power restores after an outage, the battery charger converts a 120 VAC input to a DC voltage that recharges the batteries (usually within 24 hours).

DC Power Relay

This relay disconnects the batteries from a system when using AC power. The relay also disconnects the batteries if power fails for more than 10 minutes in a standard reliability system, 5 minutes in high and critical reliability systems, and 10 minutes in an expansion port network (EPN). This protects the batteries from over-discharging.

Electromagnetic Interference (EMI) Filters

The EMI filters suppress noise voltage on the AC input line to the unit.

Ring Generator

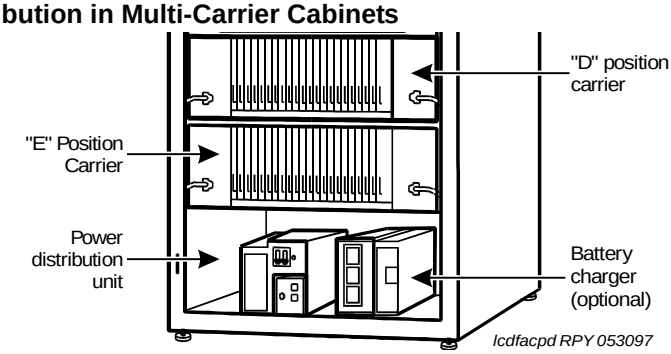
The ring generator converts the -48 VDC input to a 67 VAC to 100 VAC, 20 Hz or 25 Hz ringing voltage. The analog line circuit packs use this AC voltage output to ring voice terminals. The AC outputs route from the ring generator to port carriers, expansion control carriers, and control carriers.

Fuses

20-Amp fuses protect the power on each cable going from the AC- Power Distribution Unit to power converters in the carriers.

The following figure shows AC power distribution in some MCC1s. The DC-power distribution cables are on both sides of the cabinet. These cables supply power to each of the carriers. The optional battery charger is at the right side of the Power Distribution Unit.

AC Power Distribution in Multi-Carrier Cabinets



Power Backup

If AC power fails, three 48-VDC batteries power the system for 10 seconds in a PPN cabinet, for 15 seconds in an EPN cabinet, and for 10 minutes in the control carrier in a standard reliability system. The batteries also supply system power for 5 minutes in the control carrier in high and critical reliability systems, and for 10 minutes in the expansion control carrier in the A position of an EPN cabinet (r systems only).

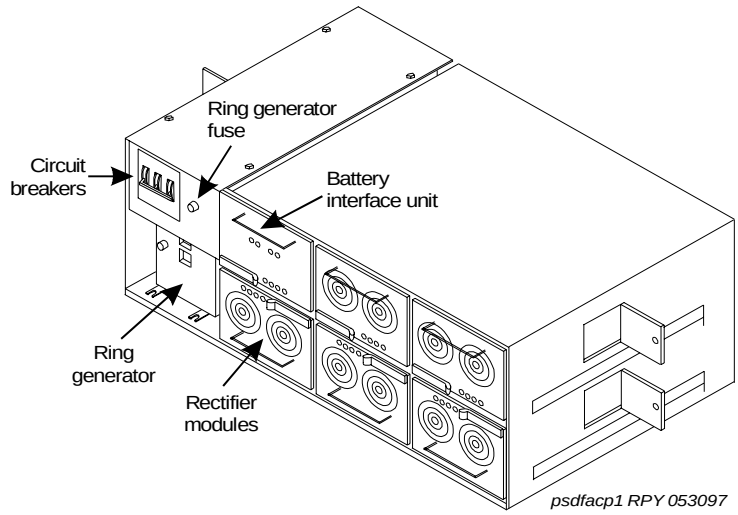
Uninterruptible Power Supply (UPS)

An external, Uninterruptible Power Supply (UPS) provides a longer backup time than holdover batteries (holdover times vary from less than 10 minutes to up to 8 hours) and can replace the batteries and battery charger. The unit connects from the AC-power source to a cabinet's AC-power cord. If AC power fails, the unit supplies its own AC power to the cabinet.

AC Power Distribution Unit (J58890CH-1)

The following figure shows a typical AC Power Distribution Unit used in some Multi-Carrier Cabinets. The unit sits at the bottom of the cabinet.

AC Power Distribution Unit (J58890CH-1) (Front)



Power Backup

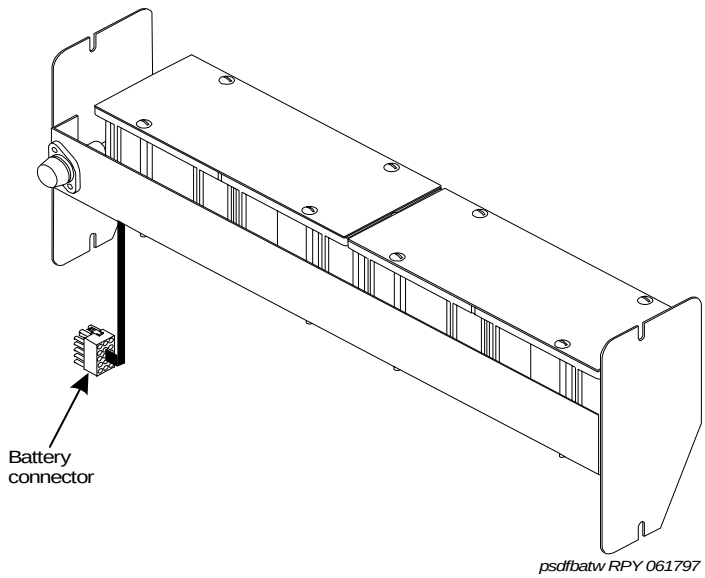
There are 2 types of battery assemblies used for power backup: small and large. The small batteries are typically located at the center rear of a Multi-Carrier Cabinet. The large batteries are typically located inside the battery cabinet.

Small Batteries

The small battery is an 8 AH (Amp-hour) battery fused for short circuit protection and is charged by the J58890CH-1. The batteries also contain a thermal sensor that changes the charging voltage depending on battery temperature. Refer to the following figure.

The small batteries provide short-term battery holdover. If AC power fails, 48 VDC batteries power the system for 10 seconds in a PPN cabinet, for 15 seconds in an EPN cabinet, and for 10 minutes in the control carrier in a standard reliability system. The batteries also provide system power for 5 minutes in the control carrier in high and critical reliability systems, and for 10 minutes in the expansion control carrier in the A position of an EPN cabinet (r systems only).

Small Battery Assembly



Large Batteries

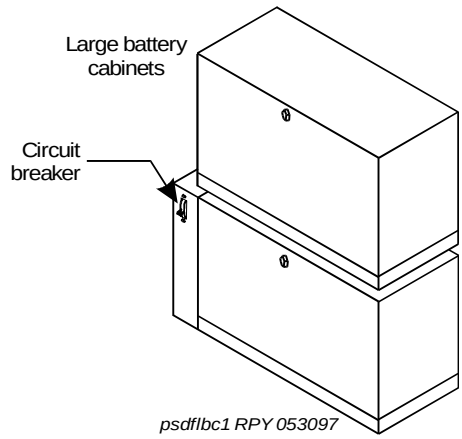
The large batteries can supply holdover times of 2 to 8 hours, depending on the number of batteries. When using large battery holdover, 1 battery cabinet is required per system. The 24-cell battery cabinet must have float voltage of 54.2 VDC. The 23-cell battery cabinet must have a float voltage of 51.75 VDC. The batteries are circuit breaker protected and are charged by the J58890CH-1.

The batteries also contain a thermal sensor that changes the charging voltage, depending on the battery temperature. The batteries provide extended holdover. Battery holdover and recharge times for a typical 2500-Watt load are shown in the following table. A typical large battery cabinet, (200 A) is shown in the figure that follows.

Battery Holdover and Recharge Times

Cabinet Type	Holdover Time	Recharge Time
100 A	2 hours	7 hours
200 A	4 hours	13 hours
300 A	6 hours	20 hours
400 A	8 hours	26 hours

Typical Large Battery Cabinets



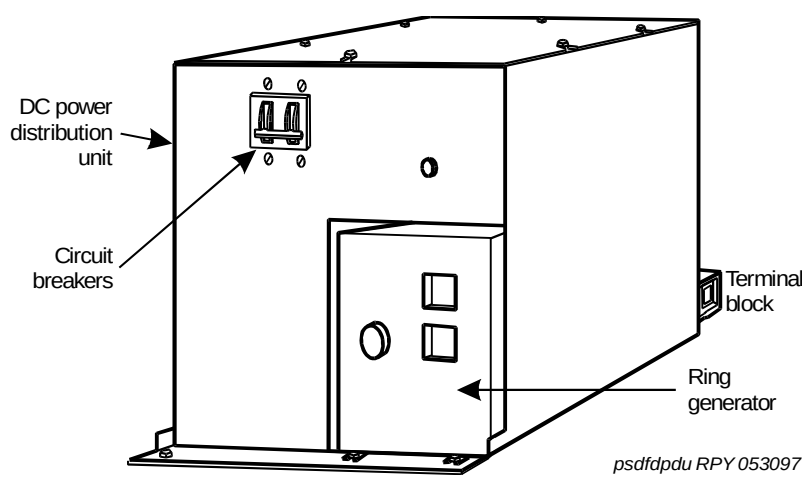
DC Power Distribution

The typical distribution system has a DC power converter and cables to provide the power to the system circuit packs.

DC Power Distribution Unit (J58890CF-2)

The following figure shows a Power Distribution Unit in some DC-powered MCC1s. The unit sits at the bottom of the cabinet and contains the ring generator, 20-amp circuit breakers, terminal blocks, and system fan power.

DC Power Distribution Unit (J58890CF-2) (Front)



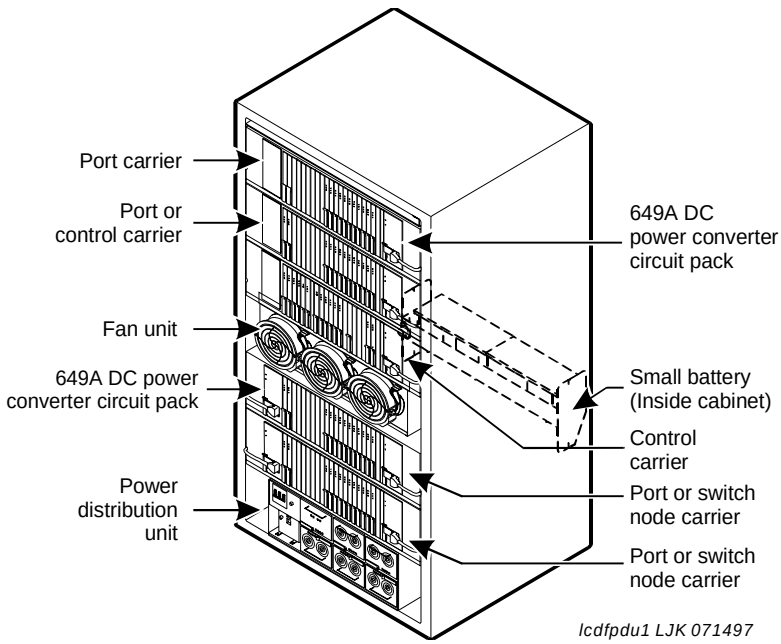
Ground Isolation

Each peripheral connecting to a switch, via the Asynchronous Electronic Industries Association (EIA) RS-232 interface, requires either a 105C, a 105D or a 116A Isolator Interface. The interface isolates ground between the system and external adjuncts.

The Isolator Interface is behind a PPN control carrier or behind an EPN expansion control carrier. The 105C, 105D, or the 116A installs at the RS-232 interface between the peripheral equipment and the interface connector.

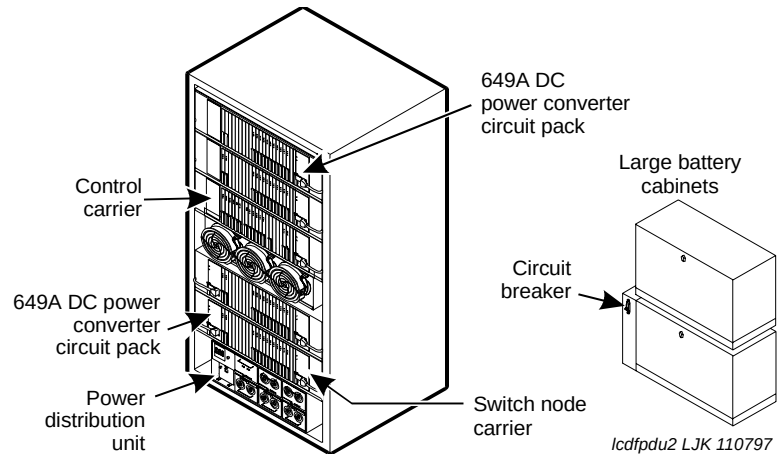
The following figure shows the power distribution in some MCC1s with short term battery holdover (small battery). In r systems, the power distribution cables are on the right hand side of the cabinet only. This is because the 649A DC power converter circuit pack is located on the right side of the carrier. Switch node (SN) carriers require two 649As and 2 cables.

Power Distribution in MCC1



The following figure shows the DC-power distribution in some MCC1s with extended battery backup (large batteries). In r systems, the power distribution cables are on the right hand side of the cabinet only. This is because the 649A DC power converter circuit pack is located on the right side of the carrier. Switch node (SN) carriers require two 649As and two cables.

DC Power Distribution in MCC1s



DC Power Converter (649A)

The 649A converts the –48 VDC from the Power Distribution Unit to outputs of –48 VDC at 10 A, +5 VDC at 60 A, and –5 VDC at 6 A. These outputs distribute power to circuit pack slots in the carriers.

AC and DC Grounding

Approved Grounds

An approved ground is the closest acceptable medium for grounding the building entrance protector, the entrance cable shield, or single-point ground of the system.

If more than one type of approved ground is available on the premises, bond the grounds together as specified in Section 250-81 of the National Electrical Code, or the applicable electrical code in the country where the equipment is installed.

Protective Grounds

Grounded Building Steel — The metal frame of the building where it is grounded by one of the following: acceptable metallic water pipe, concrete encased ground, or a ground ring.

Acceptable Water Pipe — Underground water pipe, at least 1/2 inch (1.3 cm) in diameter, needs to be in direct contact with the earth for at least 10 feet (3 meters). The pipe must be electrically continuous (or made electrically continuous by bonding around insulated joints, plastic pipe, or plastic water meters), to the point where the protector ground wire is connected.

A metallic underground water pipe must be supplemented by the metal frame of the building, a concrete encased ground, or a ground ring.

If these grounds are not available, the water pipe ground can be supplemented by one of the following types of grounds:

- Other local metal underground systems or structures — Local underground structures such as tanks and piping systems
- Rod and pipe electrodes — A 5/8 inch (1.6 cm) solid rod or 3/4 inch (1.9 cm) conduit or pipe electrode driven to a minimum depth of 8 feet (2.5 meters)
- Plate electrodes — Must have a minimum of 2 square feet (0.18 square meters) of metallic surface exposed to the exterior soil

Concrete Encased Ground — An electrode encased by at least 2 in. (5 cm) of concrete and located within and near the bottom of a concrete foundation or footing in direct contact with the earth. The electrode must be at least 20 feet (6 meters) of one or more steel reinforcing bars or rods 1/2 inch (1.3 cm) in diameter, or at least 20 feet (6 meters) of bare, solid copper, 4 AWG (26 mm^2) wire.

Ground Ring — A buried ground that encircles a building or structure at a depth of at least 2.5 feet (0.8 meter) below the earth's surface. The ground ring must be at least 20 feet (6 meters) of 2AWG (35 mm^2), bare, copper wire.

Approved Floor Grounds

WARNING:
If the approved ground or approved floor ground can only be accessed inside a dedicated power equipment room, then connections to this ground should be made by a licensed electrician.

Approved floor grounds are those grounds on each floor of a high-rise building suitable for connection to the ground terminal in the riser closet and to the cabinet equipment single-point ground terminal. Approved floor grounds may include the following:

- Building steel
- The grounding conductor for the secondary side of the power transformer feeding the floor
- Metallic water pipes
- Power feed metallic conduit supplying panel boards on the floor
- A point specifically provided in the building design for grounding

NOTE:
Electrically connect all protective grounds together to form a single grounding electrode system.

Coupled Bonding Conductor

When using Coupled Bonding Conductor (CBC) grounding in an AC- powered cabinet, maintain a minimum 1 ft. (0.3 m) spacing between the CBC and other power and ground leads.

In AC-powered systems, locate the system single-point ground terminal block on the AC load or AC protector cabinet.

SCC1 Power Systems

Each SCC1 has 1 AC or 1 DC power supply that distributes DC-power and AC ringing voltage to the circuit pack slots in the cabinet.

AC Power Supply (1217A)

In an AC-powered cabinet, a single, plug-in, multi-output AC power supply is in the power supply slot. A power cord, with a 3-prong plug on one end and an appliance connector on the other end, connects the supply to a dedicated AC power source.

The 1217A is a global power unit for SCC1s. It has a wide input voltage operating range of 90 to 264 VAC and a 50/60 Hz autoranging input, multi-output power supply that provides regulated DC output. The 1217A also has a selectable 20/25 Hz AC ringer.

The inputs to the power supply can be (depending on list version):

- 120 VAC, 60-Hz, 15-Amp to 20-Amp; 3 wires in the power cord: 1 hot wire, 1 neutral wire, and 1 ground wire
- 220 VAC or 240 VAC, 50-Hz, 10-Amp; 3 wires in the power cord: 1 hot wire, 1 neutral wire, and 1 ground wire

The AC power supply produces the following DC outputs: +5 VDC, –5 VDC, –48 VDC, +12 VDC, and a battery-charging voltage. The DC outputs distribute power on the cabinet backplane to the circuit pack slots. Additionally, the -48 VDC output current capacity has been increased from 6.85 amps to 8.25 amps. A 50 amp load inrush requirement has been added to the -48 VDC output

A holdover circuit in the power supply allows a system to operate normally during AC power interruptions. If AC input power fails, reserve batteries supply power to the memory and processor circuit packs and fans for 2 minutes. All port circuit packs are inactive during this time. The power supply contains a battery charger to charge the holdover batteries.

DC Power Supply (676C)

In a DC-powered SCC, a single, plug-in multi-output DC power supply is in the power supply slots.

The 676C DC power supply has a wide input voltage operating range of -42 to -60 VDC at up to 22A. The 676C produces the following outputs: +5.1 VDC at 0 to 55A, -5.1 VDC at 0 to 5.5A, +12 VDC at 0 to 2A (surge to 2.8A for 350 ms), -48 VDC at 0 to 8.25A. The outputs distribute power on the cabinet backplane to the slots for the circuit packs. The AC ringing voltage output value and frequency depend on the country of use. The power supply has circuit breakers and EMI filtering.

DC Power Distribution Unit (J58890CG)

The J58890CG is used with SCC1s. Individual DC output connectors can power up to 4 SCC1s. Each output connector is separately fused at 25 A (fuses are inside the unit). The input to the DC distribution unit is from the DC power cabinet.

The J58890CG is required when the distance between the DC power cabinet and the cabinet stack is greater than 30 feet (9 m).

Enhanced DC Rectifier Cabinet (J58890R)

The J58890R is used with SCC1s. Each rectifier assembly in the DC rectifier cabinet can supply up to 50 A of DC current. A minimum of two rectifiers install in each DC cabinet to supply a total of 100 A. A third rectifier assembly is used as a backup only.

Each SCC1 can draw up to 15 A. Up to 3 DC cabinets can be stacked to supply power to single-carrier cabinets stacks.

Each output connector is separately fused at 25 A (fuses are inside each DC rectifier assembly).

NOTE:
A J58890CG DC Power Distribution Unit is required if the distance between the DC cabinet and the cabinet stack is greater than 30 feet (9 m).

CMC1 AC Power Supply (650A)

In the CMC1, a power cord, with a 3-prong plug on one end and an appliance connector on the other end, connects the supply to a dedicated AC power source. The power supply is a global power factor corrected AC/DC converter providing multiple DC outputs and AC ring outputs. It is auto ranging 85 to 264 VAC, 47 to 63 Hz, at 330 Watts, 4.5 A (100-120 VAC) or 2.3 A (200-240 VAC) at 500 VA.

The inputs to the power supply can be:

- 120 VAC, 50-Hz to 60-Hz, 6-Amp; 3 wires in the power cord: 1 hot wire, 1 neutral wire, and 1 ground wire
- 220 VAC or 240 VAC, 50-Hz to 60-Hz, 3-Amp; 3 wires in the power cord: 1 hot wire, 1 neutral wire, and 1 ground wire

The AC power supply produces the following outputs: +5 VDC, –5 VDC, and –48 VDC. The outputs distribute power on the cabinet backplane to the circuit pack slots. The AC ringing voltage output value and frequency depend on the country of use. The 650A also supplies power for neon message-waiting lamps (150 VDC). The power supply has EMI filtering.

CMC1 Uninterruptible Power Supply (UPS)

The UPS provides surge protection for all connected cabinets.

- Connect the UPS to an electrical outlet capable of handling the power requirements of all cabinets. To calculate the number of amps drawn, use the following formulas.
    - a. 100-200 VAC, multiply 3.5 amps times the number of cabinets.
    - b. 200-240 VAC, multiply 1.8 amps times the number of cabinets.
- Cabinet A (control carrier) is always connected to an *unswitched* or *always on* electrical outlet on the UPS.

Cabinet Cooling Fans

CMC Fan Unit

Two variable-speed fans are at the bottom of the cabinet. They receive +8 to +14 VDC from the power supply. An air filter, which can be removed and cleaned or replaced, is located above the fans. Air flows from the outside, into the bottom of the cabinet, around the circuit packs, and out through the top of the cabinet.

If the cabinet temperature reaches 158^{o} F (70^{o} C), the temperature sensor in the power supply shuts the system down and invokes the emergency transfer.

MCC1 Fans

A fan unit consisting of 6 fans, mounts near the center of the cabinet. The 3 front fans blow up and the 3 rear fans blow down. A removable air filter is provided above and below each fan unit. Four sensors monitor the cabinet temperature; 3 sensors are inside the cabinet top and 1 sensor is inside the cabinet bottom. One of the top sensors affects the speed of the front fans and the bottom sensor affects the speed of the rear fans. A speed control and thermal alarm circuit in each fan monitors the sensors. When a sensor indicates a change in cabinet temperature, the circuit in a fan changes that fan's speed accordingly.

A power cable from the Power Distribution Unit connects -48 VDC to each fan, +5 VDC to the speed control and thermal alarm circuit in each fan, and temperature sensor signals to the equivalent circuit in each fan. One pair of wires routes to each fan circuit. Alarm signals also route to the equivalent circuit in each fan. One pair of wires routes to each fan circuit.

A minor alarm is sent to the processor circuit pack in the PPN cabinet and the maintenance circuit pack in an EPN cabinet if any fan's speed drops below minimum. A minor alarm occurs if a fan has stopped due to loss of -48 VDC. A major alarm is sent by one of the cabinet top thermal sensors if the exhaust temperature reaches 149 ^{o}F (65^{o}C).

Another cabinet top sensor senses if the exhaust temperature reaches 158 ^{o}F (70 ^{o}C). If so, the system shuts down and the Emergency Transfer is invoked.

SCC1 Fan Unit

Four constant-speed fans at the top rear of the cabinet receive -48 VDC from the backplane. An air filter is located below the fan unit. Air flows down through the filter over the circuit packs. The filter is removable and is cleaned or replaced when necessary.

If the cabinet temperature reaches 158^{o} F (70^{o} C), the temperature sensor in the power supply causes the system to shut down and invokes the Emergency Transfer.

System Protection

Protections are established to keep the switch active and on line. The following 4 types of system protection are provided:

- Overvoltage
- Sneak current
- Lightning
- Earthquake

Overvoltage Protection

Protection from hazardous voltages and currents is required for all off-premises (out-of-building) trunks, lines, and terminal installations. Both over-voltage protection (lightning, power induction, and so forth) and sneak current protection are required. The following devices protect the system from overvoltages:

- Analog trunks use the 507B Sneak Protector. Over-voltage protection is normally provided by the local telephone company.
- Analog voice and 2-wire DCP terminals can use 1 of the following (or equivalent) types of combined overvoltage and sneak current protection:
    - Carbon block with heat coil for UL code 4B1C
    - Gas tube with heat coil for UL code 4B1E-W
    - Solid state with heat coil for UL code 4C1S
- DCP and ISDN-BRI terminals use the solid state 4C3S-75 with heat coil protector, or equivalent.
- DS1/E1/T1 circuits require isolation from exposed facilities. A CSU (T1), LIU (E1), or other equipment provides this isolation.

Sneak Current Protection

Sneak current protection uses fuses to protect building wiring between the network interface and trunk circuits when exposed to extraneous power. The fuses also protect the circuit packs.

All incoming and outgoing trunks and off-premises station lines pass through the sneak fuses. 507B Sneak fuse panels install on the system side of the network interface.

Sneak current protectors must be either UL listed/CSA certified or must comply with local safety standards. Sneak current protectors must have a maximum rating of 350 mA and a minimum voltage rating of 600V, or as required by local regulations.

Lightning Protection

A Coupled Bonding Conductor (CBC) in the cabinet ground wiring protects the system from lightning. The CBC runs adjacent to wires in a cable and causes mutual coupling between itself and the wires. The mutual coupling reduces the voltage difference between ground and the switch.

When using a CBC, be sure that it connects to telecommunication cable that is firmly connected to an approved ground. In multiple-story buildings, be sure to connect the CBC to an approved ground at each floor.

The CBC can be a 10 AWG (5.3 mm^2/2.6 mm) ground wire, a continuous cable sheath surrounding wires within a cable, or 6 unused pairs of wire within a cable, twisted and soldered together. The CBC connects from the cabinet single-point ground bar in an AC-powered cabinet or the ground discharge bar in a DC-powered cabinet to the terminal bar at the cross-connect field.

When there is an Auxiliary cabinet, a 6 AWG (13.3 mm^2/4.1 mm) wire connects the system cabinet single-point ground block to the Auxiliary cabinet ground block. The ground wire routes as closely as possible to the cables that connect the system cabinet to the Auxiliary cabinet.

If equipment is not present in the Auxiliary cabinet, plug the power supply for this equipment into 1 of the 2 convenience outlets on the rear of the Multi-Carrier Cabinet, to preserve ground integrity. The convenience outlet is fused at 5 Amps. A dedicated maintenance terminal plugs into the other convenience outlet.

Earthquake Protection

For earthquake or disaster bracing, the cabinets bolt to the floor. Other areas may require additional bracing. Contact your Avaya representative for earthquake requirements at the location of the system installation.

Temperature and Humidity

Install the equipment in a well-ventilated area. Maximum equipment performance is achieved at an ambient room temperature between 40 and 120^{o} F (4^{o} and 49^{o} C) for short term operation (not more than 72 consecutive hours or 15 days in a year) and up to 110^{o} F (43^{o} C) for continuous operation.

The relative humidity range is 10 to 95% at up to 84^{o} F (29^{o} C). Above this, maximum relative humidity decreases from 95% down to 32% at 120^{o} F (49^{o} C). Installations outside these limits may reduce system life or affect operation. The recommended temperature and humidity range is 65^{o} to 85^{o} F (18^{o} to 29^{o} C) at 20 to 60% relative humidity. Refer to the following table.

Temperature and Relative Humidity

Room Temperature (Degrees Fahrenheit)	Room Temperature (Degrees Celsius)	Relative Humidity (%)
40 to 84	4.4 to 28.8	10 to 95
86	30.0	10 to 89
88	31.1	10 to 83
90	32.2	10 to 78
92	33.3	10 to 73
94	34.4	10 to 69
96	35.6	10 to 65
98	36.7	10 to 61
100	37.8	10 to 58
102	38.9	10 to 54
104	40.0	10 to 51
106	41.1	10 to 48
108	42.2	10 to 45
110	43.3	10 to 43
112	44.4	10 to 40
114	45.6	10 to 38
116	46.7	10 to 36
118	47.8	10 to 34
120	48.9	10 to 32

Cabinet Dimensions and Clearances

For maintenance access, floor plans typically allocate space around the front, ends, and rear of the cabinets. Floor area requirements vary between cabinets. Dimensions and clearances for SCC1 and MCC1 cabinet configurations are listed in the following table.

Cabinet Dimensions and Clearances

Cabinet Type	Height	Width	Depth	Clearance
SCC1 1-cabinet	20 in. (51 cm)	27 in. (69 cm)	22 in. (56 cm)	38 in. (97 cm) between cabinet and wall
2-cabinets	39 in. (99 cm)	27 in. (69 cm)	22 in. (56 cm)	
3-cabinets	58 in. (1.5 m)	27 in. (69 cm)	22 in. (56 cm)	
4-cabinets	77 in. (2 m)	27 in. (69 cm)	22 in. (56 cm)	
MCC1[1]	70 in. (1.8 m)	32 in. (81 cm)	28 in. (71 cm)	Rear 38 in. (97 cm) Front 36 in. (91 cm)
Cable slack manager[2]	7 in. (18 cm)	32 in. (81 cm)	38 in. (97 cm)	
DC power cabinet[3]	20 in. (51 cm)	27 in. (69 cm)	22 in. (56 cm)	38 in. (97 cm) Front and Rear
Large battery cabinet 100 Amp	27 in. (69 cm)	55 in. (140 cm)	21 in. (53 cm)	38 in. (97 cm) Front and Rear
200 Amp	42 in. (107 cm)	55 in. (140 cm)	21 in. (53 cm)	
300 Amp	42 in. (107 cm)	55 in. (140 cm)	21 in. (53 cm)	
400 Amp	57 in. (145 cm)	55 in. (140 cm)	21 in. (53 cm)	

1. Includes the auxiliary cabinet, the global AC cabinet, and the global DC cabinet.
2. Used with Multi-Carrier and Single-Carrier cabinets.
3. Requires a floor area of 8 square feet (0.74 square m). Also requires 38 in. (97 cm) between cabinet and wall.

Optional Components

S8300 Media Server in an LSP Configuration

The S8300 Media Server in a Local Survivable Processor (LSP) configuration uses the S8300 as its hardware component and requires a software license to activate the standby feature. This software allows the S8300 Media Server with a G700 Media Gateway to be a survivable call-processing server for remote and branch customer locations when used with the S8700 Media Server in a Multi-Connect configuration.

In the event that the link is broken between the remote G700 Media Gateway and the S8700 Media Server with Multi-Connect as the primary controller, or if the S8700 Media Server fails, the S8300 Media Server in an LSP configuration activates. The S8300 Media Server in an LSP configuration only takes over for those telephones connected to the G700 Media Gateway in which it is housed.

While the hardware for the S8300 Media Server and the S8300 Media Server in an LSP configuration is identical, it has the following software differences:

- Different IP Address than the S8300 Media Server it is supporting
- License File has six days of use once a G700 registers with it
- Receives Saved Translations from the primary server

The S8300 Media Server in an LSP configuration only supports the S8700 Media Server with Multi-Connect.

CAUTION:
The S8300 in an LSP configuration differs from other Media Modules because of its height. Plug the S8300 Media Server in an LSP configuration into **only** the left-most slot under the LED board and panel. When installing or removing, treat the S8300 Media Server and LED board as a single unit.

IP phones obtain their own IP address from a DHCP server. The DHCP server also sends a list of controllers and S8300 Media Servers in an LSP configuration and associated IP addresses. The IP phone then registers to the controller that corresponds to the first IP address in this list. When connectivity is lost between the controller and the endpoint, the endpoint registers with the second IP address in the list, and so on. This list is administrable for phones on the DHCP server.

The S8300 Media Server in an LSP configuration is bought with a special license that enables MultiVantage to run in a backup mode. When the S8300 Media Server in an LSP configuration becomes active and begins call processing, an alarm is raised notifying that the S8300 Media Server in an LSP configuration is in service, and a timer is started. This timer runs for six days, at which time call processing ceases. If the problem is fixed within that six-day period, the S8300 Media Server in an LSP configuration can be reset, all endpoints will then register with the primary controller which is the S8700 Media Server with Multi-Connect, and the timer will be reset to allow for six days upon another failure.

While the S8300 Media Server in an LSP configuration is not call preserving in all circumstances, although IP-to-IP calls will be preserved, the fail-over from the S8700 Media Server with Multi-Connect, which is the primary controller, to the S8300 Media Server in an LSP configuration is an automatic process and does not require human intervention. The fail-back from the S8300 Media Server in an LSP configuration to the primary controller is not automatic and requires a system reset on the S8300 Media Server in an LSP configuration. This breaks the communication between the S8300 Media Server in an LSP configuration and all registered endpoints and causes the endpoints to register with the primary controller. During fail-back to the primary controller, all calls will be dropped with the exception of IP-to-IP calls.

Translation changes made on the primary controller will be propagated to the S8300 Media Server in an LSP configuration through an automated process. Following the daily translation save on the primary controller, the time of which can be administered by the customer, or after any save trans is issued, the primary controller will copy translations to all LSPs.

The physical installation for the S8300 Media Server in an LSP configuration differs from other Media Modules because of its height. The S8300 Media Server in an LSP configuration still plugs into a Media Module slot but is restricted to the left most slot under the LED board and panel. When installing or removing the S8300 Media Server in an LSP configuration, the S8300 Media Server in an LSP configuration and the LED board must be removed as a single unit. This is because the LED faceplate will hit the tall components on the S8300 Media Server in an LSP configuration if it is not removed as well.

The S8300 Media Server and the S8300 Media Server in an LSP configuration cannot reside in the same G700 Media Gateway.

Avaya R300 Remote Office Communicator

The Avaya R300 offers a cost-effective method for providing full functionality at a remote site. The remote telephony available through the R300 has all the capabilities of telephony that is connected directly. Because voice and data can share the same WAN link between the DEFINITY switch and the remote site, the R300 provides voice and data convergence.

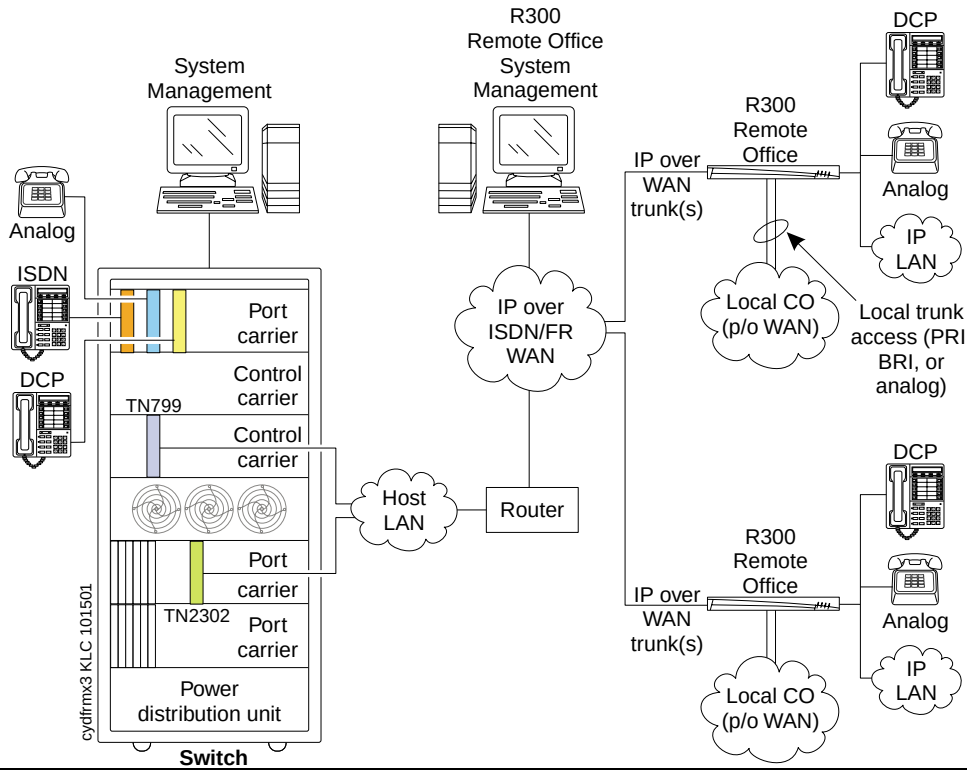
The R300 terminates at the local LAN and supports local PSTN connections that use the following methods:

- 2 DS1/E1/T1/PRI
- 6 BRI ST V.35 serial port connections
- 2 analog trunks

The R300 provides connections between remote stations and local access trunks. It is a rack-mounted box that features two expansion slots. One slot houses a DSP blade for VoIP support; the other slot houses a new combo blade that supports the DCP as well as the analog line and trunk connections. The R300 supports up to 24 DCP sets and 2 analog lines from a host switch. A single DEFINITY switch can support multiple R300 units, as shown in *DEFINITY Systems Capacities*, 555-233-604. This document can be viewed from the documentation Web site.

Typically, the host DEFINITY is linked to the R300 as shown in the following figure.

Avaya R300 Remote Office Communicator

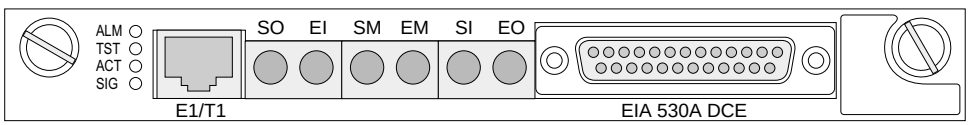


Media Modules

T1/E1 Media Module (MM710)

The MM710 media module is designed to terminate a T1/E1 connection.. It has built-in Channel Service Unit (CSU) so an external CSU is not necessary. Refer to the following figure for an example of the MM710.

MM710



mmdc710 KLC 020402

The MM710 Media Module offers these functions:

- Software selectable T1 or E1 operation
- Integrated CSU
- Both A-law (E1) and Mu-law (T1) gain control and echo cancellation ability
- Bantam Racks
- D4, ESF, or CEPT framing
- 700A Loopback Jack
- ISDN PRI capability (23B + D or 30B + D)
- Line Coding: AMI, ZCS, B8ZS (T1) or HDB3 (E1)
- Trunk signaling for supporting US and International CO or tie trunks
- Echo Cancellation in either direction
- Fractional T1 support
- OIC DB 25-pin interface

The MM710 Media Module supports the universal DS1 conforming to the ANSI T1.403 1.544 Mbps T1 standard and the ITU-T G.703 2.048 Mbps E1 standard.

NOTE:
Coded Mark Inversion line coding used in Japan is not supported.

Echo Cancellation

The MM710 can cancel echoes in either direction for any DS0. It can cancel echoes with tail end delays up to 96 ms. It is compatible with either A-law or Mu-law code.

CSU Function

The MM710 is capable of long haul or short haul transmission. The MM710 can receive signals as low as -36 dB. The MM710 can compensate for distances up to 655 feet in short haul operation. Attenuation up to -22.5 dB can be programmed when driving repeaters for long haul transmission.

Loopback/BERT Functions

In the unpowered state, the MM710 provides a passive loopback for the far end. It can be set up for line or payload loopbacks. It supports incoming and outgoing ESF FDL requests for these as well. It can generate and respond to in-band loop up and loop down codes per ANSI-T1.403. For Bit Error Rate Testing, the MM710 supports the generation and detection of test patterns as well as injection of bit errors.

E1 impedance

By itself, the MM710 can be configured for balanced 120 ohm E1 operation. An external balun is required for 75 ohm unbalanced operation.

Bantam Jacks

Six bantam jacks on the faceplate provide access to the incoming and outgoing T1 or E1 signals:

- SM permits passive monitoring of the incoming line.
- EM permits passive monitoring of the outgoing line.
- SO permits intrusive monitoring of the incoming signal from the network. When used, this jack breaks the connection of that signal to the framer.
- EI permits injection of a signal towards the framer. When used, this jack isolates the network Rx signal.
- SI permits injection of a signal towards the network. When used, this jack isolates the framer Tx signal from going out to the network.
- EO permits intrusive monitoring of the signal from the framer. When used, this jack breaks the connection of that signal to the network jack,RJ48C.

LEDs

Four LEDs are supported on the faceplate, including the three standard Media Module LEDs and the SIG LED which indicates the MM710 is receiving a valid signal

DB 25 DCE connector

This connector may be used for a DSU (data service unit) function in a future release.

Loopback Jack

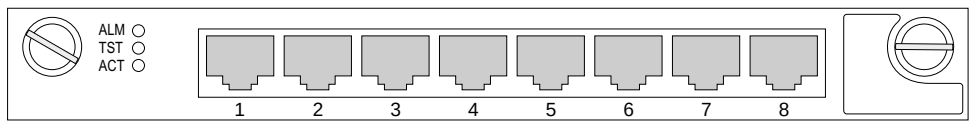
When ordering a MM710 Media Module, it is important to include the optional 700A loop back jack which allows the customer to loop back the T1 up to the network facility, without a dispatch. If the MM710 is sold with an Avaya Service Agreement, the jack must be ordered and installed to save time and money on service calls.

The jack is typically used for CO trunk installations. It is inserted as close to the network or telco T1 facility as possible. When activated from the G700 Media Gateway, it sets up loopbacks in both directions. The G700 Media Gateway can then send and receive a test pattern to verify the function of the MM710 and T1 cable up to the network T1 facility. In normal operation, it passes the T1 signals through undisturbed in both directions.

Analog Trunk/Telephone Media Module (MM711)

The MM711 Media Module provides analog trunk and telephone features and functionality. Refer to the following figure for an illustration of the MM711.

MM711



mmdc711 KLC 022702

The MM711 provides the administrator with the capability to configure any of the eight ports of this analog circuit pack as follows:

- Central Office trunk, either loop start or ground start
- Analog Direct Inward Dialing (DID) trunks, either wink start or immediate start
- 2-wire analog Outgoing CAMA E911 trunks for connectivity to the PSTN
- MF signaling is supported for CAMA ports
- Analog, tip/ring devices such as single-line telephones with or without LED message waiting indication

The MM711 also supports:

- Type 1 and Type 2 Caller ID
- Ring voltage generation for a variety of international frequencies and cadences
- A hard-wired ground wire is added for each IROB to earth ground

External Interfaces: CO Trunk Side

The following requirements apply to the external interfaces on the CO trunk side:

- The tip ring default input impedance is 600 ohms and is configurable to accommodate other tip ring impedances such as 900 ohms used in Brazil and complex impedance used in the European Union.
- A hard-wired ground wire is added for each IROB to earth ground.
- The MM711 supports DTMF, MF and Pulsing.
- The MM711 supports R2MFC address signaling, and provides -48 vdc for ports set up as DID.
- The MM711 supports DIOD for Japan.

The following trunk types are supported:

- CO [GS/LS]
- DID [Immed/Wink]
- CAMA

External Interfaces: Line Side

The following requirements apply to the external interfaces on the line side: the MM711 supports only balanced ringing.

Caller ID

The MM711 supports ICLID on analog CO loop-start trunks for all supported countries requiring this feature. It supports Type 1 CID devices, and firmware signaling requirements are implemented on a per-port basis. The firmware supports two formats:

- Single Data Message Format (SDMF)
- Multiple Data Message Format (MDMF)
- Caller ID generation on Line Ports

The MM711 accommodates on-hook transmission which is necessary to receive caller ID signals.

On an ICLID administered trunk, absence of ICLID information or error in transmission of ICLID information will not prevent the call from being terminated, with the exception of Japan.

Analog Line Interface Requirements

The MM711 provides pass through for fax signals.

The MM711 supports analog telephone sets with:

- An impedance range of Rs: 215 - 300 ohms; Rp: 750 - 1000 ohms; Cp: 115 - 220pF
- A ringing frequency range of 20, 25, or 50 Hz
- A DC current range of 20 - 60 mA
- A hook flash range of 90 ms - 1000 ms

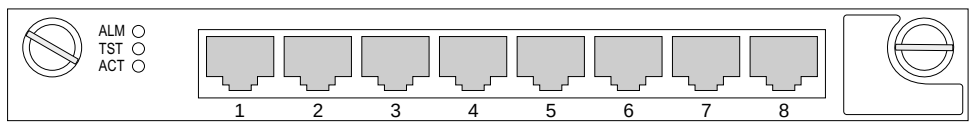
Companding

MM711 allows for A-law or mu-law selection at installation. This is a software-selectable capability that applies to all ports on the MM711.

DCP Media Module (MM712)

The MM712 Media Module allows connectivity of up to eight two-wire Digital Communications Protocol (DCP) voice terminals. Refer to the following figure for an example of the DCP Media Module.

MM712



mmdc712 KLC 022702

NOTE:
DCP is the Avaya proprietary signaling protocol used to interface the DCP-compatible Avaya digital voice to the S8300 Media Server with the G700 Media Gateway.

Hardware Interface

Signal timing specifications for MM712 support TDM Bus Timing in receive and transmit modes. The G700 Media Gateway supplies only +5 vdc and –48 vdc to the MM712. Any other required voltages must be derived on the module.

Loop range secondary protection is provided on the MM712, and the board is self-protecting from overcurrent condition on Tip and Ring interface.

Hardware Group Tests

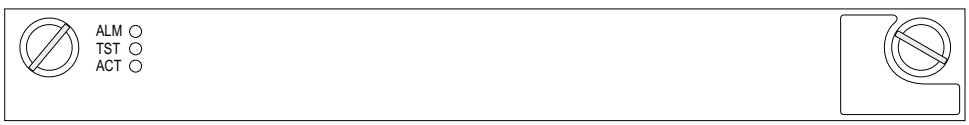
The MM712 performs the same hardware group tests available on the TN2224B circuit pack. These include:

- Control Channel Loop
- NPE+ Audit
- NPE+ Cross Talk Test
- Power Feed Test
- Local Digital Loop Around
- Remote Digital Loop Around

VoIP Media Module (MM760)

The MM760 Media Module is a clone of the motherboard VoIP engine. It provides an additional 64 VoIP channels with G.711 compression. Refer to the following figure for an example of a MM760.

MM760



mmdc760 KLC 022702

The capacity is 64 G.711 TDM/IP simultaneous calls, or 32 compression codec, G.729 or G.723, TDM/IP simultaneous calls. These call types can be mixed on the same resource; in other words, the simultaneous call capacity of the resource is 64 G.711 Equivalent Calls.

NOTE:
Customers who desire an essentially non-blocking system must add an additional MM760 Media Module if using more than two MM710 Media Modules in a single chassis. This provides for an additional 64 channels.

Ethernet Interface

The MM760 needs to have its own Ethernet address. It requires a 10/100 Base T Ethernet interface to support H. 323 endpoints for DEFINITY IP trunks and stations from another G700 Media Gateway to be switched to any resource within the G700 Media Gateway.

Voice Compression

The MM760 Media Module supports on-board resources for compression and decompression of voice for G. 711 A-law and Mu-law, G. 729 and G. 729B, and G.723 (5.3K) and(6.3K).

The VoIP engine DSP supports the following functionalities:

- RTP and RTCP interface
- Dynamic Jitter Buffers
- DTMF Detection
- Hybrid Echo cancellation
- Silence suppression
- Comfort noise generation
- Packet loss concealment

Circuit Packs and Power Supplies

Circuit Packs

1217A AC Power Supply (SCC1 Media Gateway)

The 1217A power supply provides additional protection to your equipment in the event of an overload. Overloads trigger a power-supply alarm, but the system's cooling fans continue to run, and the power supply continues to function. The new power supply replaces the WP-51193 L3 and L4-25. It is fully backward compatible.

The 1217A is a power factor-corrected, 50/60-Hz, autoranging (90-264 VAC) input, multi-output power supply providing regulated DC output and switch-selectable 20/25-Hz AC Ringer output voltages. It meets the harmonic distortion requirements of IEC1000-3-2 (PFC), the immunity requirements of IEC 1000-4, and the safety requirements of IEC 950, as well as current UL and CSA requirements.

The 1217A mounts in the power supply slot of the carrier. A power cord with a 3-prong plug on one end connects the supply to a dedicated, AC power source.

631DA1 AC Power Unit (MCC1 Media Gateway)

The 631DA1 accepts 120 VAC 60 Hz and produces the +5 VDC at 60 A available on the carrier backplanes.

If the AC input power fails, the unit converts 144 VDC from the optional batteries in the AC power distribution unit to +5 VDC. A circuit in the battery charger detects the highest equivalent AC or DC input voltage and switches in the correct input voltage.

631DB1 AC Power Unit (MCC1 Media Gateway)

The 631D B1 accepts 120 VAC 60 Hz and produces –48 VDC at 8 A and –5 VDC at 6 A available on the carrier backplanes. The –48 VDC also supplies power to the cabinet fans.

If the AC input power fails, the unit converts 144 VDC from the optional batteries in the AC power distribution unit to –48 VDC and –5 VDC. A circuit in the optional battery charger detects the highest equivalent AC or DC input voltage and switches in the correct input voltage.

649A DC Power Unit (MCC1 Media Gateway)

The 649A power converter converts a –48 VDC input into outputs of –48 VDC at 10 A, +5 VDC, and –5 VDC at 6 A. The outputs are distributed to circuit pack slots in the carriers. Only one 649A converter is required per carrier except for SN carriers. SN carriers require 2 converters; 1 on each end. The use of the 649A also allows an increase in the number of terminals supported per carrier.

650A AC Power Unit (CMC1)

The 650 A is a global power-factor-corrected AC/DC converter providing multiple DC outputs and AC ring outputs. The unit is autoranging and accepts 85-264 VAC, 47-63 Hz AC input, and provides 330 Watts total output. The unit provides multiple DC outputs as follows:

- +5.1 VDC, 28 A
- -5.1 VDC, 1.0 A
- -48 VDC, 4.5 A
- +8 to +14 VDC, 1.6 A (fan speed control)
- -150/-115 VDC, 200 mA (neon bus)

The power unit has 3 switch-selectable ring outputs:

- 85 VAC RMS, 80 mA, 20 Hz, centered about -48 VDC, 180 mA
- 72 VAC RMS, 8 to 80 mA, 25 Hz, centered about -48 VDC, 180 mA
- Two 28 VAC RMS (56 V eff), 220 mA, 50 Hz biased about -48 VDC and 0 VDC, 70 mA balanced

The fan speed is controlled by the +8 to +14 VDC (+12 VDC nominal). The output voltage is a function of the ambient inlet air temperature at the bottom of the power supply. The fan output voltage is +14 VDC if the FANALM signal is active.

676C DC Power Supply (SCC1 Media Gateway)

A –48 VDC source supplies power to the DC power supply at up to 25 A. The 676C produces the following DC outputs: +5 VDC, –5 VDC, –48 VDC, and +12 VDC. The DC outputs are distributed on the cabinet backplane to the slots for the circuit packs. The AC ringing voltage output value and frequency depend on the country of use. The power supply has circuit breakers and Electromagnetic Interference (EMI) filtering.

982LS Current Limiter for SI (PPN)

The 982LS connects to the rear of the processor circuit pack slot only in the PPN in SI configurations. The 982LS provides current-limited accessory 48 VDC, emergency transfer logic, current-limited 5 VDC to trip main circuit breaker if high temperature is detected, and duplicated 48 VDC to fans in the PPN cabinet.

CFY1B Current Limiter for R

The CFY1B is only in R configuration PPNs, MCC1 EPNs, and single-carrier cabinet EPNs. The unit connects to the rear of the maintenance circuit pack slot and provides current-limited accessory 48 VDC, emergency transfer logic, current-limited 5 VDC to trip the main circuit breaker if high temperature is detected, and duplicated 48 VDC to fans in the EPN cabinet.

ED-1E568 DEFINITY AUDIX R4

See "TN568 DEFINITY AUDIX 4.0 Voice Mail System (part of ED-1E568)" on page 304.

J58890MA-1 Multi Application Platform DEFINITY (MAPD)

The J58890MA-1 is a variation of the MAPD platform that transports ASAI links between a DEFINITY LAN gateway system and an Ethernet LAN. The J58890MA-1 circuit-pack assembly uses the TN801B MAPD (LAN Gateway Interface), a circuit pack built from industry-standard PC processors, interfaces, buses, and ISA/PCI expansion boards. The unit takes up 3 adjacent slots (unless placed in slots 6 and 7 in a CMC1 Media Gateway, then it uses only those two slots).

- List 2 supports CallVisor ASAI and LAN Gateway
- List 10 supports IP Trunking
- List 20 supports CallVisor ASAI, CentreVu Computer Telephony, and BCMS Vu
- List 30 supports IP solutions

NAA1 Fiber Optic Cable Adaptor (CMC1)

The NAA1 adapter reroutes fiber-optic cable from the front of an ATM circuit pack to the rear of a CMC1 Media Gateway. While it looks like a circuit pack, it is electrically and optically passive.

TN429D DIOD or Central Office Trunk (8 ports)

The TN429 incoming call line identification (ICLID) circuit pack provides 8 ports for direct inward/outward dialing (DIOD) trunks. Each port provides a 2-wire interface to the central office (CO) public exchange for incoming and outgoing calls. The CO provides caller names and numbers to the circuit pack, which displays them on digital telephones (DCP and BRI) equipped with a 32- or 40-character alphanumeric display, and with System 25/MERLIN 7315H and 7317H terminals supported by the switch. This feature is supported in the United States (name and number) and Japan (number only) and countries that comply with either requirements.

This circuit pack is required for the Japan ANI feature where the calling number passes through to the switch. An in-band detector/converter may be required. Contact your Avaya representative.

The TN744 or TN2182 work with TN429 to provide tone generation and tone detection. Upon seizing out, dial tone from the CO passes through the TN429 to a tone detector. The TN429 provides the required CO disconnect functions as well as the interface to CAMA/E911.

TN433 Speech Synthesizer for Italian

The TN433 provides 4 ports that retrieve fixed messages for leave word calling, Automatic wakeup, and visually impaired attendant console features. Examples of the messages are: good morning, time-of-day, and extension number. Each of the ports has touch-tone detection. The TN433 has administrable µ-Law and A-Law companding.

TN436B Direct Inward Dialing Trunk for Australia (8 ports)

The TN436B provides 8 ports for Direct Inward Dialing (DID) independently connected to a public network. Each port is an interface between a 2-wire analog line from a CO and the 4-wire TDM network in the configuration. The TN436B has administrable timers.

TN438B Central Office Trunk for Australia (8 ports)

The TN438B provides 8 ports for loop-start CO trunks. Each port has tip and ring signal leads. The TN438B can detect 12 kHz and 50 Hz periodic metering pulses from the CO. Additional features include call still held timing and automatic guard fault detection circuitry.

TN439 Tie Trunk for Australia and Japan (4 ports)

The TN439 provides 4 ports for 2-wire tie trunks with loop disconnect signaling. The TN439 has administrable A-Law and µ-Law companding and administrable timers.

TN457 Speech Synthesizer for British English

The TN457 provides 4 ports that retrieve fixed United Kingdom-accent spoken messages for leave word calling, automatic wakeup, and visually impaired attendant console features. Examples of messages are: good morning, time-of-day, and extension number. Each of the ports has touch-tone detection. The TN457 has administrable A-Law and µ-Law companding.

TN459B Direct Inward Dialing Trunk for The United Kingdom (8 ports)

The TN459B provides 8 ports for immediate-start or wink-start direct inward dialing (DID) trunks. Each port has tip and ring signal leads. Each port is an interface between a 2-wire analog line from a CO and the 4-wire TDM network in the configuration. The TN459B has administrable timers and a backward busy circuit that complies with signaling requirements.

TN464GP DS1 Interface, T1 (24-Channel) or E1 (32-Channel)

The TN464GP circuit pack provides:

- Board-level, administrable A-Law or µ-Law companding
- CRC-4 generation and checking (E1 only)
- Stratum 3 clock capability
- ISDN-PRI T1 or E1 connectivity
- Line-out (LO) and line-in (LI) signal leads (unpolarized, balanced pairs)
- Support for CO, TIE, DID, and off-premises station (OPS) port types that use robbed-bit signaling protocol, proprietary bit-oriented signaling (BOS) 24th channel signaling protocol, or DMI-BOS 24th channel signaling protocol
- Support for Russian incoming ANI
- Support for universal, digital, signal level-1 boards in wideband ISDN-PRI applications
- Test-jack access to the DS1 or E1 line and support of the 120A Integrated Channel Service Unit (CSU) module
- Support for the enhanced maintenance capabilities of the integrated channel service unit (ICSU). These circuit packs can communicate with CONVERSANT®. See "TN2185B ISDN-BRI S/T-TE Interface (4-wire, 8 ports)" on page 322.
- Firmware downloadable
- Support for echo cancellation. To enable the echo cancellation capability, the customer must purchase an Echo Cancellation Software Right-to-Use feature in addition to the hardware.

  The echo cancellation capability of the TN464GP is selectable per channel. The TN464GP automatically turns off echo cancellation when it detects a 2100-Hz phase-reversed tone put out by high-speed modems (56 Kbaud), but not when it detects a 2100-Hz straight tone generated by low-speed modems (9.6 Kbaud). Echo cancellation improves a low-speed data call.

  The TN464GP is intended for customers who are likely to encounter echo over circuits connected to the public network. The occurrence of echo is higher if the switch is configured for ATM, IP, or other complex services and interfaces to local service providers who do not routinely install echo cancellation equipment in all their circuits. A common source of echo is “hybrid” circuits, where conversions between 2-wire analog and 4-wire digital circuits take place. The TN464GP cancels echo with delays of up to 96 ms.

TN465C Central Office Trunk for Multiple Countries (8 ports)

The circuit pack supports 8 analog CO trunk ports, loop-start trunk signaling, 12-kHz and 16-kHz periodic pulse metering (PPM) detection and counting, administrable timers, battery reversed signaling, and multi-country selectable signaling. For more information, contact your Avaya representative.

TN479 Analog Line (16 ports)

The TN479 has 16 ports and supports 3 ringer loads and 3 simultaneous ringing ports. Only one telephone can have an LED message-waiting indicator (neon message-waiting indicators are not supported). The TN479 supports µ-Law companding.

The TN479 supports the telephones listed below.

Telephone	Wire Size (metric area/diameter)	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	3,000 ft. (914 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	3,000 ft. (914 m)
7100-Series	24 AWG (0.2 mm^2/0.5 mm)	3,000 ft. (914 m)
7101A-Series	not supported	not supported
7103A-Series	not supported	not supported
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	2,500 ft. (762 m)
9100-Series	24 AWG (0.2 mm^2/0.5 mm)	2,500 ft. (762 m)

TN497 Tie Trunk for Italy TGU, TGE, and TGI (4 ports)

The TN497 has 4 ports for 2-wire tie trunks with loop disconnect signaling. Each port can be administered for A-Law and µ-Law companding, timers, Translatore Giunzione Unscente (TGU) (outgoing tie), Translatore Giunzione Entrante (TGE) (incoming tie), and Translatore Giunzione Interno (TGI) (internal tie).

TN553 Packet Data Line

The TN553 has 12 ports that can connect through a cross-connect field to a TN726B circuit pack and provides software-administrable connections between the Switch Processing Element (SPE) and system access ports. Inside the system, the TN553 connects to the packet bus and converts mode-2 protocol to mode-3 protocol connecting the TN726B to the TDM bus for asynchronous EIA connections to adjuncts.

TN556D ISDN-BRI 4-Wire S/T-NT Interface (12 ports)

The TN556 has 12 ports connecting to ISDN-BRI terminals. Each port on a TN556 has TXT, TXR, PXT, and PXR signal leads. Up to 8 ports can be used for Adjunct Switch Application Interface (ASAI) links. Each port operates at 192 kbps per second and has 2 B-channels and 1 D-channel (not used to carry data).

The TN556 has a range of up to 1900 feet (579 m) maximum from the configuration to the telephone using 24 AWG (0.20 mm^2/0.51 mm) wire, and uses standard ANSI T1.605 protocol. The TN556 also has multipoint support; 24 terminals can be connected, where each terminal uses 1 B-channel and shares the D-channel. In multi-support applications, 2 telephones, or 1 telephone and 1 data terminal, or 2 data terminals can connect to each port.

The TN556 circuit pack supports A-Law or µ-Law companding. The TN556 also functions as a trunk, connecting to a TE interface (example: a TN2185B in another switch). It can be used for lines and trunks simultaneously (on the same circuit pack). The TN556 provides end-to-end outpulse signaling when the circuit pack is in tie-trunk mode with a TN2185B ISDN-BRI S/T-TE Interface (4-wire, 8 ports).

TN568 DEFINITY AUDIX 4.0 Voice Mail System (part of ED-1E568)

A component of the ED-1E568, the TN568 is a processor circuit pack that supports voice mail using an embedded 386EX processor. DEFINITY AUDIX systems can be interconnected to create large voice-mail networks that support up to 100,000 subscribers and store up to 100 hours of messages. Each circuit pack has 8 ports available for calls when networking is enabled. Without networking, 12 ports are available.

The TN568 takes up two adjacent slots, except in slot 6 of a CMC1 Media Gateway.

The circuit pack includes a writable magneto-optical disk drive for backing up and upgrading system software and a hard disk for storing messages. There is an RS-232 connection for a maintenance and administration terminal, an Ethernet port (for the Message Manager PC desktop application), an Amphenol connection to the switch, and an RS-232 port for an external, remote-maintenance modem.

TN570D Expansion Interface

The TN570 Expansion Interface (EI) is an interface between the TDM bus and packet bus, and fiber optic links interconnecting cabinets. It is used in a Port Network (PN) between a PN and another PN in a direct-connect configuration, and between a PN and an SNI in a switch node carrier in a CSS-connected configuration.

The TN570 provides control channel applications and time-slot interchanging between the PPN and EPNs. It is used when ISDN-BRI and/or ASAI is connected in an EPN.

The TN570 carries circuit-switched data, packet-switched data, network control, timing control, and DS1 control. This circuit pack also communicates with the TN775B maintenance circuit pack in an EPN to send the EPN environmental and alarm status to the SPE.

The TN570 is replaced by the TN2305 or TN2306 when an ATM switch replaces the Center Stage Switch.

The TN570 is used in an EPN that is supported by a Survivable Remote Processor (SRP).

TN572 Switch-Node Clock for DEFINITY R

The TN572 distributes the timing signals that synchronize the SN carrier. The TN572 also receives maintenance data.

TN573B Switch-Node Interface for DEFINITY R

The TN573B Switch Node Interface (SNI) routes circuit, packet, and control messages. The TN573B is an interface installed in a SN carrier in a CSS that terminates a fiber optic link from a SNI in an SN carrier to an SNI in another SN carrier, an EI in a PPN, and an EI in an EPN. One TN573B is used per PN and supports the TN574 DS1 converter circuit pack.

The TN573B and higher provides an interface to the single-mode fiber optic transceiver and supports the TN1654 and TN574 DS1 converter circuit pack.

TN577 Packet Gateway for DEFINITY R

The TN577 packet gateway (PGATE) provides 4 RS-423 physical ports for X.25 protocol interfaces between the configuration and adjuncts. In this application, PGATE functions as the data communications interface unit providing protocol conversion between the X.25 protocol and the mode 3 protocol carried across the LAN Bus.

The X.25 protocol (Levels 1 and 2) is terminated and the data reformatted into the ISDN packet mode protocol for transport across the LAN bus. Supported adjuncts include AUDIX, CMS, and Message Server Adjunct (MSA).

The TN577 also supports the Distributed Communications System (DCS) environment by providing X.25 signaling through 1 of the RS-423 physical ports, or back through the system using the TDM Bus to the appropriate DS1 or tie trunk circuit pack.

TN725B Speech Synthesizer for United States English

The TN725B has 4 ports that send voice message information to telephones to activate leave word calling, automatic wakeup, voice message retrieval, and Do Not Disturb features. The ports can detect tones.

TN726B Data Line (8 ports)

The TN726B has 8 serial asynchronous EIA ports with modem interfaces connected through asynchronous data units (ADUs) to EIA ports (such as RS-232) on DTE. The TN726B uses Mode 2 or Mode 3 data transfer protocol. The DTE can be adjuncts and peripheral equipment such as data terminals, printers, host computers, personal computers (PCs), graphics and facsimile systems, and call detail acquisition and processing systems (CDAPSs).

With software-administered system access ports, a TN726B connects through a cross-connect field to a TN553 packet data line circuit pack. The TN553 then converts mode 2 protocol to mode 3 protocol transferring the TN726B from the packet bus to the TDM bus for EIA connections.

Each port on a TN726B has TXT (terminal, transmit, and tip), TXR (terminal, transmit, and ring), PXT (port, transmit, and tip), and PXR (port, transmit, and ring) signal leads.

TN735 MET Line (4 ports)

The TN735 has 4 ports that connect to multi-button electronic telephone (MET) sets. Each port has tip and ring (analog voice) and BT, BR, LT and LR (digital signals to control terminals) signal leads.

TN744E Call Classifier Tone Detector for Multiple Countries (8 ports)

The TN744 circuit pack has 8 ports of tone detection on the TDM bus. The TN744 does not support call progress tone generation or clocking. The tone detectors are used in vector prompting, outgoing call management (OCM), and call prompting applications in the United States and Canada and call classifier options for various countries.The TN744 detects special intercept tones used in network intercept tone detection in OCM. The TN744 also detects tones when a CO answers a call.

The TN744 provides tone generation and detection for R2-MFC DID signaling used in non-United States installations. The TN744 also allows gain or loss to be applied to pulse code modulation (PCM) signals received from the bus and supports A-Law and µ-Law companding. The TN744 detects 2025 Hz, 2100 Hz, or 2225 Hz modem answerback tones and provides normal broadband and wide broadband dial tone detection.

The TN744 processor supports digital signal processing of PCM signals on each port to detect, recognize, and classify tones and other signals. Generation of signaling tones is also supported for applications such as R2-MFC, Spain MF, and Russia MF. Gain (or loss) and conferencing can be applied to PCM signals received from the TDM bus. Additional support includes DTMF detectors to collect address digits during dialing, and A-Law and µ-Law companding.

In normal operation, a port on the TN744 may serve as an incoming register for Russia MFR (multi-frequency shuttle register signaling). Use the TN744 with the TN429C analog line central office trunk for CAMA/E911.

TN746B Analog Line (16 ports)

The TN746B has 16 ports. Each port supports one telephone. Auxiliary equipment — such as fax machines, answering machines, modems, and amplifier handsets — is supported.

The TN746B supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED and neon message waiting indicators. The TN746B supports off-premises wiring (out-of-building only with certified protection equipment) with either DTMF or rotary dialing, but LED or neon message waiting indicators are not supported off-premises. The TN746B provides -48 V DC current in the off-hook state. Ringing voltage is -90 V DC.

The TN746B, along with a TN755B neon power unit per carrier or per single-carrier cabinet, supports telephones equipped with neon message waiting indicators (on-premises use only). The TN746B supports 3 ringer loads, only 1 telephone can have an LED or neon message waiting indicator.

TN746B supports A-Law and µ-Law companding and administrable timers. The TN746B supports queue warning level lights associated with the DDC and UCD features, recorded announcements associated with the Intercept Treatment feature, and PagePac paging system for the Loudspeaker Paging feature. Additional support is provided for external alerting devices associated with the TAAS feature, neon message waiting indicators, and modems. Secondary lightning protection is provided on the TN746B. The TN746B supports up to 8 simultaneous ports ringing; 4 on ports 1 through 8, and 4 on ports 9 through 16.

Combined conversion of Modem Pooling requires a port on a TN754 and a port on a TN742, TN746B or TN769 Analog circuit pack for each combined resource that is to be supported.

The TN746B is compatible with the telephone configurations shown in the table below.

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
7100-Series	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
7101A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
7103A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
9100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

TN747B Central Office Trunk (8 ports)

The TN747B has 8 ports for loop-start or ground-start CO, foreign exchange (FX), and wide area telecommunications service (WATS) trunks. Each port has tip and ring signal leads. A port can connect to a PagePac paging system. The TN747B supports the abandoned call search feature in automatic call distribution (ACD) applications (if the CO has this feature). Vintage 12 or greater of the TN747B also provides battery reversed signaling.

TN750C Recorded Announcement (16 channels)

NOTE:
The TN2501AP has replaced the TN750. However, the TN750 will continue to be supported.

The TN750 records and stores announcements to be played back on demand as part of a calling feature. The TN750 has sampling rates of 16, 32, or 64 kbps. The TN750 records announcement messages from on- or off-premises telephones and can store up to 128 recorded announcements of 8 maximum minutes total. The TN750 has 16 channels and each can play any announcement. Up to 25 call connections can listen to each channel. This means a total simultaneous call capacity of 400 calls in si. 256 callers can connect to each channel in an R system.

Equipping 10 circuit packs in a system provides a total capacity of 42.6 minutes (at 32 kbps) and 160 ports. In other words, 160 announcements can play simultaneously. The 16 kbps compression rate (adequate for VDN of origin announcements) provides a total capacity of 85.3 minutes. Use of multiple circuit packs allows a more efficient method of providing many kinds of announcements and provides improved management of integrated announcements.

TN753B Direct Inward Dialing Trunk (8 ports)

The TN753B has 8 ports used for immediate-start and wink-start direct inward dialing (DID) trunks. Each port has tip and ring signal leads. For the Czech Republic of Slovakia and the Commonwealth of Independent States, vintage 17 (or greater) is required. The TN753B supports A-Law and µ-Law companding with vintage 17 (or greater).

The TN753B is required to support Brazil Block Collect Call.

TN754C DCP Digital Line (4-wire, 8 ports)

The TN754 has 8 asynchronous, 4-wire DCP ports that can connect to 7400-series and 8400-series digital telephones, 302A/B/C attendant consoles, or data modules. The TN754 has administrable A-Law and µ-Law companding. The TN754 supports the 8400-series terminal/Data Module types and 7400-series terminals.

Maximum Range of 7400-series and 8400-series equipment

Supported Equipment	Distance Limits	Wire
7400 Data Modules	5000 feet (1524m)	24 AWG (0.2 mm^2/0.5 mm)
7400 Data Modules	4000 feet (1219m)	26 AWG
7400-series Telephones	3500 feet (1067m)	24 AWG (0.2 mm^2/0.5 mm)
7400-series Telephones	2200 feet (670m)	26 AWG
8400-series Data Modules	3500 feet (1067m)	24 AWG (0.2 mm^2/0.5 mm)
8400-series Telephones	3500 feet (1067m)	24 AWG (0.2 mm^2/0.5 mm)

The TN754 provides greater call handling capacity for high traffic applications and supports the group paging feature.

Combined conversion of Modem Pooling requires a port on a TN754 and a port on a TN746B or TN769 Analog circuit pack for each combined resource that is to be supported.

TN755B Neon Power Unit for all models except DEFINITY CSI

The TN755B produces 150 VDC to operate neon message waiting lights on terminals connected to TN746B analog line circuit packs.

A TN755B neon power unit is required for each carrier where neon message waiting indicators are connected.

This circuit pack and the neon message waiting function are not available on systems using the TN2202 ring generator circuit pack for France balanced-ringing.

TN758 Pooled Modem (2 ports)

The TN758 has 2 conversion resources ports (such as a trunk data module) for switched connections between digital data endpoints (data modules) and analog data endpoints (modems). A TN758 Circuit pack is required for each 2 conversion resources provided with the integrated type of modem pool. The TN758 supports µ-Law companding only.

TN760E Tie Trunk (4-wire, 4 ports)

The TN760 has 4 ports used for Type 1 or Type 5 four-wire E & M lead signaling tie trunks, that can be automatic, immediate-start, wink-start, and delay-dial. Each port on a TN760 has T, R, T1, R1, E, and M signal leads. The TN760 provides release link trunks required for the CAS feature and has administrable A-Law and µ-Law companding. The TN760 supports outgoing, Multilevel Precedence and Preemption (MLPP).

Option switches on each TN760 port can select connections to Type 1 E & M standard unprotected format, Type 1 E & M compatible unprotected format, Type 1 E & M compatible protected format, and Type 5 simplex format.

For Belgium, Czech Republic of Slovakia, the Commonwealth of Independent States, and the Netherlands, use vintage 11 or greater.

TN762B Hybrid Line (8 ports)

The TN762B has 8 ports connecting to multi-appearance hybrid analog and digital telephones. It can connect to 7300-series telephones, an MDC-9000 (cordless telephone), and an MDW-9000 (cordless telephone with separate base station and charging stations).

Each port on a TN762B has VT and VR (analog voice), CT, CR, P-, and P+ (digital signals that control terminals) signal leads.

TN763D Auxiliary Trunk (4 ports)

The TN763 has 4 ports. Each port has T, R, SZ, SZ1, S, and S1 signal leads. The circuit pack is used for on-premises applications such as music-on- hold, loudspeaker paging, code calling, and recorded telephone dictation access. The TN763 supports external recorded announcement equipment, and is administrable to select A-Law or µ-Law companding.

TN767E DS1 Interface, T1 (24 Channel)

The TN767 circuit pack provides a DSX1 level physical interface to the DS1 facility and require a TN464 DS1 interface. The TN767 has unpolarized line out (LO) and line in (LI) signal lead pairs.

The TN767 supports DS1 rate digital facility connectivity. The circuit pack supports CO, Tie, DID, and off-premises stations (OPS) port types using the robbed-bit signaling protocol. This circuit pack also supports ISDN-PRI connectivity. For these applications, the signaling *D* channel can connect from the TN767 to the processor interface by a permanent switched call over the TDM bus.

The TN767 is used to communicate with CONVERSANT and to provide the enhanced maintenance capabilities of the 120A enhanced integrated channel service unit (ICSU) feature.

DS1 tests include loopback tests at the DS1 board edge or the 120A (if used), bit error rate (BER) loopback tests at the far-end CSU, and BER 1-way DS1 facility tests. Other tests include loopback testing specifically designed to locate DS1 facility faults.

TN769 Analog Line (8 ports)

The TN769 has 8 ports, each with tip and ring signal leads. The TN769 supports:

- On- or off-premises wiring (with certified protection equipment) with either touch-tone or rotary dialing and with or without LED or neon message waiting indicators
- 3 ringer loads, such as 3 telephones with 1 ringer load each
- Up to 4 simultaneous ports ringing
- Queue warning level lights associated with the DDC and UCD features
- Recorded announcements of the intercept treatment feature
- Dictation machines of the recorded telephone dictation access feature
- PagePac paging system for the loudspeaker paging feature
- External alerting devices of the trunk answer any station (TAAS) feature
- Modems

The TN769 does not support off premises message waiting indicators.

The TN769 provides secondary lightning protection, and supports µ-Law companding.

Each carrier with neon message indicators requires the TN769, along with a TN755B neon power unit to support neon message waiting indicators. Only 1 telephone can have an LED or neon message waiting indicator.

Combined conversion of Modem Pooling requires a port on a TN754B and a port on a TN746B or TN769 Analog circuit pack for each combined resource that is to be supported.

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
7102-Series	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
7101A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
7103A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	10,000 ft. (3,048 m)
9100-Series	24 AWG (0.2 mm^2/0.5 mm)	10,000 ft. (3,048 m)

TN771DP Maintenance/Test

The TN771DP performs maintenance functions. These functions include packet bus reconfiguration that allows diagnosis and correction of recoverable packet bus failures before the link access procedure on the D-channel (LAPD) links fail. LAPD is a link-layer protocol on the ISDN-BRI and ISDN-PRI data link layer (level 2). LAPD provides data transfer between 2 devices and error and flow control on multiple logical links. It recovers packet bus failures involving up to 3 malfunctioning leads (1 or 2 data or parity leads and 1 control lead) by swapping spare leads with the malfunctioning leads.

Other maintenance functions include ISDN-PRI testing that originates and terminates loopback tests on ISDN facilities. It provides bit and block error rate information indicating ISDN facility quality.

The TN771DP can be updated using the firmware download feature, which requires use of the TN799 C-LAN interface.

A TN771DP is required:

- in any CSI system that is using a TN2198 BRI circuit pack. Otherwise, a TN771DP is not required.
- in an SI system PPN that is equipped with packet endpoints (ISDN-BRI lines or trunks, ISDN-PRI trunks, IP trunks, IP stations, ATM-CES, and ASAI) or is a critical reliability or fully duplicated system. A critical reliability or fully duplicated system with packet endpoints requires a TN771DP in each EPN. Otherwise, a TN771DP is not required.
- in all R system PPNs. A critical reliability R system requires a TN771DP in each EPN. An R system with ATM network duplication requires a TN771DP in each PPN and EPN.
- All CSI models when using a TN2198 BRI circuit pack

A maximum of 1 TN771DP is allowed in any port network.

A TN771DP is never used with the S8100 Media Server configurations.

TN775C Maintenance Board (EPN)

The TN775 circuit pack is used in maintenance and monitor power failure signals in an EPN cabinet. This circuit pack also monitors the clock, monitors and controls the power supplies and battery charger, and monitors air flow and high temperature sensors. This circuit pack provides 2 serial links to communicate with Expansion Interface (EI) circuit packs, and provides an RS-232 interface for connection to an administration terminal. Each circuit pack contains a 3-position switch to control emergency power transfer.

The TN775 contains a new DC to DC power converter and is used in maintenance to monitor the processor in an EPN that is supported by a Survivable Remote Processor (SRP).

TN780 Tone-Clock

The TN780 connects to and monitors an optional external Stratum 3 clock for digital frame timing. It also couples the clock output to local clocks. The TN780 supplies master timing to the system and produces the following tones: call progress, touch tones, answer-back, and trunk transmission test. It has 2 MHz, 160 kHz, and 8 kHz clocks. This circuit pack can transmit the system clock and tones on either TDM bus A, TDM bus B, or both.

The TN780 is administrable to produce 5 tone plans (for countries outside the United States) other than the United States tone plan. Six tones can be customized in each plan. The TN780 supports µ-Law or A-Law companding.

A TN780 is never used with the S8100 Media Server configurations.

TN787K Multimedia Interface

The TN787 multimedia interface circuit pack is used in conjunction with the TN788 multimedia voice conditioner circuit pack to provide service circuit functionality for the Multimedia Call Handling (MMCH) feature. This feature provides both voice and multimedia data service between multimedia complexes (endpoints). Up to 6 endpoints can conference to a single multimedia call occurrence.

The TN787 provides a TDM bus interface and a DS1 adjunct cable interface. The TN787 routes the H.221 multimedia information to the DS1 interface to free more TDM bus timeslots. This allows the system to carry more audio, video, and data bit streams between multimedia complexes (endpoints). The TN787 provides support for multiple PNs.

TN788C Multimedia Voice Conditioner

The TN788C Multimedia Voice Conditioner circuit pack is used in conjunction with the TN787F/G multimedia interface circuit pack to provide service circuit functionality for the MMCH feature. This feature provides both voice and multimedia data service between multimedia complexes (endpoints).

NOTE:
The TN788C vintage is µ-Law only. The TN788C vintage 2 or later is A-Law and µ-Law.

The TN788C is the audio processor for the Px64 multimedia conference bridge. The TN788C contains 8 digital signal processors; 4 for encoding and 4 for decoding. Each encoder/decoder pair is assigned to a Px64 endpoint to process its audio channel. Connection to and from the endpoint's audio is by way of a TN787 multimedia interface port. This connection is through the TDM bus timeslots.

Each of the 8 digital signal processors communicate with the main on-board processor through 8 individual Dual Port Random Access Memory (DPRAMs). No Read Only Memory (ROM) is available on this circuit pack; the DPRAM is used for program download.

TN789B Radio Controller

The TN789B circuit pack is an interface between a switch and 2 Wireless Fixed Base (WFB) radio units for the DEFINITY Wireless Business System. The TN789B contains a main processor to handle data line circuit (DLC) and upper medium access (MAC) layers of firmware. The TN789B also contains 2 lower MAC processors; 1 for each radio interface. Each radio interface is referred to as an I2 interface.

The I2 link is the connection between the Radio Controller (RC) and the WFB. The RC supports up to two I2 links, each link consisting of 4 pairs of twisted-pair cable: the transmit pair, the receive pair, and the local power pair. The transmit pair transfers WFB control and frame information from the RC to the WFB. The receive pair transfers status and frame information from the WFB to the RC. If the RC cannot provide power to the WFB, a third pair (to the WFB) can supply local power. When possible, the transmit pair and the receive pair provide phantom power from the RC to the WFB.

Each circuit pack includes a standard TDM bus interface from a system, 2 radio interfaces to 2 separate radio units, and 2 synchronization ports. In addition, 2 RS-232 interfaces provide for a debug terminal and for setting up the wireless terminal. Provides an interface to Wizard II base stations (DWBS).

TN791 Analog Guest Line (16 ports) (International Offers or US and Canada Offer B only)

The TN791 has 16 ports. Each port supports one telephone, such as 500 (rotary dial) and 2500 terminals (DTMF dial). LED and neon message waiting indicators are supported (a separate power supply is required for neon message indicators).

The TN791 supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED and neon message waiting indicators.

The TN791 supports 3 ringer loads, only one telephone can have an LED or neon message waiting indicator. The TN791 supports up to 8 simultaneous ports ringing; 4 on ports 1 through 8, and 4 on ports 9 through 16.

The TN791 supports A-Law and µ-law companding and administrable timers. Secondary lightning protection is provided. The TN791 supports up to 8 simultaneous ports ringing.

The TN791 is compatible with the telephone configurations shown in the table below.

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
6200-Type	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

Telephone	Wire Size	Maximum Range
7100-Series	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
7101A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
7103A-Series	24 AWG (0.2 mm^2/0.5 mm)	15,200 ft. (4,633 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
9100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

TN792 Duplication Interface for TN2404 Processor for SI

In high- and critical-reliability SI systems, duplication interfaces copy the contents of memory from the primary Switch Processing Element (SPE) to a standby SPE, so that the latter can take over immediately when the former fails. The TN792 uses the Enhanced M-Bus of the TN2404 processor for this memory shadowing function. The Enhanced M-bus supports 32-bit addressing and data access (vs. 16-bit for the M-bus), so it transfers data faster and shadows a larger area of memory than the older bus. The M-bus is still supported, however, for legacy applications.

You need two TN792 circuit packs, one for the primary control carrier and one for the standby. You can replace TN772 duplication interfaces with TN792s, but you must replace them in pairs. A TN772 cannot communicate with a TN792.

A new duplex optical cable connects the TN792 circuit packs. This cable eliminates the additional electromagnetic emissions that would otherwise result from the doubled data rate on the bus. The optical cable interface to the new DUPINT is on the front faceplate of the circuit pack.

The TN792 is compatible with existing duplication cables.

TN793B Analog Line with Caller ID (24 ports)

The TN793B is a dual coded, analog line 24-port circuit pack that supports Caller ID telephones and Caller ID devices. Each port supports one telephone, such as 500 (rotary dial) and 2500 terminals (DTMF dial). Use TN793B Vintage 1 or greater.

The TN793B supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED and neon Message Waiting Indicators. The TN793B supports off-premises wiring (out-of-building only with certified protection equipment) with either DTMF or rotary dialing, but LED or neon message waiting indicators are not supported off -premises.

The TN793B, along with a TN755B neon power unit per carrier or per single-carrier cabinet, supports telephones equipped with neon message waiting indicators (on-premises use only). The TN793B supports 3 ringer loads, only 1 telephone can have an LED or neon message waiting indicator. The TN793B allows a maximum of 12 simultaneous ports ringing; 4 on ports 1 through 8, 4 on ports 9 through 16, and 4 on ports 17 through 24.

The TN793B supports A-Law and µ-law companding and administrable timers. The TN793 supports queue warning level lights associated with the DDC and UCD features, recorded announcements associated with the Intercept Treatment feature, and PagePac paging system for the Loudspeaker Paging feature. Additional support is provided for external alerting devices associated with the TAAS feature, neon message waiting indicators, and modems The TN793B also supports secondary lightning protection. The TN793B provides -48 V DC current in the off-hook state. Ringing voltage is -90 V DC.

The TN793B is compatible with the telephone configurations shown in the table below.

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft (6,096 m)
6200-Type	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
7100-Series	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
9100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

TN797 U.S. Analog Trunk or Line Circuit Pack (8 ports)

Provides a combination 8-port Analog Trunk and Line circuit pack for the US, Canada and countries that have the same analog standards. The TN797 provides you with the capability to administer any of the 8 ports of this circuit pack as a CO trunk (loop or ground start), a CAMA E911 trunk, a DID trunk (either wink or immediate start), or as an analog line (on or off-premises and/or with or without LED Message Waiting Indication. It does not support ICLID on the analog trunk to the CO, nor Caller ID on the line side to the telephone.

TN799DP Control LAN (C-LAN) Interface

The TN799DP provides TCP/IP connectivity over Ethernet or PPP (Point to Point Protocol) connections to applications such as CentreVu CMS, INTUITY, AUDIX, DCS, printers, call accounting/CDR, and Property Management Systems (PMS). The TN799DP is required to support the TN2302AP IP Media Interface for H.323 functionality, and to support ATM interfaces and WAN PNC.

The TN799DP operates at 10Mbps or 100Mbps, full or half duplex, both of which are administrable. The TN799DP provides connectionless UDP sockets for IP solutions support. The circuit pack will also support 500 remote sockets, with support for 4Kbyte UDP sockets. The TN799DP supports variable length ping and the **traceroute** and **netstat** network testing commands.

The C-LAN circuit pack, TN799DP provides call control for all IP endpoints connected to the S8700 Media Server for IP Connect. There is a maximum number of 64 C-LANs per configuration. The number of C-LANs required depends on the number of devices connected as well as the options being utilized by the endpoint. It may be advantageous to segregate IP voice control traffic from device control traffic as a safety measure.

The default value for C-LAN socket usage of H.323 Tie Trunks is determined by dividing the total number of H.323 Tie Trunk utilizing sharing by 31. Each IP endpoint requires the use of some number of C-LAN sockets, which is the software object used to connect a TN799 board to the IP Network. The TN799DP board supports up to 500 sockets.

The C-LAN differs from an IP Media Processor in that the former controls the call and the latter provides the codecs used for the call's audio.

To take advantage of downloadable firmware capability there must be at least one TN799DP C-LAN and access to the public Internet for firmware downloads to other downloadable circuit packs.

Downloads and instructions have been posted to:

http://www.avaya.com/support/
Click on Online Services and then Download Software needed.

TN801 MAPD (LAN Gateway Interface)

The TN801 is part of the Multi-Application Platform DEFINITY (MAPD). It allows direct integration of PC-based application into the switch. This circuit pack works as the interface for such solutions as CTI, CallVisor and PC/LAN. The TN801 provides packet bus and TDM bus interfacing, physical mounting for a CPU, external interfaces, and mapping of circuit switched connections between the TDM bus and the expansion circuit pack.

TN802B MAPD (IP Interface Assembly)

The TN802 Internet Protocol Interface circuit pack supports voice and fax calls from the switch across a corporate intranet or the Internet. This circuit pack is still supported, but has been replaced by the TN2302AP IP Media Processor, the TN2302AP IP Media Processor. The IP trunking software runs on an embedded PC operating under Windows NT. The TN802 supports IP Solutions including IP Trunking and MedPro (H.323) with IP Softphones.

It is implemented using the TN802 IP Interface, which is a Windows NT server that resides on the IP Interface circuit pack inside the switch. The TN802 IP Interface operates in two modes: IP Trunk and Media Processor (MedPro/H.323). The TN802 defaults to IP Trunk mode. To use it in MedPro mode, you activate it through administration to use the H.323 trunking feature related to IP Softphones. For these features, you also must install a new TN799 C-LAN circuit pack.

TN1648B System Access and Maintenance (SYSAM) for R

The TN1648B is an SPE component used for maintenance. A processor in the TN1648B runs control routines that connect to maintenance software. The TN1648B has a 5-LED alarm panel and a toggle switch to manually inhibit automatic emergency transfer of PPN analog lines.

The TN1648B provides 2 RS-232 interfaces for connection to an administration terminal and, with duplication, connection to a standby maintenance terminal. The TN1648B also provides a tip and ring port with a built-in modem allowing a remote administration terminal to access the system.

The circuit pack allows connection to the management terminal and connection to and termination of 1 end of the processor bus. Other maintenance items include:

- PPN alarm monitors and outputs including auxiliary alarms for auxiliary equipment and environmental monitoring for over- temperature conditions
- Power supply sense and control for the carriers
- Emergency transfer control for catastrophic failure. The circuit consists of a 3-position toggle switch. The switch may be moved into the manual on position, manual off position, or automatic position.
- Non-volatile memory containing the time-of-day clock, the INitialization and ADministration System (INADS) telephone number, login password, and product identification
- Alarm panel information with major, minor, and warning LEDs, an ACKnowledge LED, and an emergency transfer LED

The TN1648B increases the internal and external modem speeds from 2400 bps to 9600 bps with error control implementation. Appropriate software administration is added to choose between external modem access to the maintenance port, and direct loop start trunk access. The former is used in countries that do not support the analog loop signaling protocol supplied by the internal modem.

For non-U.S. use, a cable from the TN1648B connects directly to an external modem. When the external modem is selected, the internal modem is disabled. The TN1648B is backward-compatible with the TN1648.

TN1650B Memory for R

The TN1650B memory circuit pack contains 32 MB of Dynamic Random Access Memory (DRAM) and error detection and correction circuitry to ensure information integrity. The TN1650B is used for system software, customer translations, and call processing maintenance. All new systems are shipped with four memory boards and a TN1657 disk drive. All upgrades to R10 require a fourth memory board and a TN1657 disk drive V9 or later.

TN1654 DS1 Converter, T1 (24-Channel) and E1 (32-Channel)

The TN1654 converter complex installs in place of the conventional fiber and supports from 1 to 4 T1 or E1 facilities, providing a total of 92 T1 channels (or 120 E1 channels) in each direction between the PPN and the EPN. This capacity is enough for an EPN to easily support several hundred stations.

The switch architecture provides for EPNs that are remotely located from the PPN. EPNs within 5 miles (8 km) of the PPN may be coupled using multimode fiber optic cable or within 22 miles (35.4 km) using single-mode fiber optic cable. When the distance between the PPN and the EPN exceeds 5 miles (8 km) (multimode), or 22 miles (35.4 km), single-mode, or private right-of-way is not available, connect using a DS1 converter complex. A DS1 converter complex consists of a pair, 1 at each end, of DS1 converter circuit packs and the associated T1/E1 facilities.

A new set of Y-cables is required by the TN1654 to connect to a TN570B Expansion Interface circuit pack for system communication.

TN1655 Packet Interface for R

The TN1655 provides the communication path between the SPE and the packet bus in the PPN. The packet bus connects to EI circuit packs in the PPN that communicate with EPNs and the CSS.

The TN1655 provides the link access procedure on the D-channel (LAPD) for up to 8,192 links at a sustained rate of 2Mbits per second. These are the digital multiplexed interface (DMI) mode-3 terminations of communication links across the packet bus that link to the processor circuit pack. Some data communication will use the X.25 data phase protocol at level 3.

The TN1655 provides termination for ISDN-BRI and ISDN-PRI signaling links, expansion archangel links connecting the processor to the expansion archangels on EI circuit packs in each PN, and center stage control network links connecting the processor with SNI circuit packs in the CSS.

The TN1655 supports firmware downloading. It also provides X.25 termination to the DCS links and adjuncts such as CDR and AUDIX.

TN1657 Disk Drive for R

The TN1657 contains a 180 MB small computer system interface (SCSI) disk drive. The TN1657 reduces the boot time of the system, stores translations, bootstrap image, and core dumps. Vintage 9 or later is required.

TN2138 Central Office Trunk for Italy (8 ports)

The TN2138 provides 8 analog loop start CO trunk ports. Each port has tip and ring signal leads. The TN2138 has 50 Hz, 12-kHz, and 16-kHz Periodic Pulse Metering (PPM).

TN2139 Direct Inward Dialing Trunk for Italy (8 ports)

The TN2139 provides 8 analog direct inward dialing (DID) trunk ports for analog DID signaling. Each port has tip and ring signal leads.

TN2140B Tie Trunk for Hungary and Italy (4-wire, 4 ports)

The TN2140B provides 4 ports for 4-wire E & M lead signaling tie trunks. The TN2140 provides continuous E & M signaling and discontinuous E & M signaling. Administrable A-Law and µ-Law companding and standard Type 1 and Type 5 signaling is provided. The TN2140B is required for Hungary.

TN2146 Direct Inward Dialing Trunk for Belgium and the Netherlands (8 ports)

The TN2146 provides 8 analog DID trunk ports. Each port has tip and ring signal leads. The TN2146 uses 4 (1 for each pair of ports) Dual Subscriber Line Audio processing Circuits (DSLACs) administered to meet trunk transmission characteristics. The DSLACs can be set to either a resistive or complex balance impedance in the voice or AC talk path on the trunk interfaces. The DSLACs convert analog signals to digital signals and vice-versa to match the analog DID trunks to the system's digital TDM bus.

Companding can be set to either A-Law or µ-Law.

TN2147C Central Office Trunk for Multiple Countries (8 ports)

The TN2147 has 8 analog CO trunk ports. Each port has tip and ring signal leads. The TN2147 uses 4 (1 for each pair of ports) Dual Subscriber Line Audio processing Circuits (DSLACs) to be administered to meet a given transmission and impedance requirement. The DSLACs convert analog signals to digital signals and digital signals to analog signals to interface the analog CO trunks to the system's digital TDM bus.

The TN2147C provides multi-country signaling based on trunk type: loop-start, ground start, or battery reverse loop-start.

TN2181 DCP Digital Line (2-wire, 16 ports)

The TN2181 has 16 DCP ports that can connect to 2-wire terminals such as the 6400, 8400, and 9400 Series digital telephones and the 302C and 302D Attendant Console. The maximum range of the 8400 and 9400 Series terminals using 24 AWG (0.5 mm) wire is 3,500 feet (1067 m).

The TN2181 supports either A-Law or µ-Law companding (software selectable). The TN2181 also supports the 8400 Series data modules.

TN2182C Tone-Clock, Tone Detector, and Call Classifier (8 ports)

The TN2182 tone-clock integrates the tone generator, tone detection-call classifier, system clock, and synchronization functions onto one circuit pack for all system reliability configurations. The TN2182 supports 8 ports for tone detection and allows gain or loss applied to PCM signals received from the bus. The TN2182 provides Stratum 4 enhanced clock accuracy, supports MFC signaling (such as Russia MF), supports Russia MFR (multi-frequency shuttle Register signaling), and supports A-Law and µ-Law companding.

The TN2182CP provides continuous, cadenced and mixed tones, allows administrable setting of tone frequency and level, detects 2025 Hz, 2100 Hz, or 2225 Hz modem answerback tones, and provides normal and wide broadband dial tone detection.

In most configurations, the 2- or 3-circuit pack combination of tone generator, tone detector, and/or call classifier can be replaced with this one circuit pack, freeing 1 or 2 port slots.

Use the TN2182CP with the TN429D analog line central office trunk for CAMA/E911 and ICLID. A TN2182 is required for the on-board tone detection or for additional tones to support CCRON, Russian ANI, and others.

TN2183/TN2215 Analog Line for Multiple Countries (16 ports)

See "TN2215/TN2183 Analog Line for Multiple Countries (16 ports) (International Offers or US and Canada Offer B only)" on page 325.

TN2184 DIOD Trunk for Germany (4 ports)

The TN2184 Direct Inward/Outward Dialing (DIOD) trunk contains 4 port circuits, each interfacing a 2-wire analog CO trunk with the TDM switching network of the system. Each port allows incoming and outgoing calls to include addressing information being received from the CO for incoming calls and addressing information being sent to the CO for outgoing calls. It detects Periodic Pulse Metering (PPM) signals for call charge accounting on outgoing calls.

The TN2184 combines the features of both a CO trunk and a DID trunk to provide both outgoing and incoming calls with addressing information in both directions.

TN2185B ISDN-BRI S/T-TE Interface (4-wire, 8 ports)

The TN2185B supports eight 4-wire ISDN BRI line S interfaces, each operating at 192 Kbps, with 2 B-channels (64 Kbps) and 1 D-channel (16 Kbps). The TN2185B interfaces to the LAN bus and the TDM bus to provide the TE side of the BRI interface.

The TN2185B is similar to the TN2198 except the TN2185B is a 4-wire S interface instead of a 2-wire U interface. Another difference is the function of the SCOTCH/NPE and SAKI are replaced by the network control element (NCE).

For each port, information communicates over two 64 Kbps bearer channels called B1 and B2, and over a 16 Kbps channel called the demand channel, or D channel. The D channel is used for signaling. Channels B1 and B2 can be circuit switched simultaneously, or either of them may be packet switched, but not both at once. The D channel is always packet switched. The circuit switched connections have a µ-Law or A-Law option (on a per-board basis) for voice operation and operate as 64 kbps clear channels when in the data mode. The packet switched channels support the LAPD protocol, however, the TN2185B does not terminate on LAPD protocol. The S interface does not support switching of both B channels together as a 128 Kb/s wideband channel.

The TN2185B has a long range up to 18,000 feet (5486 m) maximum from the system to the NT1 device. In a multiple terminal environment, the B channels are shared only on a per-call basis. For example, if the B2 channel is for data, then the B2 by 1 terminal excludes the others from having access to it. When a used device communicates over the D channel to access B1 or B2, that channel is owned until the call is taken down (no party on line). The D channel is always shared among the terminals. This circuit pack can be used as an alternative to the TN464 or the TN2464.

The ability of outpulse in-band DTMF signals or end-to end signaling is supported by the TN2185B.

QSIG Call Completion is supported, however, QSIG Supplementary Services are not. ISDN-BRI trunks can be used as inter-PBX tie lines using the QSIG peer protocol.

TN2198B ISDN-BRI U Interface (2-wire, 12 ports)

The TN2198 allows connection to the ANSI standard 2-wire U-Interface. The 2-wire interface from the TN2198 connects to an NT1 network interface. The 4-wire interface on the other side of the NT1 may connect to 1 or 2 telephones. The TN2198 does not provide a trunk-side interface as does the TN2185 circuit pack.

The TN2198 contains 12 ports that interface at the ISDN U reference point. For each port, information communicates over two 64 Kbps bearer channels called B1 and B2, and over a 16 Kbps channel called the demand channel, or D channel. The D channel is used for signaling. Channels B1 and B2 can be circuit switched simultaneously. The D channel is always packet switched. The TN2198 requires a packet control circuit pack. Each port supports 1 telephone, such as 500 (rotary dial) and 2500 terminals (DTMF dial).

The D channel supports the LAPD protocol and is consistent with the CCITT Q.920 Recommendations for D channel signaling.

In a multiple terminal environment, the B channels are shared only on a per-call basis. For example, if the B2 channel is used for data, then the use of B2 by 1 terminal excludes the others from having access to it. When a device communicates over the D channel to access B1 or B2, that channel is owned until the call is taken (no party on line). The D channel is always shared among the terminals. The TN2198 interfaces with the TDM bus and packet bus in the switch backplane and terminates with 12 ISDN basic access ports.

The TN2198 has a long range to 18,000 feet (5486 m) maximum from the system to the NT1 device and uses standard protocol ANSI T1.601. The TN2198 has a 160 Kbps line rate, consisting of 2 bearer channels at 64 Kbps channels each, the D channel at 16 Kbps, framing at 12 Kbps, and maintenance at 4 Kbps. The circuit pack uses demand channel (ringing, tone, and so forth), 16 Kbps channel, and supports up to 24 telephones or data modules per circuit pack.

The TN2198 is not offered as a BRI Tie Trunk.

TN2199 Central Office Trunk for Russia (3-wire, 4 ports)

The TN2199 is a 4-port, 3-wire, loop-start trunk circuit pack that can be used as a DID trunk, CO trunk (2-way, 1-way incoming, or 1-way outgoing CO). The TN2199 combines the functionality of the DID trunk and the 1-way outgoing CO trunk (DIOD trunk). This circuit pack is used with the call classifier circuit pack (TN744D or later) or the Tone-Clock/ tone detector (TN2182B or later) to accomplish MF shuttle signaling (similar to R2-MFC signaling). Supports incoming ANI.

TN2202 Ring Generator for France

The TN2202 ring generator circuit pack supplies 50 Hz ringing power. The TN2202 supplies balanced ringing, by a modified backplane, to terminals connecting to the TN2183 multi-country analog line circuit pack when administered for France analog transmission.

The TN2202 plugs into the power unit slot and is required for each carrier containing analog lines (1 in an single-carrier cabinet or 1 in each port carrier of a MCC1). A 1-lead modification is required in each backplane using TN2202. This is true for all products made for France. The TN2202 produces 2 symmetrical voltages (typically 28V rms) with respect to ground, and takes –48 VDC, –5 VDC, and ground from the backplane and generates 2 x 28V rms with added –48 VDC.

TN2207 DS1 Interface, T1 (24-Channel) and E1 (32-Channel), for MMCH

The TN2207 circuit packs support digital signal level 1 (DS1) rate (24 channel) and E1 rate (32 channel) digital facility connectivity. All TN2207 suffixes support CO, Tie, DID, and off premises station (OPS) port types using the robbed-bit signaling protocol, the proprietary Bit-Oriented Signaling (BOS) 24th channel signaling protocol, and the DMI-BOS 24th channel signaling protocol. The circuit packs also support ISDN-PRI connectivity T1 or E1.

In DS1 (24 channel) mode, a DS1 interface is provided to the DS1 facility. The TN2207 circuit packs provide board-level administrable A-Law and µ-Law companding, CRC-4 generation and checking (E1 only), and Stratum 3 clock capability.

The TN2207 provides test jack access to the DS1 or E1 line and support the 120A integrated channel service unit (CSU).

All suffixes have line out (LO) and line in (LI) signal leads. The Line Out and Line In leads are unpolarized balanced pairs.

The TN2207 has additional hardware to support direct cabling to a TN787 MMI circuit pack.

TN2209 Tie Trunk for Russia (4-wire, 4 ports)

The TN2209 has 4 ports used for Type 1 or Type 5 four-wire E & M lead signaling tie trunks, that can be automatic, immediate-start, wink-start, and delay-dial. The TN2209 provides an interface between these four frequency signaling tie trunk lines and the switch TDM network. Based on a TN760D each port has modified E & M signal leads for universal hardware compatibility. The TN2209 provides release link trunks required for the CAS feature and has administrable A-Law and µ-Law companding.

TN2211 Optical Drive for R

The TN2211 Optical Drive provides removable storage for software upgrades, translation backups, announcement file backups, core dumps, and so on. The Optical Drive has several advantages over tape drive technology:

- Writing to the optical disk is considerably shorter than writing to tape. A full backup will take approximately 20 minutes, whereas the tape drive takes up to 95 minutes.
- Writing to the optical disk is more reliable due to:
    - block allocation (has several spare blocks available)
    - number of accesses to the media

The UN332C MSSNET must be used with the TN2211.

TN2214B DCP Digital Line (2-wire, 24 ports) (International Offers or US and Canada Offer B only)

The TN2214 has 24 DCP ports that can connect to 2-wire digital telephones such as the 6400-, 8400-, or 9400-Series telephones and the 302C/302D Attendant Console.

The TN2214 supports either A-Law or µ-Law companding (software selectable) and supports the telephones listed below.

Telephone	Wire Size	Maximum Range
302C/D Attendant Console	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)
6400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)
8400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067 m)
9400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067 m)

A new version of the TN2214 will be released in the future as a firmware downloadable circuit pack. When this occurs, the circuit pack suffix will have a “P” appended to the circuit pack number.

TN2215/TN2183 Analog Line for Multiple Countries (16 ports) (International Offers or US and Canada Offer B only)

The TN2215/TN2183 provides 16 analog port interfaces. Each port supports one telephone, such as 500 (rotary dial) and 2500 telephones (DTMF dial) from a tip/ring pair. Each port also sends or receives signaling to and from a device, such as an analog telephone, answering machine, facsimile and loop-start CO port AUDIX. The TN2215/TN2183 provides rotary digit 1 recall, ground-key recall, and programmable flash timing. Additional support is provided for selectable ringing patterns, LED message waiting, and secondary lightning protection.

The TN2215/TN2183 supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED message waiting indicators. It supports off-premises wiring (out-of-building only with certified protection equipment) with either DTMF or rotary dialing, but LED message waiting indicators are not supported off-premises. Neon message waiting indicators are not supported.

A maximum of 6 to 8 simultaneous ports ringing is allowed depending on the ringing cadence selected. The TN2215/TN2183 supports A-Law and µ-Law companding and administrable timers.

The TN2215/TN2183 also supports balanced ringing (when configured for France with TN2202 ring generator circuit pack) and DTMF sending levels appropriate for CONVERSANT.

The TN2215/TN2183 is impedance and gain selectable for multiple countries. For more information, contact your Avaya representative.

The TN2215/TN2183 supports the following telephones:

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
6200-Type	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
7102A-Series	24 AWG (0.2 mm^2/0.5 mm)	3,100 ft. (945 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

TN2224B DCP Digital Line (2-wire, 24 ports)

The TN2224 has 24 DCP ports that can connect to 2-wire digital telephones such as the 6400-, 8400-, or 9400-Series telephones and the 302C/302D Attendant Console.

The TN2224 supports either A-Law or µ-Law companding (software selectable) and supports the telephones listed below.

Telephone	Wire Size	Maximum Range
302C/D Attendant Console	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)
6400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)
8400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)
9400-Series	24 AWG (0.2 mm^2/0.5 mm)	3,500 ft. (1,067) m)

A new version of the TN2224 will be released in the future as a firmware downloadable circuit pack. When this occurs, the circuit pack suffix will have a “P” appended to the circuit pack number.

TN2242 Digital Trunk (Japan 2MB TTC)

The TN2242 supports versions of Channel Associated Signaling and ISDN-PRI signaling that are peculiar to the TTC private networking environment used in Japan. It supports the special line-coding and framing used on 2.048-Mbps Japanese trunks. The TN2242 connects the switch

with other vendor equipment and with other DEFINITY and MultiVantage configurations via the TDM device that is commonly used throughout Japan for this purpose.

TN2301 Logic Switch for R

Provides service to the customer when the link to the main processor fails or is severed, or when the processor or Center Stage Switch fails. The TN2301 Survivable Remote Switch (SRS) circuit pack connects the EPN links (fiber or T1/E1) to the appropriate PPN for call processing. It does this under control of the TN775C maintenance circuit pack which monitors the health of the expansion interface TN570B.

This circuit pack is not used in an ATM-PNC.

TN2302AP IP Media Processor

The TN2302AP provides VoIP (voice over internet protocol) audio access to the switch for local stations and for outside trunks. The TN2302AP can perform echo cancellation, silence suppression, fax relay service, and DTMF detection. The TN2302AP is the H.323 audio platform, includes a 10/100 BaseT Ethernet interface, supports the T.30 and T.38 standards for fax transmission, and is firmware downloadable.

The TN2302AP provides audio processing for between 32 and 64 voice channels, depending on the CODECs in use. It supports hairpin connections and shuffling of calls between TDM connections and IP-IP direct connections.

TN2305B ATM-CES Trunk/Port-Network Interface for Multi-Mode Fiber (Not available with Category B)

The TN2305 provides an ATM-based replacement for the TN570D Expansion Interface expansion interface. This interface uses OC-3c or STM-1 155-Mbps multimode fiber. The TN2305 supports both trunk and port-network connectivity. As a trunk, the TN2305 uses Circuit Emulation Service (CES) to emulate up to 8 ISDN-PRI trunks on an ATM facility. As a port-network expansion interface, the TN2305 connects port networks to an ATM switch that provides port network connectivity. The TN2305 provides echo cancellation.

The TN2305 does not support hybrid port-networks that use both ATM and CSS simultaneously. TN2305s must connect all port networks through the ATM switch. Direct connect EPNs are not supported.

DEFINITY systems and MultiVantage configurations may contain a Class 1 LASER device. The LASER device operates within the following parameters:

- Maximum Power Output: -8 dBm
- Wavelength: 1310 nm
- Mode Field Diameter: 8.8 microns

CLASS 1 LASER PRODUCT
IEC 825 1993

CAUTION:
Use of controls or adjustments or performance of procedures other than those specified herein may result in hazardous radiation exposure.

TN2306B ATM-CES Trunk/Port-Network Interface for Single-Mode Fiber (Not available with Category B)

The TN2306 circuit pack has the same features as the TN2305B ATM-CES Trunk/Port-Network Interface for Multi-Mode Fiber (Not available with Category B) but supports single-mode fiber.

TN2308 Direct Inward Dialing Trunk for Brazil (8 ports)

The TN2308 uses 8 ports for immediate-start and wink-start direct inward dialing (DID) trunks. Each port has tip and ring signal leads.

The switch requires the TN2308 to support Brazil Block Collect Call. The TN2308 transmission characteristics comply with Brazilian telecom standards for PBXs.

TN2312AP IP Server Interface

The TN2312AP IP Server Interface (IPSI) provides the S8700 Media Servers with tone generation, tone detection, call classification equivalent to a TN2182C, and clock generation. Each IPSI will have one 10/100BaseT Ethernet connection to the Ethernet switch.

The IPSI has the following characteristics:

- Always resides in the tone/clock slot II
- Has 10/100 BaseT interface for connection to server
- Has 10/100 BaseT interface for connection to services laptop
- Supports 8 global Call Classification ports
- Supports network diagnostic capabilities
- Provides PN clock generation and synchronization (Stratum 4 type II only)
- Provides PN tone generation
- Provides PN tone detection/global call classifier/international protocols

- Provides distributed PN packet interface
- Supports IPSI firmware download
- One IPSI per configuration provides serial number support for License File Feature Activation

IPSI IP addresses are typically assigned automatically using DHCP service. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance and firmware downloads.

TN2313AP DS1 Interface (24-Channel)

The TN2313AP DS1 port board interfaces a DS1 trunk to the switch backplane via port slots that are standard for DEFINITY and MultiVantage products. The TN2313AP is compatible with previous 24-channel DS1 circuit packs, including the TN464F (V19 and below), the TN2464 (V19 and below), and the TN767E DS1, except that it does not provide for packet adjunct capabilities. The TN2313AP supports a variety of applications, including networking of DEFINITY and MultiVantage solutions, international trunk types, video teleconferencing, and wideband data transmission.

The TN2313AP DS1 interface can be configured for 24-channel, 1.544 Mbps. The TN2313 can supply two 8-Khz reference signals to the switch backplane for optional use by the tone/clock board in synchronizing the system clock to the received line clock.

The TN2313AP is firmware downloadable.

TN2314 S8100 Media Server

The S8100 Media Server supports voice stations with co-resident voice switching, voice and fax messaging and system applications that are running on a Microsoft Windows 2000 operating system. The communication between the firmware and the software is done by an ethernet connection. An Intel processor Message Link or IML is the ethernet control link between the Pentium processor and the MPC860 processor. This allows for the message based communication between the two processors.

The S8100 Media Server has the following characteristics:

- Processor – The processor is a 500 MHz Pentium III.
- RAM – There are two slots for SDRAM memory modules, with a minimum of 256-MB and a maximum of 512-MB.
- Front panel ethernet access – Services can access the switch via an RJ45 ethernet jack added to the circuit pack faceplate.
- Hard disk – The circuit pack has a 20-GB hard disk.

TN2401 Network Control/Packet Interface for SI

The TN2401 Net/Pkt interface circuit pack provides the network control interface (NETCON), the packet interface (PACCON), and, if BX.25 connectivity is not required, the processor interface (PI). The TN2401 provides eight asynchronous data channels. The TN2401 does not include modems. The TN2401 is required for the SI model to save translations to the 5-volt ATA flash memory card.

TN2401/TN2400 Net/Pkt Interface sandwich board assembly for SI/I upgrades

The TN2401/TN2400 Net/Pkt interface sandwich board circuit pack provides the network control interface (NETCON), the packet interface (PACCON), and, if BX.25 connectivity is not required, the processor interface (PI). The TN2401/TN2400 provides eight asynchronous data channels. The TN2401/TN2400 does not include modems. The TN2401/TN2400 is required for the SI model to save translations to the 5-volt ATA flash memory card.

The sandwich board assembly and the TN2404 processor is required for the following upgrades:

- A G1 or G3iV1 MCC1 with a TN773 Processor
- An S, I, OR SI configuration with a TN786B Processor when reusing the existing control carrier cabinet
- An SI configuration with a TN790 or 790B Processor. Any R5 or R6 configuration will have the old control carrier backplane and will require the TN2401/TN2400. For R7 and R8 configurations, it is possible to have the old control carrier backplane or the new control carrier backplane. It is critical that the backplane type is verified prior to placing the upgrade order so that the right characteristic selection can be made. If the type of carrier is not known, then a visual inspection will be required of the R7 or R8 system. The old backplane is being used if the system has a TN794/TN2400 in the Network Control and Packet Control Slots. If nothing is in the Packet Control Slot then the new backplane exists.

TN2402 Processor for CSI

The TN2402 processor platform runs at 25 MHz, includes a 32-bit RISC CPU complex and a maintenance processor complex providing serial communications and maintenance functions. In addition, the TN2402 terminates ISDN LAPD signaling over the TDM bus from PRI and BRI trunk circuit packs.

The RISC CPU complex provides 4 to 32 MBytes of Flash PROM. The DRAM is provided via one SIMM. The TN2402 contains 32 MBytes of DRAM. The flash is not interleaved.

The TN2402 processor does not provide X.25 communications, nor does it provide a duplication option. The TN2402 does not contain an on-board modem. Instead, an external modem must be connected to the RS-232E port previously used for the internal modem.

The TN2402 is required for the CSI model to save translations to the 5-volt ATA flash memory card.

TN2404 Processor for SI

The TN2404 processor circuit pack has 32 MB of DRAM memory and flash memory. It is designed to handle errors associated with the EM-BUS and must be used with the C-LAN (TN799) and the Net/Pkt (TN2401) in SI systems.

TN2464BP DS1 Interface with Echo Cancellation, T1/E1 (International Category A or B)

The TN2464BP DS1 circuit pack has echo cancellation circuitry and firmware download capability. The TN2464BP supports T1 (24-channel) and E1 (32-channel) digital facilities. In ISDN-PRI applications, the ISDN D-channel connects the TN1655 Packet Interface for R packet interface via the LAN bus. The TN2464BP has the same functionality as the TN464GP, which is for US and Canada offers only.

The TN2464BP circuit pack provides:

- Test jack access to the T1/E1 line
- Board-level administrable A-Law and µ-Law companding
- CRC-4 generation and checking (E1 only)
- Stratum 3 clock capability
- Support for the 120 A Channel Service Unit Module
- CO, TIE, DID, off-premises station (OPS) port types that use robbed-bit signaling protocol, proprietary bit-oriented signaling (BOS) 24th-channel signaling protocol, or DMI-BOS 24th-channel signaling protocol
- Unpolarized, balanced-pair, line-out (LO) and line-in (LI) signal leads
- Support for Russian incoming ANI
- Support for the enhanced maintenance capabilities of the enhanced integrated channel service unit (ICSU)
- Support for CONVERSANT®
- Channel-associated signaling protocols for many countries (for details, contact your Avaya representative)

The TN2464BP can be updated using the firmware download feature, which requires use of the TN799 C-LAN interface.

TN2501AP Voice Announcements over LAN (VAL)

The TN2501AP is an integrated announcement circuit pack that:

- offers up to 1 hour of announcement storage capacity.
- provides shorter backup and restore times.
- is firmware downloadable.
- plays announcements over the TDM bus, similar to the TN750C.
- has 33 ports, including
    - 1 dedicated telephone access port for recording and playing back announcements (port number 1).
    - 1 ethernet port (port number 33).
    - 31 playback ports (ports 2–32).
- uses a 10/100 Mbps ethernet interface, allowing announcement and firmware file portability over your LAN (FTP server functions).
- uses announcement files that are in .wav format (CCITT A-Law and µ-law, 8 KHz, 8-bit mono).
- works in r, si, csi, and S8100 Media Server configurations.

TN2793B Analog Line with Caller ID (24 ports) (International Offers)

The TN2793B is a dual coded, analog line 24-port circuit pack. Each port supports one telephone, such as 500 (rotary dial) and 2500 terminals (DTMF dial). Use Vintage 1 or greater for the TN2793B.

The TN2793B supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED and neon message waiting indicators. The TN2793B supports off-premises wiring (out-of-building only with certified protection equipment) with either DTMF or rotary dialing, but LED or neon message waiting indicators are not supported off-premises.

The TN2793B along with a TN755B neon power unit per carrier or per single-carrier cabinet, supports telephones equipped with neon message waiting indicators (on-premises use only). The TN2793B supports 3 ringer loads, only 1 telephone can have an LED or neon message waiting indicator. The TN2793B allows a maximum of 12 simultaneous ports ringing; 4 on ports 1 through 8, 4 on ports 9 through 16, and 4 on ports 17 through 24.

The TN2793B supports A-Law and µ-law companding and administrable timers. The TN2793B supports queue warning level lights associated with the DDC and UCD features, recorded announcements associated with the Intercept Treatment feature, and PagePac paging system for the Loudspeaker Paging feature. Additional support is provided for external alerting devices associated with the TAAS feature, neon message waiting indicators, and modems. The TN2793B also supports secondary lightning protection. The TN793B provides -48 V DC current in the off-hook state. Ringing voltage is -90 V DC.

Telephone	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
2500-Type	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
6200-Type	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)
7102A-Series	24 AWG (0.2 mm^2/0.5 mm)	20,000 ft. (6,096 m)
8100-Series	24 AWG (0.2 mm^2/0.5 mm)	12,000 ft. (3,657m)

TNCCSC-1 PRI to DASS Converter

The TNCCSC-1 circuit pack converts ISDN-PRI to a DASS interface. DASS is a 2-Mbps interface using a 75-Ohm coaxial transmission facility. One TNCCSC-1 circuit pack can support up to 2 TN464 DS1 Interface circuit packs. A Y cable and an 888B 75-Ohm coaxial adapter connect to the public network facility.

TNCCSC-2 PRI to DPNSS Converter

The TNCCSC-2 circuit pack converts ISDN-PRI to a DPNSS interface. DPNSS is a 2-Mbps interface using a 75-Ohm coaxial transmission facility. One TNCCSC-2 circuit pack can support up to 2 TN464 DS1 interface circuit packs. A Y cable connects to the public network facility.

TNCCSC-3 PRI to DPNSS Converter

The TNCCSC-3 circuit pack is the same as the TNCSSC-2 with a 120-Ohm twisted pair interface.

TN-C7 PRI to SS7 Converter

Provides a gateway interface between the TN464 and the public signaling network. Integrates DASS, DPNSS and SS7 into a single board type. The TN-C7 is intended to support international service provider Call Center customers. It is not designed for operation in the US or Canada.

TN-CIN Voice/Fax/Data Multiplexer

The TN-CIN Provides QSIG and private networking transparency on demand across a switched network. The TN-CIN integrates up to three G.728 LD-CELP voice/fax circuits, or six CAFT voice/fax and two data circuits over a single subrate digital link. The three or six voice/fax circuits are presented as a G.703 E1 data stream using either QSIG peer-to-peer or channel-associated signalling.

All voice/fax circuits support low bit rate voice compression at 8Kbps to 16 Kbps when using CAFT, or at 16 Kbps when using LD-CELP. LD-CELP voice compression supports fax at V.29 (7200 bps). CAFT voice compression supports fax at V.27ter (4800 bps). The Composite port supports V.11/V.35 at speeds up to 128 Kbps.

The TN-CIN features an on-demand voice networking mode for use with time-based communications links like ISDN. A High Speed data port (V.24/V.11/V.35 at up to 115.2 Kbps synchronous or V.24 at up to 115.2 Kbps asynchronous), incorporating dynamic bandwidth allocation (variable data clocking), is available for data applications. A low speed V.24 data port (up to 96 Kbps synchronous or 57.6 Kbps asynchronous) is available for data applications.

UN330B Duplication Interface for R

In high- and critical-reliability configurations with 2 SPEs, 1 UN330B resides in each SPE and connects to the other UN330B. The UN330Bs provide control and communication paths between the SPEs to keep the standby SPE ready to assume control if the active SPE fails. The UN330Bs select active/ standby mode for the 2 SPEs, shadow (copy) the active SPE memory writes into the standby SPE memory, and support inter-SPE communications.

The duplication channel is a bi-directional high-speed path between the 2 SPEs. When memory shadowing is active, all shadowed memory writes on the active processor’s bus are sent across the link and written into the standby processor’s memory. Standby memory writes are not sent to the active processor.

UN331C Processor for r

The UN331C controls the system and executes stored programs that perform call processing activity and maintenance. The UN331C is a RISC designed around a MIPS R3000A CPU operating at 33 MHz. It employs 32-bit address and data buses to obtain and execute instructions at a rate approaching 1 instruction per clock cycle. The 256 kbyte instruction cache with burst-mode refill and 256 kbyte data cache are key to the performance of the processor. A read/write buffer chip tailors the UN331C to the call processing environment.

Peripheral devices residing on the UN331C are positioned outside the CPU cache structure and interface to the CPU through the read/write buffers. These peripherals include 512 kbytes of ROM for the monitor, counters/timers, UARTs, control/status/error registers, and the logic that provides bus arbitration and the Bus Time-Out feature.

The UN331C interfaces to the 32-bit multiplexed address/data processor bus (PM-Bus) and the 32-bit processor expansion bus (PX-Bus). The PM-Bus is for all processor write operations and single-word (4-byte) read operations. Multiple-word or burst reads are performed using the PM-Bus to transfer the address to main memory, then the words of the burst are returned using both the PM-Bus and PX-Bus.

UN332C Mass Storage/Network Control for R

The UN332 provides an interface between the UN331C processor and the small computer system interface (SCSI) bus for access to the mass storage system (MSS) such as a disk drive. The UN332 also provides TDM network control for the PPN, and terminates one end of the processor-multiplexed bus.

The UN332C allows the interface of the switch to the TN2211 optic disk drive. The circuit pack includes the SCSI Host Adapter, the ArchAngel and the Tone/Clock switching logic.

Phones and Endpoints

Avaya 2420 Digital Telephone

The new (5/1/02) 2420 is a two-wire digital telephone which provides the latest in Avaya technology. Using existing TN circuit packs as well as the new Media Module hardware the 2420 bridges the gap between old and new. Functions the 2420 include a handset, a twelve-button dial pad, Conference, Transfer, Drop, Hold, Redial, Mute, Volume Up/Down, Speaker, Headset button, dedicated Exit, Previous and Next buttons. The Mute, Speaker and Headset buttons have an associated red LEDs. Additionally the 2420 telephone has a separate Message button to access the MultiVantage auto-dial feature used to call the user's voice mail system and an associated high visibility LED Message Waiting Indicator.

The 2420 display consists of a FSTN (Film Super-Twisted Nematic) monochrome liquid crystal display (LCD) which is 29 characters wide by 7 lines tall. Display characters are defined by a 5 column by 8 row matrix of dots which is used to support 5 x 7 dot Eurofont or Katakana characters plus an underscore.

2420 Telephone Feature Summary:

- 7 line x 29 character LCD
- Handset and 12 button dialpad
- Adjustable viewing angle
- Wall mountable (without optional adjuncts)
- 8 general purpose buttons to access up to 24 system call appearance / features (3 pages of 8 buttons)
- Downloaded call appearance / feature button labels (paperless designation)
- 4 local softkey feature buttons
- Exit, Previous, and Next buttons for display navigation
- Highly visible Message Waiting Indicator
- Message button for expedited access to voice mail
- Conference, Transfer, Drop, Hold, Redial buttons
- Headset jack (separate from the handset jack)
- Built-in Speakerphone with Group Listen operation 12
- Speaker, Headset, Mute buttons, each with LED indicators
- Volume Up/Down buttons for Handset, Headset, Speakerphone, Ringer
- 100 Entry Call Log (total incoming answered, incoming unanswered, and outgoing calls)
- Up to 104 -Entry Local Directory / Dialer
- Downloadable firmware (flashware) for future upgrades
- Automatic Gain Control on all audio interfaces

- Electronically stored part ID and serial number for use with Automatic Customer Telephone Rearrangement
- Optional 24 button Feature Key Expansion Unit
- Optional application modules: Analog Interface

2500/2554 Analog Terminals

The 2500/2554 Series telephones are made up of several analog telephones that are basically the same but are equipped with small different attributes. The models include the 2500 MMGN, 2500 MMGL, 2500 YMGL, 2500 YMGP, 2500 MMGM and the 2500 YMGM desk model phones and the 2554 MMGN,2554 MMGM, 2554 YMGP, and 2554 YMGM wall model telephones. All models are single appearance analog telephones with conventional touch-tone dialing. The 2500 MMGL, 2500 MMGM and 2554 MMGM telephones have NO feature buttons. The 2500 YMGL, 2500 YMGM, 2554 YMGP, and 2554 YMGM telephones are equipped with a FLASH button, a Message Waiting light, a REDIAL button, a HOLD button and a MUTE button. All six of these telephone models provide access to features through the use of the * or # dial pad keys and the appropriate feature access codes.

Features on the 2500 YMGL, 2500 YMGM, and 2554 YMGM Telephones

The 2500 YMGL, 2500 YMGM, and 2554 YMGM telephones have four fixed feature buttons:

- MUTE button
- FLASH button
- REDIAL button
- HOLD button

MUTE button

Pressing and holding the MUTE button down blocks your conversation from being heard by the other party.

FLASH button

The FLASH button provides access to system features or use of custom calling services that require a press of the switchhook. Depressing the FLASH button will provide access to these features while depressing the switchhook will completely disconnect the call and provide dial tone for a completely new call.

REDIAL button

The REDIAL button is used to dial the last number dialed (up to 28 digits).

HOLD button

The HOLD button is used to put a call on station hold. Below the HOLD button, there is a red light. When the user puts a call on hold, the light goes on.

Message light

The red Message light flashes when a message has been left, and flutters when the telephone rings. The telephones recognize either LED or neon message waiting signals.

Other Physical Features

Positive Disconnect

2500 MMGN and 2554 MMGN telephones are manufactured without Positive Disconnect and without a FLASH button. 2500 YMGP and 2554 YMGP have Positive Disconnect permanently enabled. When the FLASH button is depressed, access is provided to switch features as provided. When the switchhook is depressed, the call is automatically disconnected and a dial tone is provided for a completely new call.

Older models have Positive Disconnect switch on the bottom of the telephone has two positions: ON and OFF.

- The ON position hangs up the telephone for approximately two seconds, even if the switchhook depression is less.This prevents inadvertent switchhook flashes.To initiate switchhook flash in this mode, press the FLASH button.
- In the OFF position, the switchhook will function normally.

Handset

All of these 2500 and 2554 Series telephones are equipped with a K-type handset.

Dial Pad

All of these 2500 Series telephones are equipped with a 12-button touch-tone dial pad. On all telephones, the letters "Q" and "Z" have been added to the appropriate dial pad keys, and the "5" button on the dial pad has raised bars for visually-impaired users.

Jacks

All of these 2500 Series telephones contain two jacks. The handset cord jack is on the left side of the telephone.(The handset also contains a jack for the other end of the handset cord.)The line cord jack is on the right rear of the set.

All of these 2554 Series telephones have one jack and one mounting cord. The handset cord jack is on the bottom of the telephone. The line cord is on the rear of the telephone to plug into the wall outlet.

Cords

Two cords are supplied with all four of these 2500 Series model telephones: a coiled 6-foot modular handset cord and a 7-foot modular line cord. Optional longer cords are available: a 12-foot handset cord, and 14-foot and 25-foot line cords. Two cords are supplied with 2554 Series model telephones: a coiled 6-foot modular handset cord and a permanently attached 4-inch modular mounting cord. An optional longer 12-foot handset cord is available.

Ringing

All of these 2500 Series telephones have an electronic tone ringer. There is a 3-position ringer volume control located on the bottom of the 2500 telephone and the side of the 2554 telephone.

Mounting Options

The 2500 MMGL, 2500 MMGM, 2500 YMGP, and 2500 YMGM telephones can only be desk-mounted. They cannot be wall-mounted. The 2554 MMGL, 2554 MMGM and 2554 YMGP telephones are wall-mounted telephones. They cannot be desk-mounted.

Color Options

All of these 2500 Series telephones are available in black or misty cream.

Power Requirements

All of the 2500/2554 Series telephones are powered by the tip and ring leads. The telephones do not require any external power supply.

Power Failure Operation

All of the 2500/2554 Series telephones can be used as an emergency station during power failure transfer conditions. The 2554 sets can ONLY be used as a Power Failure set in a Loop Start environment; the 2500 sets can be used as a Power Failure set in either a Loop Start or a Ground Start environment. Use in a Ground Start environment required the optional Modular Ground Start Button.

FCC Registration

These 2500 and 2554 telephones are FCC-registered.

3127 SoundPoint Speakerphone Products

The SoundPoint speakerphone adjunct connects to a telephone and provides high quality audioconferencing for desktop environments. The Analog version plugs into any standard analog phone jack. The DCP version plugs into the adjunct port of the 7400, 7500, 8400 and 8500 series terminals.

The full-duplex operation eliminates the clipping of speech and the de-reverberation technology helps to reduce the hollow, “bottom of the well” sound associated with common speakerphones.

SoundPoint automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. Additionally, this adaptation is an automatic and continuous process to accommodate for changing room conditions.

MODELS:

3127-ATR (40785355): AVAYA SOUNDPOINT ANALOG

The analog version of SoundPoint plugs into any analog port or terminal and comes will all appropriate cabling. In Addition, the analog version of SoundPoint can automatically answer (Auto-Answer) calls placed to its extension.

3127-DCP (407645779): AVAYA SOUNDPOINT DCP

Connects to the adjunct port of the 7400, 7500, 8400 and 8500 series terminals for added convenience. Comes with all appropriate cabling.

The SoundStation speakerphone has three microphones providing 360 degree of coverage for offices and small conference rooms. The amplified speaker provides enough audio for groups of up to 10 people. The analog version plugs into any standard analog phone jack. The DCP version plugs into the adjunct port of the 7102, 8102, 7400 and 8400 series terminals.

The full-duplex operation eliminates the clipping of speech and the Acoustic Clarity Technology helps to eliminate background noise. SoundStation automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. Additionally, this adaptation is an automatic and continuous process to accommodate for changing room conditions.

HIGHLIGHTS

SoundStation Speakerphone

- 360° of microphone coverage – ideal for desktop applications
- Digitally tuned speaker with Acoustic Clarity Technology™
- Full Duplex Operation – talk and listen at the same time
- Tabletop, user installable solution for ease of setup and use
- Extension microphones available for up to 20 people
- Integrated dial pad
- Microphone muting for privacy
- Dual color LED for on and mute display
- Wireless, lapel microphone available for stand-up presenters

MODELS

3127-STD (112238): AVAYA SOUNDSTATION ANALOG

The analog version of SoundStation plugs into any analog port or terminal and comes will all appropriate cabling. Requiring just an analog line and a power outlet, this system is easy to install and use. Good for groups of up to 10 people.

3127-EXP (112239): AVAYA SOUNDSTATION EX ANALOG

This analog version of SoundStation includes two extension microphones, which can be used to enhance the room coverage to support up to 15 people. Plugs directly into an analog port.

3127-DCS (407617000): AVAYA SOUNDSTATION DCP

The DCP version connects to the adjunct port of the 7102, 8102, 7400 and 8400 series terminals. **This SoundStation does not work with a 6400 series terminal.** Comes with all appropriate cabling and is user installable. Good for groups of up to 10 people.

3127-DCE (407617018): AVAYA SOUNDSTATION EX DCP w/Mics

This DCP version of SoundStation includes two extension microphones, which can be used to enhance the room coverage to support up to 15 people. The DCP version connects to the adjunct port of the 7102, 8102, 7400 and 8400 series terminals. **This SoundStation does not work with a 6400 series terminal.** Comes with all appropriate cabling and is user installable.

SoundStation Premier Audoconferencing Speakerphone

The SoundStation Premier speakerphone has three microphones providing 360° of coverage for offices and small conference rooms. The amplified speaker provides enough audio for groups of up to 25 people.

The analog version plugs into any standard analog phone jack, the DCP version plugs into the adjunct port of the 7102, 8102, 7400 and 8400 series terminals and the 6400-SSDP version plugs directly into a DCP jack.

The full-duplex operation eliminates the clipping of speech and the Acoustic Clarity Technology helps to eliminate background noise. In addition, SoundStation Premier focuses a microphone on the main speaker, effectively eliminating the hollow sound common with standard speakerphones. Also included is a full feature remote control and Display.

SoundStation Premier automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. This adaptation is an automatic and continuous process to accommodate for changing room conditions.

HIGHLIGHTS

SoundStation Premier Speakerphone

- 360° of microphone coverage – ideal for desktop applications
- Digitally tuned speaker accurately reproduces the human voice
- Full Duplex Operation – talk and listen at the same time
- De-reverberation – reduces the hollow, bottom of the well sound
- Tabletop, user installable solution for ease of setup and use
- Extension microphones available to support up to 25 people
- Integrated dial-pad and full-featured remote control
- Microphone muting for privacy
- Dual color LED for on and mute display
- Wireless, lapel microphone optional for stand-up presenters

MODELS

3127-APE (113987): AVAYA SOUNDSTATION PREMIER EX ANALOG

The analog version of SoundStation Premier plugs into any analog port or terminal and comes will all appropriate cabling. Expansion ports are available for adding on the optional extension microphones. Requiring just an analog line and a power outlet, this system is easy to install and use. Good for groups of up to 15 people.

3127-APX (113988): AVAYA SOUNDSTATION PREMIER EX/MICS ANALOG

This analog version of SoundStation Premier EX/Mics includes two extension microphones, which can be used to enhance the room coverage to support up to 25 people. Plugs directly into an analog port.

3127-DPE (407795285): AVAYA SOUNDSTATION PREMIER DCP EX

A DCP version of SoundStation Premier EX connects to the adjunct port of the 7102, 8102, 7400 and 8400 series terminals. **(Does not work with a 6400 Series Terminal.)** Expansion ports are available for adding on the optional extension microphones. Comes with all appropriate cabling and is user installable. Good for groups of up to 15 people.

3127-DPX (407795293): AVAYA SOUNDSTATION PREMIER DCP EX w/Mics

This DCP version of SoundStation Premier EX includes two extension microphones, which can be used to enhance the room coverage to support up to 25 people. The DCP version connects to the adjunct port of the 7102, 8102, 7400 and 8400 series terminals. **(Does not work with a 6400 Series Terminal.)** Comes with all appropriate cabling and is user installable. Good for groups of up to 25 people.

3127-DDP (700052863): AVAYA 6400-SSDP - SOUNDSTATION DCP PREMIER EX

A DCP version of SoundStation Premier EX that connects directly into a DCP Jack. It can work with or without a terminal. It can work with any 6400, 7400 or 8400 series terminals. Expansion ports are available for adding on the optional extension microphones. Comes with all appropriate cabling and is user installable. Good for groups of up to 15 people.

3127-DDX (700052871): AVAYA 6400-SSDP - SOUNDSTATION DCP PREMIER w/Mics

This 6400-SSDP includes two extension microphones, which can be used to enhance the room coverage to support up to 25 people. The 6400-SSDP connects directly into a DCP Jack. It can work or without a terminal. It can work with any 6400, 7400 or 8400 series terminals. Comes with all appropriate cabling and is user installable.

3127-MIC (407428432): AVAYA SOUNDSTATION WIRELESS LAPEL MIC

This clip-on microphone is designed for stand-up presenters. Consisting of a small transmit/receive pack that clips onto your belt and collar, this microphone has a 100’ range and comes in two individual frequencies.

3127-PMI (407795277): EXTENSION MICROPHONES FOR SOUNDSTATION PREMIER

Two extension microphones to enhance and expand room coverage. Compatible with 3127-APE, 3127-DPE and 3127-DDP

302D Attendant Console

The 302D series Attendant Console is a 2-wire unit with an optional expansion module 26C. The Console uses the power of the TN2181 or TN2224 DCP circuit card.

All Attendant Consoles require power either from the desktop or closet. The 302D comes in three colors, Dark Gray, Black and Avaya White. They will be CE2001 compliant for the EMEA region and the only console offered as of June 5, 2001. The display will be a 1x40 universal Display that will support Katakana, Roman and Euro font Character set. Label Languages are Japanese, English, French, Dutch, Spanish Latin America, Italian, German, Canadian French, Brazilian Portuguese and two of each are included with each Console.

The physical appearance of the 302D Attendant Console:

- Colors are dark gray, black and white.
- The terminals are branded on the faceplate with Avaya.
- All 302D Attendant Consoles are desk mounted only.
- Handset/Headset connection is moved to the front and is a single modular plug. For headset an H1C or M12L is Optional.
- Service observing must be done through the MultiVantage Software via the feature of Service Observing. An optional Training-Y connector can be used in conjunction with headset connection for desktop listen only supervisor support.

AVAYA 4602 IP Telephone

4602 IP Telephone provides advanced feature functionality with an intuitive and innovative user interface. The 4620 offers best in class features at a low price.

Highlights of the 4602 IP Telephone:

- Entry Level terminal with 2 x 24 character display
- 2 call appearances/feature buttons
- Global ready
- Fixed button for voice mail retrieval

Avaya 4606 IP Telephone

The 4606 IP Telephone is a fully Internet-capable set supporting Internet Protocol (IP) standards, providing extensive phone features and functionality, along with a high level of reliability and voice quality, in both the handset and speakerphone. The 4606 IP telephone comes equipped with an integrated two-line, liquid crystal display, six (6) programmable call appearance/feature buttons and a full duplex speakerphone. Feature and line appearance buttons have a red and green LED associated with each to indicate line states and feature status. The display informs the user of date, time, elapsed call time, calling party information and provides access to user-state dependent system features.

The 4606 IP telephone's integrated display is a 32-character liquid crystal display (2-lines with 16 characters each) for messaging and call management.

Highlights of the 4606 IP Telephone:

- 6 Programmable appearance/feature buttons with dual red/green LEDs
- 5 Fixed feature buttons - Speaker, Mute, Hold, Volume Up and Down
- 3 Fixed feature buttons under display - Conference, Transfer, and, Redial
- Full Duplex Speakerphone with Echo Cancellation
- 10/100Base-T Ethernet network connection with RJ-45 interface
- Integrated Ethernet Repeater hub for optional PC connection
- G.711, G.723.1, G.723.1A, G.729A, and G.729B audio voice coders
- H.323 V2 and NetMeeting compatible
- IP Address assignment - DHCP client or statically configured
- Infrared port to support future applications
- Downloadable Firmware for future upgrades
- Hearing aid compatible
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Integrated RJ-11 Headset Jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)
- Eight personalized ringing options
- K-Style handset with nine foot modular cord
- 14-Foot modular line cord
- Available in Black or White

Additional options include:

- 12-Foot modular handset cord
- 25-Foot modular line cord
- Base stand (15 degree angle)
- Avaya/Plantronics headsets
- Amplifier handset
- Noisy environment handset
- Push-to-talk handset

Avaya 4612 IP Telephone

The 4612 IP Telephone is a fully Internet-capable set supporting Internet Protocol (IP) standards, providing extensive phone features and functionality, along with a high level of reliability and voice quality, in both the handset and speakerphone. The 4612 IP telephone comes equipped with an integrated two-line, liquid crystal display, 12 programmable call appearance/feature buttons and a full duplex speakerphone. Feature and line appearance buttons have a red and green LED associated with each to indicate line states and feature status. The display informs the user of date, time, elapsed call time, calling party information and provides access to user-state dependent system features.

The 4612 IP telephone's integrated display is a 48-character liquid crystal display (2-lines with 24 characters each) for messaging and call management.

Highlights of the 4612 IP Telephone:

- 12 Programmable appearance/feature buttons with dual red/green LEDs
- 8 Fixed feature buttons - Speaker, Mute, Hold, Transfer, Conference, Redial, Volume Up and Down
- 4 Fixed navigation buttons - Menu, Exit, Previous (<-) and Next (->)
- Softkeys under display for browser user interface
- Full Duplex Speakerphone with Echo Cancellation
- 10/100 Base-T Ethernet network connection with RJ-45 interface
- Integrated Ethernet Repeater hub for optional PC connection
- G.711, G.723.1, G.723.1A, G.729A, and G.729B audio voice coders
- H.323 V2 and NetMeeting compatible
- IP Address assignment - DHCP client or statically configured
- Universal Serial Bus and Infrared port to support future applications
- Downloadable Firmware for future upgrades
- Hearing aid compatible
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Integrated RJ-11 Headset Jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)

- Station user may be allowed to program, remove, or rearrange the following features on set:

Account Code Entry	Directed Call Pickup
Automatic Dialing Buttons	Group Page
Blank (to remove feature)	Send All Calls
Call Forward	Whisper Page
Call Park	Whisper Page Answer
Call Pickup	Whisper Page Off

NOTE:
The System Administrator may substitute other soft key features for the above.

- Eight personalized ringing options
- K-Style handset with nine foot modular cord
- 14-Foot modular line cord
- Available in Black or White

Additional options include:

- 12-Foot modular handset cord
- 25-Foot modular line cord * Base stand (15 degree angle)
- Avaya/Plantronics headsets
- Amplifier handset
- Noisy environment handset
- Push-to-talk handset

AVAYA 4620 IP Telephone

4620 provides advanced feature functionality with an intuitive and innovative user interface. The 4620 can provide telephony, Speed Dial, Call Log, and Web browsing functionality. The 4620 supports the same Feature Key Expansion Unit (FKEU) as the 2420 telephone. The 4620 can use either the "resident" or "thin-client" approach, or both, to software architecture. With "resident" (also known as "thick-client") software installed, the 4620 holds the complete software for all the applications with the server handling all the call control. With "thin-client" software installed, a MultiVantage server still handles call control, while a separate application server in conjunction with the controls most of the display.

Features of the 4620 include:

- High end feature set
- Large Screen graphic display (168-by-132 pixel 4-grayscale)
- Advance user interface
- Local all log and speed dial productivity applications
- WML Browser Capability
- 24 call appearance/feature buttons on 2 pages
- Integrated switched ports
- Adjustable desk stand
- Function Key expansion unit jack
- Fully global ready (Icons)
- 4 softkeys

4624 IP Telephone

The 4624 IP Telephone is a fully Internet-capable set supporting Internet Protocol (IP) standards, providing extensive phone features and functionality, along with a high level of reliability and voice quality, in both the handset and speakerphone. The 4624 IP telephone comes equipped with an integrated two-line, liquid crystal display, 24 programmable call appearance/feature buttons and a full duplex speakerphone. Feature and line appearance buttons have a red and green LED associated with each to indicate line states and feature status. The display informs the user of date, time, elapsed call time, calling party information and provides access to user-state dependent system features.

The 4624 IP telephone's integrated display is a 48-character liquid crystal display (2-lines with 24 characters each) for messaging and call management.

The 4624 IP phone comes equipped with the following:

- 24 Programmable appearance/feature buttons with dual red/green LEDs
- 8 Fixed feature buttons – Speaker, Mute, Hold, Transfer, Conference, Redial, Volume Up and Down
- 4 Fixed navigation buttons – Menu, Exit, Previous (<-) and Next (->)
- Softkeys under display for browser user interface
- Full Duplex Speakerphone with Echo Cancellation
- 10/100 Base-T Ethernet network connection with RJ-45 interface
- Integrated Ethernet Repeater hub for optional PC connection
- G.711, G.723.1, G.723.1A, G.729A, and G.729B audio voice coders
- H.323 V2 and NetMeeting compatible
- IP Address assignment – DHCP client or statically configured
- Universal Serial Bus, Infrared and DSS ports to support future applications
- Downloadable Firmware for future upgrades
- Hearing aid compatible
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Integrated RJ-11 Headset Jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)

- Station user may be allowed to program, remove, or rearrange the following features on set:

Account Code Entry	Directed Call Pickup
Automatic Dialing Buttons	Group Page
Blank (to remove feature)	Send All Calls
Call Forward	Whisper Page
Call Park	Whisper Page Answer
Call Pickup	Whisper Page Off

NOTE:
The System Administrator may substitute other soft key features for the above

- Eight personalized ringing options
- K-Style handset with nine foot modular cord
- 14-Foot modular line cord
- Available in Black or White

Additional options include:

- 12-Foot modular handset cord
- 25-Foot modular line cord
- Base stand (15 degree angle)
- Avaya/Plantronics headsets
- Amplifier handset
- Noisy environment handset
- Push-to-talk handset

4630 IP Screenphone

The 4630 Screenphone is a fully Internet-capable IP appliance supporting Internet Protocol (IP) standards, while providing a user-friendly window into new IP enabled applications, as well as the full suite of MultiVantage Telephony, LDAP directory, and AUDIX voice mail features. Six telephony related applications will be provided through a unique user interface, developed with ease-of-use and minimal touch access in mind. All applications are enabled through a 1/4 VGA color, touch-screen display. The 4630 Screenphone features a built-in Ethernet hub, headset jack and full duplex speakerphone to provide a high level of voice quality, in both the handset and speakerphone, as well as a high level of reliability. The 4630 Screenphone's integrated 1/4 VGA color touch-screen display supports 320 dots horizontal by 320 dots vertical, with softkeys under the display for browser user interface. The 4630 Screenphone comes equipped with the following:

- 1/4 VGA color touch-screen display
- Softkeys under display for browser user interface
- 5 Fixed feature buttons – Speaker, Mute, Hold, Headset, Volume Up and Down
- Full Duplex Speakerphone with Echo Cancellation
- ~100 Speed dial buttons organized into groups for easier access
- Call log of incoming and outgoing calls
- Two 10/100 Base-T Ethernet RJ-45 interfaces—one for network to telephone connection and one "hub port" for connection from telephone to PC
- Six Telephony related applications
- Multi-button business telephone capabilities with DEFINITY ECS
- Directory access to corporate telephone directory information on a LDAP (Lightweight Directory Access Protocol) server
- Voice mail access to voice mail messaging capabilities of a DEFINITY AUDIX or an Intuity AUDIX
- Web "browsing" access to web-based information, including support for downloaded Java applets
- G.711, G.723.1, G.723.1A, G.729A, and G.729B audio voice coders
- H.323 V2 and NetMeeting compatible
- IP Address assignment – DHCP client or statically configured
- Infrared and DSS ports to support future applications
- Downloadable Firmware for future upgrades
- Hearing aid compatible
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Integrated RJ-11 Headset Jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)

- K-Style handset with nine foot modular cord
- 14-Foot modular line cord
- Color: multiple Gray tones

Additional options include:

- 12-Foot modular handset cord
- 25-Foot modular line cord
- Base stand
- Avaya/Plantronics headsets
- Amplifier handset
- Noisy environment handset
- Push-to-talk handset

500/554 Series Analog Terminals

The Avaya 500/554 analog voice terminals are single line rotary sets that are ideal for users requiring basic voice functions. They offer excellent economy, yet provide access to all of the features available for single line voice terminals using MultiVantage Software. The 500 sets are available only as desk models and the 554 sets are available only as wall models.

The 500/554 Series voice terminals come equipped with the following:

- Either a K-Style handset or a G-style handset (depending upon vintage) with six foot modular cord
- Seven foot modular line cord
- Either a 3-Position electronic tone ringer volume control switch or an adjustable bell ringer, depending upon vintage.
- Line powered
- FCC-Registered for emergency power failure transfers
- Available in Beige, Ivory, Misty Cream or Black, depending upon vintage.

In addition to the standard features, the 500/554 terminal set may include:

- Flash button
- Message waiting light supports either LED or neon signals, but not both.

Additional options include:

- 12-Foot handset cord
- 14-Foot and 25-foot modular line cords

Avaya /Plantronics headsets Power Requirements

The 500/554 telephone is powered by the tip and ring leads. It does not require any external power supply. Power Failure Operation The 500/554 telephone can be used as an emergency station during power failure transfer conditions in either a Loop Start or a Ground Start environment. Use in a Ground Start environment requires the use of an optional Modular Ground Start Button. 554 telephones may be used only in a Loop Start environment. FCC Registration The 500/554 telephone is FCC registered.

6210 Analog Voice Terminal

The 6210 Analog Voice Terminal is the ideal solution for single line station users requiring enhanced voice functions. It offers excellent economy, yet provides access to all the features available for single line voice terminals on the Avaya MultiVantage Solutions.

The 6210 Analog Voice Terminal features include the following:

- Contemporary style handset with nine foot modular cord
- Seven foot modular line cord
- Handset volume control
- Ringer volume control
- Message waiting light
- Flash button
- Set hold button with LED Indicator
- Last number re-dial button
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Positive disconnect via switchhook
- Desk/wall mount housing
- RJ-11 Data jack
- FCC approved for emergency power failure transfers
- Line powered
- Available in Gray or White Additional options include:
- 12-Foot handset cord
- 14-Foot and 25-foot modular line cords
- Avaya/Plantronics headsets

6211 Analog Voice Terminal

The 6211 Analog Voice Terminal is the ideal solution for single line station users requiring enhanced voice functions. It offers excellent economy, yet provides access to all the features available for single line voice terminals on the Avaya MultiVantage Solutions.

The 6211 Analog Voice Terminal features include the following:

- Contemporary style handset with nine foot modular cord
- Seven foot modular line cord
- Handset volume control
- Ringer volume control
- Message waiting light
- Flash button
- Set hold button with LED Indicator
- Last number re-dial button
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Positive disconnect via switchhook
- Desk/wall mount housing
- RJ-11 Data jack
- FCC approved for emergency power failure transfers
- Line powered
- Available in Gray or White

Additional options include:

- 12-Foot handset cord
- 14-Foot and 25-foot modular line cords
- Avaya/Plantronics headsets Status: Currently Sold and Supported

6219 Analog Voice Terminal

The 6219 Analog Voice Terminal is the ideal solution for single line station users requiring enhanced voice functions. It offers excellent economy, yet provides access to all the features available for single line voice terminals on the Avaya MultiVantage Solutions.

The 6219 Analog Voice Terminal features include the following:

- Contemporary style handset with nine foot modular cord
- Seven foot modular line cord
- Handset volume control
- Ringer volume control
- Message waiting light
- Flash button
- Set hold button with LED Indicator
- Last number re-dial button
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Positive disconnect via switchhook
- Desk/wall mount housing
- RJ-11 Data jack
- FCC approved for emergency power failure transfers
- Line powered
- Available in Gray or White
- System Administrator Programmable System Hold
- Repertory Dialing (10 buttons)
- Personalized Ringing

Program Keylock Additional options include:

- 2-Foot handset cord
- 14-Foot and 25-foot modular line cords
- Avaya/Plantronics headsets

6402 Digital Telephone

The features include:

- Single line with switchhook flash access to second call. (No Call Appearance button)
- Built-in group listening speaker only. (Can NOT be optioned as 2-way speakerphone)
- 6 Fixed buttons - Speaker, Feature, Hold, Redial, Transfer, Conference
- The "Feature" button allows access via the dialpad to 12 MultiVantage features that do not require indicators or display messages.
- Adjustable volume control (Handset, Speaker, & Ringer) * Message Waiting Light (LED)
- 2-wire connectivity through 2-wire Digital line circuit packs only, Digital voice technology
- Accepts download from MultiVantage software of Voice & Touchtone transmission parameters as required by individual countries
- Internal self test to determine if LEDs light
- Choice of 8 ringing patterns * Voice terminal is powered from the switch
- Can be used with or without stand
- Desk or wall mountable
- Matching 9 ft. (2.7 m) handset cord & 7 ft. (2.1 m) modular line cord
- Available in Dark Gray and White
- NO Adjunct jack interface for external speakerphones or headset modules
- Headsets must be connected through the handset.

6402D Display Digital Telephone

The features include:

- 2 line x 16 character LCD display showing time and date when terminal in idle status.
- The display is not tiltable.
- Display shows only messages sent from the switch.
- There are no softkey buttons or softkey features available on the 6402D display.
- Single line with switchhook flash access to second call. (No Call Appearance button)
- Built-in group listening speaker only. (Can NOT be optioned as 2-way speakerphone)
- 6 Fixed buttons - Speaker, Feature, Hold, Redial, Transfer, Conference
- The "Feature" button allows access via the dialpad to 12 MultiVantage features that do not require indicators, softkeys, or lengthy display messages.
- Adjustable volume control (Handset, Speaker, & Ringer)
- Message Waiting Light (LED)
- 2-wire connectivity through 2-wire Digital line circuit packs only, Digital voice technology
- Accepts download from MultiVantage of Voice & Touchtone transmission parameters as required by each country
- Internal self test to determine if LEDs light
- Choice of 8 ringing patterns
- Voice terminal is powered from the switch
- Can be used with or without stand
- Desk or wall mountable
- Matching 9 ft. (2.7 m) handset cord & 7 ft. (2.1 m) modular line cord
- Available in Dark Gray and White
- NO Adjunct jack interface for external speakerphones or headset modules Headsets must be connected through the handset jack.

6408D+ Digital Display Telephone

The features include:

- 2 line x 24 character LCD display showing time and date when terminal in idle status
- The display is tiltable with 3 viewing angles
- 8 Call Appearance/Feature buttons with dual LEDs
- Built-in 2-way speakerphone which can also be optioned as 1-way Group Listening speaker
- 6 Fixed buttons - Speaker, Mute, Hold, Redial, Transfer, Conference

NOTE:
Drop must be administered on a softkey or a Call Appearance/Feature button

- 12 system features can be administered on softkeys associated with the display
- 4 buttons to access softkey features (Menu, Exit, Previous, & Next)
- One Next button - for use with softkeys and Directory function
- When the "Headset" feature is administered, it is no longer necessary to take handset off hook to answer phone when using a headset
- When system administrator provides the user with permission, user may add, remove or rearrange the following features on their Call Appearance/Feature buttons
- Adjustable volume control (Handset, Speaker/Speakerphone, & Ringer)
- Message Waiting Light (LED)
- 2-wire connectivity through 2-wire Digital line circuit packs only, Digital voice technology
- Accepts download from MultiVantage of Voice & Touchtone transmission parameters as required by each country
- Internal self test to determine if LEDs light
- Choice of 8 ringing patterns
- Voice terminal is powered from the switch
- Can be used with or without stand
- Desk or wall mountable
- Matching 9 ft. (2.7 m) handset cord & 7 ft. (2.1 m) modular line cord
- Available in Dark Gray and White
- NO Adjunct jack interface for external speakerphones or headset modules Headsets must be connected through the handset jack

6416D+M Digital Display Telephone

The 6424D+M is a two-wire digital display voice terminal equipped with 24 call appearance/feature buttons and a slot on the back for integrating application modules for more functionality. With the addition of the optional XM24 Expansion Module the total number of call appearance/feature buttons increases to 48. The 6424D+M is ideal for executives, secretaries, or call center supervisors requiring multiple lines for call handling, call coverage, conferencing, and sophisticated voice communications.

The 6424D+M's integrated tiltable display is a two-line by 24 character liquid crystal display for messaging and call management.

The 6424D+M digital voice terminal comes equipped with the following:

- 24 Call appearance/feature buttons with dual LEDs
- 10 fixed features - Speaker, Mute, Conference, Transfer, Hold, Redial, Menu, Exit, Previous, and Next
- 12 Assignable soft key features associated with the display
- Built-in two-way programmable speakerphone can be optioned by user on an individual call basis for group listening
- Headset jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)
- Station user may be allowed to program, remove, or rearrange the following features:

Account Code Entry	Directed Call Pickup
Automatic Dialing Buttons	Group Page
Blank (to remove feature)	Send All Calls
Call Forward	Whisper Page
Call Park	Whisper Page Answer
Call Pickup	Whisper Page Off

- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Eight personalized ringing options
- K-Style handset with nine foot modular cord

- 14-Foot modular line cord
- Pull-out feature reference card tray
- Stand for desk or wall mount configuration
- International portability
- Downloadable Transmission Parameters
- Available in Gray or White
- Meets Class B requirements for use in residential location
- Meets FCC Hearing Aid Compatibility Act requirements that go into effect January 1, 2000

Additional options include:

- 100A Analog Interface Module provides an analog interface for connection of full-duplex analog speakerphones, computer modems (up to V.90), fax machines, analog telephones, TDD, and other devices that require an analog port.
- XM24 Expansion Module with 24 buttons increases the total set button capacity to 48 buttons. All 24 buttons have dual LED lights and can be administered for either call appearances or features
- Adjunct station or closet power is necessary when connecting XM24 Expansion Module or 100A Analog Interface Module
- 12-Foot modular handset cord
- 25-Foot modular line cord
- HIC-1 Headset interface cord
- Headset modular base unit M10L or M12LU
- Avaya/Plantronics headsets
- 8400B+ data module
- Amplifier handset
- Noisy environment handset

6416D+M Digital Display Telephone

The 6424D+M is a two-wire digital display voice terminal equipped with 24 call appearance/feature buttons and a slot on the back for integrating application modules for more functionality. With the addition of the optional XM24 Expansion Module the total number of call appearance/feature buttons increases to 48. The 6424D+M is ideal for executives, secretaries, or call center supervisors requiring multiple lines for call handling, call coverage, conferencing, and sophisticated voice communications.

The 6424D+M's integrated tiltable display is a two-line by 24 character liquid crystal display for messaging and call management.

The 6424D+M digital voice terminal comes equipped with the following:

- 24 Call appearance/feature buttons with dual LEDs
- 10 fixed features - Speaker, Mute, Conference, Transfer, Hold, Redial, Menu, Exit, Previous, and Next
- 12 Assignable soft key features associated with the display
- Built-in two-way programmable speakerphone can be optioned by user on an individual call basis for group listening
- Headset jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)
- Station user may be allowed to program, remove, or rearrange the following features:

Account Code Entry	Directed Call Pickup
Automatic Dialing Buttons	Group Page
Blank (to remove feature)	Send All Calls
Call Forward	Whisper Page
Call Park	Whisper Page Answer
Call Pickup	Whisper Page Off

- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Eight personalized ringing options
- K-Style handset with nine foot modular cord

- 14-Foot modular line cord
- Pull-out feature reference card tray
- Stand for desk or wall mount configuration
- International portability
- Downloadable Transmission Parameters
- Available in Gray or White
- Meets Class B requirements for use in residential location
- Meets FCC Hearing Aid Compatibility Act requirements that go into effect January 1, 2000

Additional options include:

- 100A Analog Interface Module provides an analog interface for connection of full-duplex analog speakerphones, computer modems (up to V.90), fax machines, analog telephones, TDD, and other devices that require an analog port
- XM24 Expansion Module with 24 buttons increases the total set button capacity to 48 buttons. All 24 buttons have dual LED lights and can be administered for either call appearances or features
- Adjunct station or closet power is necessary when connecting XM24 Expansion Module or 100A Analog Interface Module
- 12-Foot modular handset cord
- 25-Foot modular line cord
- HIC-1 Headset interface cord
- Headset modular base unit M10L or M12LU
- Avaya/Plantronics headsets
- 8400B+ data module
- Amplifier handset
- Noisy environment handset

6424D+ Digital Telephone

The 6424D+ is a two-wire digital display voice terminal equipped with 24 call appearance/feature buttons. With the addition of the optional XM24 Expansion Module the total number of call appearance/feature buttons increases to 48. The 6424D+ is ideal for executives, secretaries, or call center supervisors requiring multiple lines for call handling, call coverage and conferencing.

The 6424D+'s integrated tiltable display is a two-line by 24character liquid crystal display for messaging and call management.

The 6424D+ digital voice terminal comes equipped with the following:

- 24 Call appearance/feature buttons with dual LEDs
- 10 fixed features - Speaker, Mute, Conference, Transfer, Hold, Redial, Menu, Exit, Previous, and Next
- 12 Assignable soft key features associated with the display
- Built-in two-way programmable speakerphone can be optioned by user on an individual call basis for group listening
- Headset jack for direct connection of headset
- Adjustable volume control (handset, speaker, and ringer)
- Station user may be allowed to program, remove, or rearrange the following features:

Account Code Entry	Directed Call Pickup
Automatic Dialing Buttons	Group Page
Blank (to remove feature)	Send All Calls
Call Forward	Whisper Page

- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired * Message waiting light (LED)
- Eight personalized ringing options
- K-Style handset with nine foot modular cord
- 4-Foot modular line cord
- Pull-out feature reference card tray
- Stand for desk or wall mount configuration
- International portability

- Downloadable Transmission Parameters
- Available in Gray or White
- Meets Class B requirements for use in residential location
- Meets FCC Hearing Aid Compatibility Act requirements that go into effect January 1, 2000

Additional options include:

- XM24 Expansion Module with 24 buttons increases the total set button capacity to 48 buttons. All 24 buttons have dual LED lights and can be administered for either call appearances or features
- Adjunct station or closet power is necessary when connecting XM24 Expansion Module or 100A Analog Interface Module
- 12-Foot modular handset cord
- 25-Foot modular line cord
- HIC-1 Headset interface cord
- Headset modular base unit M10L or M12LU
- Avaya/Plantronics headsets
- 8400B+ data module
- Amplifier handset
- Noisy environment handset

6426D+M Display Terminal

The features include:

- 2 line x 24 character LCD display showing time and date when terminal in idle status.
- The display is tiltable with 3 viewing angles.
- 24 Call Appearance/Feature buttons with dual LEDs
- Built-in 2-way speakerphone which can also be optioned as 1-way Group Listening speaker
- 6 Fixed buttons - Speaker, Mute, Hold, Redial, Transfer, Conference

NOTE:
Drop must be administered on a softkey or a Call Appearance/Feature button

- 12 system features can be administered on softkeys associated with the display
- 4 buttons to access softkey features (Menu, Exit, Previous, & Next)
- Only one "Next' button - for use with softkeys and Directory function.
- Ribbon Connector under terminal to connect optional modules that fit into the stand.
  - — First module is 100A Analog Interface Module (T/R module) (See descriptor sheet on 100A Analog Interface Module)
  - — Not compatible with S201/S203 speakerphone adjunct or headset adjunct
- Headset jack under terminal, next to handset jack, for direct connection of headset.
  - — Must administer a line/feature button with the "Headset on/off" feature for the headset to work when plugged into jack.
  - — Only Lucent/AT&T/Plantronics headsets will be supported.
  - — May connect a Lucent/AT&T/Plantronics headset using either the new Headset Interface Cord (HIC-1) or either a M10L or M12LU Modular base unit.
- When the "Headset" feature is administered, it is not necessary to take handset off hook to answer phone when using a headset.
- When using headset via "Headset" feature button, handset becomes listen-only for monitoring until headset button turned off.
- No Adjunct jack interface for external S201/S203 speakerphone adjuncts or headset adjuncts
- When system administrator provides the user with permission, user may add, remove or rearrange the following features on their Call Appearance/Feature buttons. Refer to the following table.

Acct Account Code Entry	**DPkUp** Directed Call Pickup (Pick up call not in your pickup group)
AutoD Auto Dial (up to 16 characters)	**GrpPg** Group Page
Blank Blank to remove feature before adding new feature	**SAC** Send All Calls
CFrwd Call Forwarding All Calls	**WspPg** Whisper Page
Cpark Call Park	**WspAn** Whisper Page Answer
CPkUp Call Pickup	**WsOff** Whisper Page Off

- Adjustable volume control (Handset, Speaker/Speakerphone, & Ringer)
- Message Waiting Light (LED)
- Supports optional XM24 expansion module (24 additional Call Appearance/Feature buttons with dual LEDs)
- The 6424D+M terminal is powered from the switch.
- Adjunct station or closet power is necessary only when connecting an XM24 expansion module or the 100A Analog Interface Module. If both modules are connected to the 6424D+M, only one power supply is necessary. The 6424D+M will continue to work if the aux power is interrupted, but the modules will not work
- 2-wire connectivity through 2-wire Digital line circuit packs only, Digital voice technology
- Accepts download from MultiVantage of Voice & Touchtone transmission parameters as required by each country
- Internal self test
- Choice of 8 ringing patterns
- Can be used with or without stand when the 100A Analog Interface module not in stand.
- Desk or wall mountable. If 100A Analog Interface Module is connected, terminal can not be wall mounted.
- Matching 9 ft. (2.7 m) handset cord & clear or light gray 14 ft. (4.2 m) modular 2-wire line cord.
- Available in Dark Gray and Lucent White
- Meets Class B requirements for use in residential location.
- Is US & Canadian UL listed and meets Canadian D.O.C. requirements.
- Meets the US FCC Hearing Aid Compatibility Act requirements that go into effect January 1, 2000.

Specifications

6424D+M	US	METRIC
Height – without stand – display down or up	3.72 in.	9.45 cm
- with stand - display down	5.06 in.	12.85 cm
- display up	5.75 in.	14.61 cm
Length	10.45 in.	26.54 cm
Width	9.00 in	22.86 cm
Terminal Weight	2.82 lb.	1.27 kg
Boxed Weight	4.10 lb.	1.85 kg

PEC information (Note: PECs are orderable only in the US & Canada)

Description	PEC & SAP BOM	SAP / Comcode	Color	Prod ID
6424D+M with Display, 2-way Speakerphone, built-in headset jack, module ready, Class B	3307-MUG 151516	108331240	Gray	BBQ
	3307-MUW 151515	108331257	White	BBQ

NOTE:
Choose a variety of Lucent/AT&T/Plantronics headset headpieces. All headpieces must connect to the terminal via one of the following methods.

Description	PEC	Comcode
HIC-1 Headset Interface Cord (long curly cord from quick disconnect plug to directly plugging into jack on set)	3124-HIC	408122950
Modular Base Unit for Headset M10L or Modular Base Unit for Headset M12LU (Box unit with controls and long curly cord to quick disconnect plug. This curly cord can NOT plug directly into set. It will NOT work.)	3122-022	407639715 OR 408094704

7101A Voice Terminal

The 7101A voice terminal is a single-line analog model that requires one tip and ring pair for operation. It is equipped with a Message Waiting light and a RECALL button for activating the system's special features. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

NOTE:
The 7101A has been discontinued.

Physical Description

Dimensions

The following dimensions are approximate.

Width = 3-3/4 inches
Depth (front to back) = 8-3/4 inches
Height (maximum with handset in place) = 5-1/2 inches
Thickness of housing = 1-1/2 inches

Features

- Two Fixed Feature Buttons
- RECALL button
- DISCONNECT button

Message Light

The green Message light goes on when a message is left for the user. It goes off when the user retrieves the message.

Other Physical Features

Handset

The 7101A Voice Terminal is equipped with an R-type handset. Dial pad The 7101A Voice Terminal has a 12-button touch-tone dial pad.

Jacks

The 7101A housing contains two jacks. The handset cord jack is just under the left front edge of the housing. (The handset contains a jack for the other end of the handset cord.) The line jack is on the bottom center of the housing.

Cords

Two cords are supplied with the 7101A terminal: a coiled 7-foot modular handset cord and a 7-foot modular line cord. Optional longer cords are available: a 12-foot handset cord, and 14-foot and 25-foot line cords.

Ringing

The 7101A Voice Terminal has electronic tone ringing. The volume control is on the left side of the housing.

Mounting Options

The 7101A Voice Terminal comes equipped with a non-adjustable desk stand. An optional wall mounting kit may be ordered.

Color Options

The 7101A Voice Terminal is available in black only.

7102A/7102 Plus Voice Terminals

The 7102A and the 7102 Plus voice terminals are single-line analog models that require one tip and ring pair for operation. It allows access to system features via the RECALL Button and dial codes.

The 7102A and 7102 Plus voice terminal faceplate contains a Feature Access Code Card that can be customized to reflect the most frequently accessed features.

NOTE:
Both of the 7102 voice terminals have been discontinued.

Physical Description

Dimensions

The following dimensions for the 7102A and the 7102 Plus are approximate.

Width = 7 inches Depth (front to back) = 8-3/4 inches
Height (maximum with handset in place) = 5-1/2 inches
Thickness of housing = 1-1/2 inches

Features

There is one Fixed Feature Button - it is the RECALL Button

Message Light

The red Message light flashes when a message is left for the user. It goes off when the user retrieves the message.

Handset

The 7102 voice terminals are equipped with an R-type handset.

Dial Pad

The 7102 voice terminals have a 12-button touch-tone dial pad.

Jacks

The 7102A01A housing contains two jacks. The handset cord jack is just under the center front edge of the housing. (The handset contains a jack for the other end of the handset cord.) The line jack is on the bottom center of the housing. The 7102 Plus housing contains three jacks. The handset cord jack is just under the center front edge of the housing. (The handset contains a jack for the other end of the handset cord.) The line jack is on the bottom center of the housing. The modular speakerphone/headset jack is located on bottom of the set. Cords Two cords are supplied with the 7102 voice terminals: a coiled 7-foot modular handset cord and a 7-foot modular line cord. Optional longer cords are available: a 12-foot handset cord, and 14-foot and 25-foot line cords.

Ringing

The 7102 voice terminals have electronic tone ringing. The volume control is on the left side of the housing.

Mounting Options

The 7102 voice terminals come equipped with both a non-adjustable desk stand and a wall mounting bracket.

Color Options

The 7102 voice terminals are available in two colors: black and misty cream.

7103A Voice Terminal

The 7103A Fixed Feature and Programmable Feature voice terminal are single-line analog models with feature buttons that can be programmed by the system administrator (Fixed Feature) or programmed by the user (Programmable Feature). They requires one tip and ring pair for operation. They cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

NOTE:
The 7103A Fixed Feature and Programmable Feature voice terminals has been discontinued.

Physical Description Dimensions

The following dimensions are approximate.

Width = 7 inches Depth (front to back) = 8-3/4 inches
Height (maximum with handset in place) = 5-1/2 inches
Thickness of housing = 1-1/2 inches

Features

There are two Fixed Feature Buttons:

- RECALL button
- DISCONNECT button

Eight Feature Buttons

The system administrator designates and programs all buttons, except fixed feature buttons, for the common needs of all voice terminal users. Within any one system, all 7103A Fixed Feature voice terminals have identical features at the same button positions.

Message Light

The green Message light goes on when a message is left for the user. It goes off when the user retrieves the message.

Other Physical Features

Handset

The 7103A Fixed Feature voice terminal is equipped with an R-type handset.

Dial pad

The 7103A Fixed Feature voice terminal has a 12-button touch-tone dial pad.

Jacks

The 7103A Fixed Feature voice terminal housing contains three jacks. The handset cord jack is just under the left side of the housing. (The handset contains a jack for the other side of the handset cord.) The LINE and OTHER jacks are on the bottom center of the housing.

Cords

Two cords are supplied with the 7103A Fixed Feature voice terminal: a coiled 7-foot modular handset cord and a 7-foot modular line cord. Optional longer cords are available: a 12-foot handset cord, and 14-foot and 25-foot line cords.

Ringing

The 7103A Fixed Feature voice terminal has electronic tone ringing. The volume control is on the left side of the housing.

Mounting Options

The 7103A Fixed Feature voice terminal comes equipped with a nonadjustable desk stand. An adjustable desk stand or a wall mounting bracket can be optionally ordered.

Color Options

The 7103A Fixed Feature voice terminal is available in black only.

7104A Voice Terminal

The 7104A Voice Terminal is a single-line analog model. This voice terminal is equipped with a display that is used to display stored numbers. It requires one tip and ring pair for operation. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

NOTE:
The 7104A Voice Terminal is no longer manufactured. However, some remanufactured terminals may still be ordered.

Physical Description

Dimensions

The following dimensions are approximate.

Width = 8 inches
Depth (front to back) = 8-3/4 inches
Height (maximum with handset in place) = 5-1/2 inches
Thickness of housing = 1-1/2 inches

Features

Two Fixed Feature Buttons:

- RECALL button
- DISCONNECT button

Eight Special Fixed Feature Buttons

- SHIFT button - used for toggling between the two modes of the following four dual function feature buttons and the eight dual function one-touch dialing buttons
- SET CLOCK/TIMER button - used for setting time and for timing a call
- AM/DLYD RING button - used in time-setting procedure and to delay ringing on incoming calls
- PM/ABRV RING button - used in time-setting procedure and to provide abbreviated ringing on incoming calls
- SAVE/SEND button - used to save a dialed number, then call it again
- PROGRAM button - used in recording numbers for one-touch dialing
- DISPLAY button - used to display stored numbers
- SPACE, PAUSE, and STOP buttons - used in recording numbers for one-touch dialing

Eight dual function buttons

These buttons are divided into gray and blue halves and can be programmed for one-touch dialing of sixteen frequently called numbers and/or system features.

Loudspeaker

The 7104A Voice Terminal has a built in loudspeaker for group listening and on-hook dialing. The loudspeaker can be turned on and off with the SPEAKER button.

Display

A 14-Character Liquid Crystal Display (LCD) provides the following displays. Each display is local to the individual terminal.

- Abbreviated Ring mode in effect Number programmed
- Date Program mode in effect
- Delayed Ring mode in effect Shift mode in effect
- Display mode in effect Speaker on
- Low Battery Time
- Number called

Pull out tray

The 7104A Voice Terminal is equipped with a pull out tray with an instruction card for voice terminal operation.

Other Physical Features

Handset

The 7104A Voice Terminal is equipped with an R-type handset.

Dial pad

The 7104A Voice Terminal has a 12-button touch-tone dial pad.

Jacks

The 7104A voice terminal housing contains two jacks. The handset cord jack is just under the left side of the housing. (The handset contains a jack for the other end of the handset cord.) The line jack is on the bottom center of the housing. Cords Two cords are supplied with the 7104A Voice Terminal: a coiled 7-foot modular handset cord and a 7-foot modular line cord. Optional longer cords are available: a 12-foot handset cord, and14-foot and 25-foot line cords.

Ringing

The 7104A Voice Terminal has electronic tone personalized ringing. The personalized ringing option is controlled using a ring switch located under the rear end cap of the 7104A Voice Terminal. The volume control is on the left side of the housing.

Mounting Options

The 7104A Voice Terminal comes equipped with a nonadjustable desk stand.

Color Options

The 7104A Voice Terminal is available in three colors: black, misty cream, and chocolate brown.

8102 Analog Telephone

The basic 8102 telephone and the 8102M telephone are single-line analog sets that require one tip and ring pair for operation. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons, automatic redial (with the REDIAL button), selected personalized ringing pattern, a Message light, a HOLD button, a Data jack, and an Adjunct jack. The 8102 and 8102M telephones also allow access to system features with the FLASH button and appropriate dial codes.

Physical Description

Dimensions

NOTE:
The dimensions listed here for the 8102 and 8102M telephones are approximate.

- Width = 6-3/4 inches
- Depth (front to back) = 9-1/2 inches
- Height (maximum with handset in place) = 3-1/2 inches
- Thickness of housing = At the top of the telephone, the thickness measures
- 1-5/8 inches; at the bottom of the telephone, the thickness measures 1 inch.

Feature Buttons

Fixed Feature Buttons

The 8102 and 8102M telephones have the following fixed feature buttons:

- **FLASH** button: For using features such as Conference with business communications systems.
- **REDIAL** button: Can be used for redialing the last number dialed from the dial pad, either an extension or an outside number. The number redialed can be up to 20 digits. With the (Redial) Auto-Pause feature, the telephone can be programmed to insert pauses after a dial access code (such as 9) in a redialed number. If the user usually waits a short period to obtain a second dial tone after dialing an access code, this waiting time can be set by programming the automatic pause between the system access code and the telephone number. The (Redial) Auto-Pause feature will help eliminate misdialing.
- **HOLD** button: Used for putting a call on hold. If users want to put a call on hold and place another call, they must use the system Hold feature by using the FLASH button.
- **Message Light**: The red Message light flashes when a message is left for the user. It goes off when the user retrieves the message. The Message light also serves as a visual ringing indicator by flashing quickly while the telephone is ringing.

- **Programmable Dialing Buttons**: The 8102 and 8102M telephones provide 12 programmable dialing buttons on which the user can store frequently dialed telephone numbers or feature codes. To use these preprogrammed features, the user merely lifts the handset and then presses the appropriate programmable dialing button.
- **PROGRAM** and **PAUSE** buttons: Both of these buttons are used for storing numbers on the preprogrammed dialing buttons on the upper half of the faceplate of the telephone. The PAUSE button can be used to insert a one half second pause/delay into a telephone number stored on a programmable dialing button or into a dialing access code in the Auto-Pause feature of Redial.

Other Physical Features

Ringing

The 8102 and 8102M telephones have electronic tone ringing. The ringer volume control is a 3-position switch on the right side of the housing. The user can also use a Tone Selector (labeled “LO HI”) for selecting one of two-tone ringer frequencies or a second Tone Selector (labeled “SLOW FAST”) for selecting one of two-tone ringer modulation rates. Both Tone Selector switches are on the right side of the telephone.

Mounting Options

The 8102 and 8102M telephones come equipped with a non-adjustable desk stand which can be turned upside-down and used for wall mounting the telephone.

Jacks

The 8102 and 8102M telephones have a line jack on the back of the telephone for connecting a line cord, and a Handset cord jack for connecting the handset. These telephones also have a Data jack for connecting a device such as a modem to the telephone. The 8102 and 8102M telephones have an Adjunct (equipment) jack which allows the connection of a speakerphone to the telephone and an adjunct power jack for connecting an auxiliary power source for using the adjunct equipment.

Handset

The 8102 and 8102M telephones are equipped with a K-type handset.

Dial pad

The 8102 and 8102M telephones have a 12-button touch-tone dial pad. The letters “Q” and “Z” have been added to the appropriate dial pad keys for directory access, and the “5” button on your dial pad has raised bars for visually-impaired users.

Cords

Two cords are supplied with the 8102 and 8102M telephones: a coiled 9-foot modular handset cord and a 7-foot modular line cord.

Color Options

The 8102 and 8102M telephones are available in black and in white.

Power Requirements

The 8102 and 8102M telephones must have auxiliary power for adjunct equipment when such equipment is connected..

Power Failure Operation

The 8102 or 8102M telephone can be used as an emergency station during power failure transfer conditions.

Ringer Equivalency Numbers

The Ringer Equivalency Number for the 8102 telephone is: 0.5A and 1.6B. For the 8102M the Ringer Equivalency Number is: 1.5B and 0.8A.

Hearing Aid Compatible

These telephones are compatible with the inductively coupled hearing aids as required by the FCC.

Specialty Handsets

NOTE:
The 8102M is compatible with Avaya K-style Specialty Handsets while the 8102 is not.

8102M Analog Telephone

The basic 8102 telephone and the 8102M telephone are single-line analog sets that require one tip and ring pair for operation. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons, automatic redial (with the REDIAL button), selected personalized ringing pattern, a Message light, a HOLD button, a Data jack, and an Adjunct jack. The 8102 and 8102M telephones also allow access to system features with the FLASH button and appropriate dial codes.

Physical Description

Dimensions

NOTE:
The dimensions listed here for the 8102 and 8102M telephones are approximate.

- Width = 6-3/4 inches
- Depth (front to back) = 9-1/2 inches
- Height (maximum with handset in place) = 3-1/2 inches
- Thickness of housing = At the top of the telephone, the thickness measures
- 1-5/8 inches; at the bottom of the telephone, the thickness measures 1 inch.

Feature Buttons

Fixed Feature Buttons

The 8102 and 8102M telephones have the following fixed feature buttons:

- **FLASH** button: For using features such as Conference with business communications systems.
- **REDIAL** button: Can be used for redialing the last number dialed from the dial pad, either an extension or an outside number. The number redialed can be up to 20 digits. With the (Redial) Auto-Pause feature, the telephone can be programmed to insert pauses after a dial access code (such as 9) in a redialed number. If the user usually waits a short period to obtain a second dial tone after dialing an access code, this waiting time can be set by programming the automatic pause between the system access code and the telephone number. The (Redial) Auto-Pause feature will help eliminate misdialing.
- **HOLD** button: Used for putting a call on hold. If users want to put a call on hold and place another call, they must use the system Hold feature by using the FLASH button.
- **Message Light**: The red Message light flashes when a message is left for the user. It goes off when the user retrieves the message. The Message light also serves as a visual ringing indicator by flashing quickly while the telephone is ringing.

- **Programmable Dialing Buttons**: The 8102 and 8102M telephones provide 12 programmable dialing buttons on which the user can store frequently dialed telephone numbers or feature codes. To use these preprogrammed features, the user merely lifts the handset and then presses the appropriate programmable dialing button.
- **PROGRAM** and **PAUSE** buttons: Both of these buttons are used for storing numbers on the preprogrammed dialing buttons on the upper half of the faceplate of the telephone. The PAUSE button can be used to insert a one half second pause/delay into a telephone number stored on a programmable dialing button or into a dialing access code in the Auto-Pause feature of Redial.

Other Physical Features

Ringing

The 8102 and 8102M telephones have electronic tone ringing. The ringer volume control is a 3-position switch on the right side of the housing. The user can also use a Tone Selector (labeled “LO HI”) for selecting one of two-tone ringer frequencies or a second Tone Selector (labeled “SLOW FAST”) for selecting one of two-tone ringer modulation rates. Both Tone Selector switches are on the right side of the telephone.

Mounting Options

The 8102 and 8102M telephones come equipped with a non-adjustable desk stand which can be turned upside-down and used for wall mounting the telephone.

Jacks

The 8102 and 8102M telephones have a line jack on the back of the telephone for connecting a line cord, and a Handset cord jack for connecting the handset. These telephones also have a Data jack for connecting a device such as a modem to the telephone. The 8102 and 8102M telephones have an Adjunct (equipment) jack which allows the connection of a speakerphone to the telephone and an adjunct power jack for connecting an auxiliary power source for using the adjunct equipment.

Handset

The 8102 and 8102M telephones are equipped with a K-type handset.

Dial pad

The 8102 and 8102M telephones have a 12-button touch-tone dial pad. The letters “Q” and “Z” have been added to the appropriate dial pad keys for directory access, and the “5” button on your dial pad has raised bars for visually-impaired users.

Cords

Two cords are supplied with the 8102 and 8102M telephones: a coiled 9-foot modular handset cord and a 7-foot modular line cord.

Color Options

The 8102 and 8102M telephones are available in black and in white.

Power Requirements

The 8102 and 8102M telephones must have auxiliary power for adjunct equipment when such equipment is connected..

Power Failure Operation

The 8102 or 8102M telephone can be used as an emergency station during power failure transfer conditions.

Ringer Equivalency Numbers

The Ringer Equivalency Number for the 8102 telephone is: 0.5A and 1.6B. For the 8102M the Ringer Equivalency Number is: 1.5B and 0.8A.

Hearing Aid Compatible

These telephones are compatible with the inductively coupled hearing aids as required by the FCC.

Specialty Handsets

NOTE:
The 8102M is compatible with Avaya K-style Specialty Handsets while the 8102 is not.

3127 SoundPoint Speakerphone Products

The SoundPoint speakerphone adjunct connects to a telephone and provides high quality audioconferencing for desktop environments. The Analog version plugs into any standard analog phone jack. The DCP version plugs into the adjunct port of the 7400, 7500, 8400 and 8500 series terminals.

The full-duplex operation eliminates the clipping of speech and the de-reverberation technology helps to reduce the hollow, “bottom of the well” sound associated with common speakerphones.

SoundPoint automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. Additionally, this adaptation is an automatic and continuous process to accommodate for changing room conditions.

MODELS:

3127-ATR (40785355): AVAYA SOUNDPOINT ANALOG

The analog version of SoundPoint plugs into any analog port or terminal and comes will all appropriate cabling. In Addition, the analog version of SoundPoint can automatically answer (Auto-Answer) calls placed to its extension.

3127-DCP (407645779): AVAYA SOUNDPOINT DCP

Connects to the adjunct port of the 7400, 7500, 8400 and 8500 series terminals for added convenience. Comes with all appropriate cabling.

The SoundStation speakerphone has three microphones providing 360 degree of coverage for offices and small conference rooms. The amplified speaker provides enough audio for groups of up to 10 people. The analog version plugs into any standard analog phone jack. The DCP version plugs into the adjunct port of the 7102, 8102, 7400 and 8400 series terminals.

The full-duplex operation eliminates the clipping of speech and the Acoustic Clarity Technology helps to eliminate background noise. SoundStation automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. Additionally, this adaptation is an automatic and continuous process to accommodate for changing room conditions.

HIGHLIGHTS

SoundStation Speakerphone

- 360° of microphone coverage – ideal for desktop applications
- Digitally tuned speaker with Acoustic Clarity Technology™
- Full Duplex Operation – talk and listen at the same time
- Tabletop, user installable solution for ease of setup and use
- Extension microphones available for up to 20 people
- Integrated dial pad
- Microphone muting for privacy
- Dual color LED for on and mute display
- Wireless, lapel microphone available for stand-up presenters

MODELS

3127-STD (112238): AVAYA SOUNDSTATION ANALOG

The analog version of SoundStation plugs into any analog port or terminal and comes will all appropriate cabling. Requiring just an analog line and a power outlet, this system is easy to install and use. Good for groups of up to 10 people.

3127-EXP (112239): AVAYA SOUNDSTATION EX ANALOG

This analog version of SoundStation includes two extension microphones, which can be used to enhance the room coverage to support up to 15 people. Plugs directly into an analog port.

SoundStation Premier Audoconferencing Speakerphone

The SoundStation Premier speakerphone has three microphones providing 360° of coverage for offices and small conference rooms. The amplified speaker provides enough audio for groups of up to 25 people.

The analog version plugs into any standard analog phone jack, the DCP version plugs into the adjunct port of the 7102, 8102, 7400 and 8400 series terminals and the 6400-SSDP version plugs directly into a DCP jack.

The full-duplex operation eliminates the clipping of speech and the Acoustic Clarity Technology helps to eliminate background noise. In addition, SoundStation Premier focuses a microphone on the main speaker, effectively eliminating the hollow sound common with standard speakerphones. Also included is a full feature remote control and Display.

SoundStation Premier automatically adapts itself to both the room and line conditions in order to ensure the best full-duplex operation. This is especially critical when you have multi-party international conference calls operating over a bridge. This adaptation is an automatic and continuous process to accommodate for changing room conditions.

HIGHLIGHTS

SoundStation Premier Speakerphone

- 360° of microphone coverage – ideal for desktop applications
- Digitally tuned speaker accurately reproduces the human voice
- Full Duplex Operation – talk and listen at the same time
- De-reverberation – reduces the hollow, bottom of the well sound
- Tabletop, user installable solution for ease of setup and use
- Extension microphones available to support up to 25 people
- Integrated dial-pad and full-featured remote control
- Microphone muting for privacy
- Dual color LED for on and mute display
- Wireless, lapel microphone optional for stand-up presenters

MODELS

3127-APE (113987): AVAYA SOUNDSTATION PREMIER EX ANALOG

The analog version of SoundStation Premier plugs into any analog port or terminal and comes will all appropriate cabling. Expansion ports are available for adding on the optional extension microphones. Requiring just an analog line and a power outlet, this system is easy to install and use. Good for groups of up to 15 people.

3127-APX (113988): AVAYA SOUNDSTATION PREMIER EX/MICS ANALOG

This analog version of SoundStation Premier EX/Mics includes two extension microphones, which can be used to enhance the room coverage to support up to 25 people. Plugs directly into an analog port.

8410D Digital Telephone

The 8410D terminal is a multi-appearance digital display telephone that will work in either 2-wire or 4-wire environments. The 8410D is equipped with 10 call appearance/feature buttons, four standard fixed feature buttons, a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, a Volume control button, and 12 feature softkeys.

Physical Features

Dimensions

NOTE:
The dimensions described here for the 8410 voice terminal are approximate.

- Width = 7.8 inches
- Depth (front to back) = 9.5 inches
- Height (maximum with handset in place) = low position, 5.265 inches high position, 6.625 inches

Features

Eight Fixed Feature Buttons

The 8410D has the following eight fixed feature buttons:

- CONFERENCE (labeled CONF) button
- DROP button
- TRANSFER button
- A red HOLD button
- Either a blue SHIFT button or a RING button
- MUTE button
- SPEAKER button
- TEST button

The SHIFT Button

Some 8410D terminals have a blue SHIFT button that can be used in the following ways: (The red light next to the SHIFT button goes on steadily when the button is activated.)

- Used with the HOLD button to select your own personalized ring from among eight available patterns
- Used with the SPEAKER button to perform an acoustic test of the environment and adjust the speakerphone to the surrounding acoustic environment for optimal performance.

Call Appearance/Feature Buttons

The 8410D terminal has 10 call appearance/feature buttons, each equipped with a red light and a green status light. Usually, at least three of these buttons are administered as call appearances for the terminal's primary extension number, where calls are placed and answered. The rest of the buttons can be used for more call appearances (including bridged appearances of other stations' extensions) and features. Buttons administered for features have only their green status lights active.

Message Light

The red Message light goes on when a message is left for the user. It goes off when the user retrieves the message or when the message is erased.

Speakerphone

The 8410D terminal is equipped with a built-in speakerphone. (The voice terminal can be optioned for a 1-way, listen-only speaker instead.) The Speakerphone capability allows a user to engage in a hands-free conversation with the far-end party. The speakerphone can be turned on or off with the SPEAKER button; can be muted with the MUTE button, and the volume can be raised and lowered with the VOLUME "arrow" button. If the voice terminal has a SHIFT button and Rest Spkr appears below the SPEAKER button, the user can access the Reset Speakerphone feature to adjust the speakerphone for optimal performance.

Speaker (Listen-Only) Feature

Although the 8410D terminal is shipped from the factory with the terminal set for the Speakerphone feature, it can be optioned instead for a 1-way, listen-only speaker. If this is the case, use the SPEAKER button on the front of the voice terminal for turning on and off the speaker. The Speaker (listen-only) capability allows a user to engage in a 1-way conversation with the far-end. Specifically, the user can only listen to the far-end. To speak with the far-end, the user must use the handset, which turns off the speaker. The Speaker function may also be disabled.

MUTE Button

The MUTE button is a fixed feature button with a red status light. The MUTE button turns off the microphone of the built-in speakerphone or the handset so that a person can converse with another person in the room without the other party listening in.

VOLUME Control Button

The 8410D voice terminal has a VOLUME "arrow" button on the front of the set. This button has several functions. When the user is using the speakerphone or the 1-way speaker, the VOLUME button controls the volume of the speaker. When the user is on a call using the handset, the VOLUME button controls the handset volume. When the voice terminal is idle, the button controls the volume of the tone ringer. When the user sets the volume for the speaker, speakerphone, handset, or the tone ringer, the display on the 8410D terminal shows a "bar array" that indicates the volume setting.

TEST Button

When the voice terminal is initially powered up, the green light next to the TEST button flashes if the link with the PBX is not (or not yet) operational. The light changes to steady green when the voice terminal is able to communicate with the PBX. After the voice terminal is powered up, you can press the TEST button to test the lights and the display on your voice terminal.

Display

The display on the 8410D set has two lines with 24 characters on each line. The display, along with the softkeys and the display control buttons, can be used to access 12 softkey features (the default softkey features and the alternative softkey features are listed under “Softkeys”).

Softkeys

The four buttons located below the display and labeled with arrows correspond to features listed on the second line of the display screen. (There are four features on each feature option screen.) The user can access any of the 12 features by pressing the softkey below the feature abbreviation.

Handset

The 8410D voice terminal is equipped with a K-type handset.

Dial Pad

The 8410D terminal has a 12-button touch-tone dial pad. The letters “Q” and “Z” have been added to the appropriate dial pad keys for directory access, and the “5” button on the dial pad has raised bars for visually-impaired users.

Jacks

The 8410D terminal’s housing has a Line jack located on the back of the housing, and a Handset cord jack located on the side of the set. There is also a Speakerphone/headset adapter jack to which the user can connect adjunct equipment.

Cords

Two cords are supplied with the 8410D: a coiled 9-foot handset cord and a 7-foot line cord. Optional longer cords are available: a 12-foot handset cord and 14-foot and 25-foot line cords.

Ringing

The 8410D voice terminal has electronic tone ringing with eight possible ringing patterns.

Mounting Options

The 8410D terminal comes equipped with a 2-position desk stand. This stand can also be removed so the user can mount the set on the wall.

Color Options

The 8410D terminal is available in black and in white.

Wiring Information

The 8410D works in both 4-wire and 2-wire configurations.

Power Requirements

The 8410D terminal receives power from the system and does not require any external power supply. Additional external power IS required when the 8410D voice terminal is equipped with any adjuncts, such as an external speakerphone. In this instance, an 1151A1 local power supply or a 1151A2 power supply with Battery Holdover is recommended.

Power Failure Operation

The 8410D terminal cannot be used as an emergency station during power failure transfer conditions.

FCC Registration

The 8410D terminal is FCC-registered along with the switch (as a system), but does not have a separate FCC registration label.

UL and CSA Approval

The 8410D terminal has been tested and has met the Underwriters Laboratories (UL) Standards UL 1459 and has also met the Canadian Standards Association (CSA) Standards CSA-C22.2 No.225-M90.

Hearing Aid Compatible

The 8410D terminals are compatible with the inductively coupled hearing aids prescribed by the FCC.

8434DX Digital Voice Terminal

The 8434DX is a two-wire or four-wire digital voice terminalthatis ideal for executives, secretaries, or call center supervisors requiring multiple lines for call handling, call coverage, conferencing, and sophisticated communications. For unsurpassed readability of call information, messages, and the current date or time, the 8434DX features a built-in two line by 40 character vacuum fluorescent display. The 8434DX offers this state of the art technology which virtually eliminates blurring and distortion of the display, even under harsh lighting or at odd angles.

The 8434DX voice terminal is equipped with the following:

- 34 Call appearance/feature buttons
- Nine fixed features - Hold, Conference, Transfer, Drop, Volume Control, Mute, Speaker, Test, and Shift
- 2-line by 40 character vacuum fluorescent display 15 Assignable soft key features
- Built-in programmable speakerphone which can be converted to a one-way, listen-only speaker
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Eight personalized ringing options
- Adjunct jack for speakerphone or headset
- K-Type handset with nine foot modular cord
- Seven foot modular line cord
- International portability
- Downloadable transmission parameters
- Available in Black Additional options include:
- 12-Foot modular handset cord
- 14-Foot and 25-foot modular line cords
- Avaya/Plantronics headsets
- ompatible modules
- Speakerphone S201A/CS201A/Polycom
- 500A Headset adapter
- 7400B+ data module (four-wire mode)
- 8400B+ data module (two-wire mode)
- Amplifier handset

8510T ISDN BRI Voice Terminal

The 8510T ISDN BRI (Basis Rate Interface) Voice Terminal is an all around, multi-purpose, mid-range product intended to provide a full array of features for the very active telephone user or busy manager allowing quick and easy access to switch and personal features. This terminal is ideal for the majority of staff workers, lower through top level managers, for conference rooms, small office/home office environments and wall mounted applications (with optional wall mount) where displays are necessary. The terminal is designed with the future in mind, with a modular processor interface to allow for the addition of feature access boards for data, video and other (as yet undefined) functions which make this terminal a true application platform.

The 8510T ISDN Voice Terminal delivers market driven styling with a hearing aid compatible K-type handset, rubber dome buttons and a plug-in ROM board that allows for conformity with evolving standards and international transmission requirements.

The 8510T provides the following:

- Ten buttons for call appearances (each with a red and green LED) that may also be administered as switch feature buttons
- Nine fixed feature buttons (Hold, Transfer, Conference, Drop, Redial, Speaker, Mute, Volume, and Exit)
- A red message waiting LED, a built-in speakerphone which can be optioned as a monitor only or completely disabled
- A plug-in ROM Board
- Adjunct jack for the Digital and Analog SoundPoint Speakerphone Modules
- 500A Headset Adapter
- A super-twist 2x24 liquid crystal display (LCD) with four control keys and four soft keys
- Soft keys are keys whose functions and labels change dynamically, depending on the function performed by the user. Features that the soft keys offer are a local directory of up to 30 names and numbers, incoming and outgoing call logs, display contrast adjustment, ringer pattern selection, self-test, clock setting, and a 3-digit password lock.

Power Requirements

Power for the 8510T Voice/Data Terminal Telephone is provided by a Auxiliary Power Supply in the wiring closet or in the office near the telephone.

8520T ISDN BRI Voice Terminal

The 8510T ISDN BRI (Basis Rate Interface) Voice Terminal is an all around, multi-purpose, mid-range product intended to provide a full array of features for the very active telephone user or busy manager allowing quick and easy access to switch and personal features. This terminal is ideal for the majority of staff workers, lower through top level managers, for conference rooms, small office/home office environments and wall mounted applications (with optional wall mount) where displays are necessary. The terminal is designed with the future in mind, with a modular processor interface to allow for the addition of feature access boards for data, video and other (as yet undefined) functions which make this terminal a true application platform.

The 8510T ISDN Voice Terminal delivers market driven styling with a hearing aid compatible K-type handset, rubber dome buttons and a plug-in ROM board that allows for conformity with evolving standards and international transmission requirements.

The 8510T provides the following:

- Ten buttons for call appearances (each with a red and green LED) that may also be administered as switch feature buttons
- Nine fixed feature buttons (Hold, Transfer, Conference, Drop, Redial, Speaker, Mute, Volume, and Exit)
- A red message waiting LED, a built-in speakerphone which can be optioned as a monitor only or completely disabled
- A plug-in ROM Board
- Adjunct jack for the Digital and Analog SoundPoint Speakerphone Modules
- 500A Headset Adapter
- A super-twist 2x24 liquid crystal display (LCD) with four control keys and four soft keys.
- Soft keys are keys whose functions and labels change dynamically, depending on the function performed by the user. Features that the soft keys offer are a local directory of up to 30 names and numbers, incoming and outgoing call logs, display contrast adjustment, ringer pattern selection, self-test, clock setting, and a 3-digit password lock.

Power Requirements

Power for the 8510T Voice/Data Terminal Telephone is provided by a Auxiliary Power Supply in the wiring closet or in the office near the telephone.

TransTalk 9040

The TransTalk 9040 is unique in the marketplace, so you get the competitive edge! Its smaller size, full feature set, and alphanumeric display give users complete control of their wireless communications. And, its 8-ounce flip-less design incorporates all the capabilities of our previous TransTalk System handsets.

Here are some of the highlights:

Display

A 1x16 alphanumeric display shows internal calling party information and the external called number. Users have more control because they can see who's calling and decide to take the call in real time or transfer it to voice mail. The backlit display also includes three rows for line/intercom terminations and one-button feature access. In addition, icons such as out-of-range, low battery, and message waiting are visible on the display. Customers can also use their Pocket Phones to access their host system directory.

Smaller Size

The TransTalk 9040 weighs just eight ounces, and its dimensions are 6" x 2" x 1"

Capacity

The TransTalk 9040 has 10 or 12 virtual button appearances for lines, intercoms, and features.

Rapid/Upright Battery Charger

- With fast charging battery capability built into both the handset cradle charger and the spare charger, batteries charge fully in only 1.5 hours. With each charge:
- Batteries are discharged and recharged, which eliminates the memory effect that reduces battery life (spare is automatically reconditioned; handset battery is reconditioned in the cradle if manually selected)
- Users get 3.5 hours of talk time, and 22+ hours of standby time
- Users can continue to screen calls because the upright position keeps the display clearly visible In addition, an optional extended-use battery, will provide up to 8 hours of talk time and 72 hours of standby time.

Easy Carrying

Your customers get the convenience of a new removable belt/pocket clip and a new lanyard (wrist strap) built into the Pocket Phone for easy carrying.

Other Great Handset Features

The TransTalk 9040 Pocket Phone offers:

- Backlit display: for easier use in poorly lit areas such as warehouses and manufacturing plants.
- Vibrator Alert: for times when a ringing phone would be intrusive (standard on all Pocket Phones).
- Redial Button: fixed redial for busy numbers.
- Replaceable Antenna: customers can replace their own antennas in case of breakage.
- Headset Option: works with the Supra (over-the-head) or Radium (over-the-ear) mobility headsets. Can be used with the wireless headset for the MDW 9000 and MDW 9010 with an adaptor. New 2.5mm jack makes connecting a headset to the 9040 easier than ever.
- Field registration: If a handset becomes needs to be replaced, only the handset needs to be returned (i.e. the corresponding Radio Module does not have to be shipped back with the handset). When a new handset is received, the user or technician simply registers it with the appropriate DRM.

Callmaster IV Digital Terminal

The Avaya Callmaster IV is engineered to support applications involving the Automatic Call Distribution (ACD) feature of the MultiVantage software. The ergonomic design of the Callmaster IV enables agents to handle large volumes of calls more quickly, efficiently, and productivelyæin customer service, order processing, collections, account management, or any communications-intensive activity. VuStats display of agent and call center statistics on the Callmaster IV provides agents with real-time information they can use to improve their own performance and that of the call center.

Callmaster IV was designed to work on a 16- or 24-port, 2-wire Digital Line Circuit Card, but it will also work in a 4-wire mode with an 8-port card associated with earlier DEFINITY system releases. The Callmaster IV can automatically detect whether it is plugged into a 2-wire or 4-wire circuit card, providing for an easy transition to either environment, as well as reduced wiring expenses and installation change adjustments.

The Callmaster IV includes as standard a built-in Recorder Interface Module (RIM) supporting both 2-wire and 4-wire connections to agent recording equipment.

The Callmaster IV digital terminal is equipped with the following:

- Six rubber domed administerable call appearance/flexible feature buttons
- 15 Rubber domed administerable flexible feature buttons
- Eight fixed feature buttons - Conference, Transfer, Drop, Hold, Mute, Volume, Release, and Login
- Built-in 80 character (2-line by 40 character) alphanumeric LCD display with adjustable viewing angle
- 12-Button touch-tone dial pad with raised bar on "5" for the visually impaired
- Message waiting light (LED)
- Recorder Interface Module (2-wire or 4-wire)
- Adjunct jack for speakerphone
- Dual headset jacks
- Eight personalized ringing options
- Receiver and ringer volume control
- Stand for desk * International portability
- Downloadable transmission parameters allow the terminal to automatically detect whether it is plugged into a 2-wire or 4-wire digital line circuit card

Callmaster V Digital Terminal

The Avaya Callmaster V has been specially designed to support applications involving the Automatic Call Distribution (ACD) feature of the MultiVantage software. The ergonomic design of the Callmaster V enables agents to handle large volumes of calls more quickly, efficiently, and productively-in customer service, order processing, collections, account management, or any communications-intensive activity. VuStats display of agent and call center statistics on the Callmaster V provides agents with real-time information they can use to improve their own performance and that of the call center.

The Callmaster V has the same look and feel of the 6400 Series telephones. There are two significant additional features that maximize the value of this telephone in a Call Center environment.

- **2-Built-in Headset Jacks** - The Callmaster V is designed to use Avaya headsets.
- **Built-in Recorder Interface Module (RIM) with Warning Tone** - Will support recording of both the agent's and caller's voice on a voice activated analog tape recorder. A soft beep warning tone is repeated every 13.5 seconds to notify the agent and calling party that the call is being recorded (user can deactivate).

The Callmaster V voice terminal features an adjustable 48 character (2¦lines by 24 characters) Liquid Crystal Display which provides agents with display of ACD messages, unified messaging access, and call related information, including Dialed Number Identification Service (DNIS), Automatic Number Identification (ANI), and VuStats.

The Callmaster V digital terminal is equipped with the following:

- 16 Dual LED call appearance/feature buttons
- 10 Fixed features - Speaker, Mute, Conference, Transfer, Hold, Redial, Menu, Exit, Previous, and Next
- 12 Assignable soft key features associated with the display
- Built-in one-way, listen-only speaker for group listening, on¦hook dialing, or hands free listening
- Adjustable volume control (handset, speaker, and ringer)

Avaya IP Agent

Avaya IP Agent

The growth of call centers has brought many new challenges to business, including retaining agents and maintaining agent satisfaction, running a continuous 24-hour call center operation, and managing fixed location sizes in the face of growth. Companies are looking for new technologies and ideas to help them manage these situations. Avaya IP Agent addresses these business issues by providing a Softphone remote agent solution that accommodates all the Avaya call center agent features and capabilities for agents working remotely from home or a satellite location. Avaya IP Agent uses the Avaya softphone specifically intended for call center/CRM agents, whereas the IP Softphone is intended for other kinds of business applications.

Avaya IP Agent includes a set of features that facilitates an agent's tasks via:

- One-button access to frequently used features, such as Answer, Conference, Transfer, login, logout, and release
- Screen Pop Capability (via HTML or DDE hooks)
- LDAP access to corporate directories
- Incoming Call Alerter (audio and visual)

Avaya IP Softphone

AVAYA IP Softphone is for employees who work out of the main office, on the road, or at home, communications have just gotten easier with the Avaya Internet Protocol (IP) Softphone Release 3. Accessed through a simple graphical user interface on a personal computer or laptop, the Avaya IP Softphone gives the mobile worker anywhere, anytime access to voice, fax, data, and e-mail, as well as the full suite of MultiVantage features and functions. It brings the power of the MultiVantage multiple appearance digital voice terminal to the remote user's computer screen.

IP Softphone delivers MultiVantage software to remote users. These features include high-quality Voice over IP, multiple Call Appearances, Caller ID, six-party Conference, Autodial, Abbreviated Dialing Lists, Send All Calls, Call Coverage, visual Message Waiting Indication, and many other features that increase productivity and overall cost savings. The Avaya IP Softphone Release 3 is the third generation of the Avaya IP Softphone. IP Softphone is a client-based telephony application for the desktop PC or laptop.

MultiVantage solutions deliver call process via a IP connection to the remote IP Softphone, providing the remote user with the exact same capabilities of a local digital, multi-appearance extension user. Call control/feature control are also provided via the IP connection. An example of call control is the signaling and call setup/teardown required for delivery of six-party conferencing to IP Softphone.

Call traffic (simultaneous voice and data conversations) are delivered to IP Softphone in one of two ways. Two configurations are available to ensure the right solution for each remote user: Voice over IP (Road Warrior) or circuit-switched voice (Telecommuter). Avaya *iClarity* IP Audio, which is integrated in the Road Warrior application, provides world-class IP voice sound quality.

The Telecommuter configuration utilizes two MultiVantage ports, one for call control and one for voice traffic. An excellent application is a home office extension user who requires a PC, prefers to talk over a traditional telephone line, and needs the same capabilities provided to local digital extension users.

In the Road Warrior configuration, *a single* MultiVantage port provides LAN connection to a remote PC running IP Softphone software, which provides IP call control *and* call traffic handling. MultiVantage single-extension protocol simplifies administration, reduces hardware connections, and increases system capacities. An employee who travels on business can simply access the Internet to have all the features of his/her digital, multi-appearance office phone with display.

Lightweight Directory Access Protocol (LDAP) client allows IP Softphone access to LDAP-compatible databases, such as Enterprise Directory Gateway. The Avaya IP Softphone is also CTI/TAPI-enabled.

IP Softphone provides a Call History Log (both incoming and outgoing calls), local phone directory and other features designed to improve user productivity.

The Avaya IP Softphone Release 3 features include:

- Choice of a totally IP solution (Road Warrior), or a combination of IP and traditional voice solution (Telecommuter). This provides the flexibility to fully meet the specific needs of each remote user. It also opens choices regarding the number of ports your business chooses to use for remote users (single port for Road Warrior, and dual port for Telecommuter).
- Integrated iClarity (developed by Avaya Labs) H.323 client provides superior Voice over IP sound quality.
- Support for alternate Gatekeeper addresses increases reliability. IP Softphone accepts several Gatekeeper addresses via industry standard Dynamic Host Configuration Protocol (DHCP) or manual administration. IP Softphone provides a user pull-down menu of alternate gatekeeper addresses from which the user can select, if attempts to contact the top priority address fail.
- Administrable Quality of Service (QOS) parameters enhance voice quality. IP voice needs to be delivered quickly to ensure quality. QOS prioritizes Voice over IP traffic over other network traffic.
- Software upgrades are remotely downloadable to IP Softphone, providing an efficient upgrade process, as well as future support for expanded features and applications.
- Avaya IP endpoints (such as IP Softphone) are industry standards-based IP endpoints. This allows the flexibility to support emerging standards and applications, which provides your business with long-term investment protection.
- Enhanced E911 service options: Administrable options, on a station-by-station basis, to either send the IP endpoint’s extension number, send a number specified by the caller, or block E911 calls and force the caller to place E911 call over local trunk.
- Flexibility via emulation of a variety of MultiVantage voice terminals
- Easy-to-use voice terminal controls via graphical PC representation of digital, multi-appearance voice terminal, with point-and-click feature button control. Feature button LEDs illuminate when in use, allowing simplified control and use of a wealth of available features.
- Six-party conferencing delivered to remote office.
- Conference, Hold, Mute, Redial, and Volume control via fixed feature buttons.
- Programmable feature buttons with LED, and display-associated, programmable softkeys make features readily available to user.

- Access to the industry-leading 525+ feature-set of Avaya Call Processing (except call center agent features) help increase productivity, save time and money, and enhance customer service.
- Access to additional productivity tools – Phonebook, Call History Log, LDAP Directory client.
- Launch calls from TAPI-compliant Personal Information Managers.
- Support for standard audio codecs used in the IP industry: G.711 (A-law and Mu-law), G.723.1, and G.729 (including G.729A, G.729B, and G.729A+B) equates to enhanced IP voice quality.
- Multilanguage support: Brazilian Portuguese, English, European French, German, Japanese, Simplified Chinese, Korean, and Latin American Spanish.
- Choice of two easy-to-use graphical user interfaces: Call Bar View and Phone Picture View.

Administrable Quality of Service (QoS)

IP voice needs to be delivered quickly to ensure quality. QoS prioritizes Voice over IP traffic ahead of other network traffic in your Local Area Network (LAN) or Wide Area Network (WAN). Avaya IP Softphone QoS parameters include proven industry-standard Differentiated Services (DiffServ), 802.1p, and 802.1Q.

DiffServ uses the Type of Service (TOS) value to prioritize voice packets at Network Layer 3 of IP Protocol stack. The 802.1p and 802.1Q values provide eight levels of priority to increase throughput of voice packets at Data Link Layer 2 of IP Protocol stack. 802.1p and 802.1Q also allow identification of several Virtual LAN (VLAN) identifiers. The VLAN identifier at Layer 2 permits segregation of traffic to reduce traffic on individual links, increasing Voice over IP quality.

Voice over IP traffic can also be prioritized by administering IP User Datagram Protocol (UDP) port range values (used at Transport Layer 4 of the IP Protocol stack).

Avaya solutions allow administration by Network Region. A Network Region is a portion of your network that contains multiple IP endpoints (typically a group of endpoints with a common interest or function, such as a single campus in a multi-campus university). Network Regions can contain LAN Segments, which are a more specific grouping of IP endpoints within the Network Region. Administration by Network Region can provide additional flexibility to control traffic on your network, and reduce overall network traffic. One major advantage is enhanced IP voice quality.

Avaya solutions support the download of QoS parameters to IP endpoints (such as IP Softphone). QoS parameters download to an IP endpoints whenever the IP endpoint registers with a MultiVantage server, and whenever an administered QoS parameter changes values.

Avaya provides industry-leading Voice over IP solutions. These tools are only a few examples of how Avaya provides your business with effective control over the *quality of IP voice*.

IP Softphone Remote Worker Support for E911 Emergency Services

In an E911 emergency services network, equipment and dispatchers at the Public Safety Answering Point (PSAP) use the caller's number to specifically determine where to send emergency personnel. Avaya IP Softphone Release 3 allows options for your system administrator that can be administered on a station-by-station basis. MultiVantage solutions can be administered to send the IP endpoint's extension number, send a number specified by the caller during registration, or block the call and force the caller to place a 911 call locally.

This IP Softphone feature works in conjunction with the other industry-leading E911 solutions of its host including CAMA trunks, Automatic Number Identification over ISDN-PRI, as well as crisis alerting to attendant consoles, to multi-button sets, and to digital pagers. Avaya delivers optimum E911 services.

IP Softphone Minimum Requirements

- MultiVantage software (9.5 or later).
- MultiVantage server
- IBM-compatible PC with Intel Pentium 200 MHz (300 MHz recommended for Road Warrior) or compatible processor.
- Hard Disk with at least 30 MB of space available.
- 64 MB of RAM for Microsoft Windows NT 4.0 Server or Workstation with Service Pack 4.0, 128 MB of RAM for Microsoft Windows 2000 Server or Professional, and 32 MB (64 MB recommended for Road Warrior) for Microsoft Windows 98. Windows 95 is not supported.
- Sound card that supports full-duplex operation (for the Road Warrior and standalone H.323 configurations only).
- A speaker/headset and a microphone (for the Road Warrior and stand-alone H.323 configurations only). External microphone recommended for laptops.
- Second telephone line for telecommuter dual connect configuration only.
- Network Interface Card (NIC) for local area network (LAN) connectivity and/or a modem (28.8 Kbps or faster) for dial-up networking.

Softconsole

AVAYA Softconsole is a next generation software attendant console solution for Avaya customers. It builds upon the success of previous attendant solutions, PC Console R2.5 and the 302 hardware attendant console. AVAYA Softconsole is available for industry standard IP and Avaya Digital Communications Protocol (DCP) connectivity to the switch included on the same compact disk (CD).

AVAYA Softconsole Highlights

User Interface

- New emphasis on keystroke reduction/speed of operations for keyboard-only attendants.
- Busy Lamp Fields (BLF), Directory and Display windows may all be on the same screen and the same time.
- Flexible screen arrangement for the attendant. Screen arrangement saved from session to session.
- Application window scales intelligently from a minimum useful size to full screen. Useful information is added to the display as the attendant increases the window size.
- On request line status (On/Off-hook, Send All Calls) displayed for the selected entry in the directory window.
- Queue status display.
- Feature buttons offered as tools in multiple tool bars with pop-up, full word, tool tip displays for each.
- 32-bit Application

Enhanced Directory functionality

- Up to 100 directories can be created.
- Download of MultiVantage directory now integrated with Master Directory.

E-mail

- Ability to generate e-mail to users at the click of a tool bar button or keyboard command.

Comprehensive guidance during installation

- Step-by-step wizard for both installation and initial administration with help and warning text presented with each step.
- Targeted to reduce service call volumes for installation assistance.

Compatible Optional Applications and Benefits

Enterprise Directory Gateway (EDG)

The Enterprise Directory Gateway expands the scope of traditional data management by delivering a directory infrastructure for Avaya telephony servers (MultiVangae, INTUITY™ Multimedia Messaging). EDG uses Avaya Lightweight Trigger Access Process (LTAP) patented technology to deliver the notifications required to synchronize data from voice and messaging servers with an LDAPv3 data store. Avaya Enterprise Directory Gateway consists of an Administration Module, DataStore Managers, a Synchronization Engine, a browser-based client, a data manager (MasterDirectory), and a customer-supplied LDAPv3-compliant data store.

PhoneticOperator

PhoneticOperator is only available in the U.S. through the direct sales channel. Check with all channels for availability and support prior to proposing.

PhoneticOperator is an auto-attendant software application that answers incoming calls and uses advanced speech recognition to quickly and easily route the calls to the destinations that the callers request. PhoneticOperator resides on a PC that is connected to analog extensions on a MultiVantage solution via a Natural Microsystems analog telephony card. The number of ports connected to the server determines the number of concurrent calls that PhoneticOperator can handle. These ports (referred to as *channels*) are assigned to the telephone number you want PhoneticOperator to cover.

When an incoming call rings on the telephone number that PhoneticOperator is covering, PhoneticOperator answers the call and plays a script that you have specified. Typically, this script will greet the callers and then prompt them to say the name of the person to whom they want to speak. PhoneticOperator then searches a directory that you created with MasterDirectory to find the extension for the requested person. Once the extension is located, PhoneticOperator routes the call automatically to the appropriate extension.

MasterDirectory Data Manager

MasterDirectory Data Manager is included as part of AVAYA Softconsole and the Enterprise Directory Gateway. It is a database application that is specifically designed for directory data management. It serves as an information management tool–importing and consolidating directory information from voice and data systems, and exporting it to directory-enabled applications. MasterDirectory can import/export and transfer data via standards-based protocols, including:

- ODBC – Open Data Base Connectivity
- LDAP – Lightweight Directory Access Protocol
- FTP – File Transfer Protocol
- SMTP – Simple Mail Transfer Protocol
- CSV – text delimited files

Using these protocols, MasterDirectory can extract data from multiple sources, apply filters and business logic to consolidate the data, and populate directory services and databases for use by applications. For example, MasterDirectory can collect information from multiple MultiVantage/Definity servers, consolidate the data with Human Resource databases, and send the processed data to an LDAP directory service used by phone attendant applications, intranet white and yellow pages, and other applications.

Languages

AVAYA Softconsole will initially be available only in U.S. English. Additional languages will follow in a later release.

Microsoft NetMeeting

NetMeeting Microsoft conferencing is software that you can use to communicate with both audio and video, collaborate in Windows-based programs, exchange graphics on an electronic whiteboard, transfer files, or use a text-based chat program. Common uses of NetMeeting include real-timesoft document collaboration, technical support in a helpdesk environment, training and distance learning, and conducting remote meetings.

XM24 Expansion Module

The internationally styled XM24 24 button expansion modules work with all versions of the 6416D+, 6416D+M, 6424D+, or the 6424D+M digital voice terminals. The XM24 is universal and not language or region specific.

The features include:

- 24 Call Appearance/Feature buttons with dual LEDs
- The XM24 expansion module requires power either at the terminal with a station power supply or in the wire closet via a bulk power supply.
- Can be used with or without stand.
- There are two different stand heights. Order appropriate code to match the terminal the XM24 will go with.
- Lower stand for use with the 6416D+ & 6424D+
- Higher stand for use with the 6416D+M & 6424D+M.
- Not wall mountable.
- 2 ft. 3 in. (.69 m) modular cord that connects the XM24 to the voice terminal.
- Available in Dark Gray and White

Adjuncts

Optional External Equipment

INTUITY™ AUDIX®

On a typical day, employees retrieve voice messages from their telephones, documents from the FAX machine down the hall and email at the computer. Message management devours time, and urgent communication can easily be overlooked. With the INTUITY AUDIX message server, employees can more quickly retrieve, organize and respond to messages by priority because all messages, including voice, email and fax can be accessed from either a telephone or PC, at work, at home or on the road. Everything is standard. Integrated email and FAX messaging make it easier to collaborate with partners, suppliers and others outside your messaging network. With access to voice, fax and email messages from the same place, employees can more quickly spot urgent messages, imporving collaboration and customer service.

Highlights of INTUITY AUDIX include:

- Visual desktop access to messages via Message Manager
- Integrated FAX messaging
- Analog, Digital and IP network capability
- LAN/IP card for integrating email messaging and digital networking
- Text to speech conversion for retrieving emails by phone
- Traditional user interface
- PC based mailbox administration

Avaya Call Management System (CMS)

Avaya CMS is an integrated analysis and reporting tool used in a Call Center Environment. It keeps tabs on everything that is going on such as gathering information on just one agent, a group of agents, a single contact center or multiple locations around the world.

Avaya CMS is flexible and scalable. It’s ideal for small single-location contact centers, in large multi-loation applications.

CMS Highlights:

- Analyze call flow
- Match your personnel resouces and skills to call vloumes and caller needs
- Identify areas where you can increase productivity and reduce costs
- Plan new marketing and promotional campaigns
- Identify training needs
- Take fast, effective action to improve the overall effeciency of your contact center operations

- Integrate data with mainframe-based documents such as sales and marketing reports
- More than 200 preformatted reports help call center managers see where they are achieving and where they need improvements

Call Detail Reporting

Call Detail Reporting (CDR) is the name of a type of software application that resides on a server connected to the MultiVantage or DEFINITY solution. The CDR appliation collects call records from the server or switch and stores them for reports. The reports can be used for many purposes including billing, administration and proactive maintenance. Some example of vendors that provide CDR software are Veramark and Quantam.

Example report Categories:

- Organization reports track calls made by individuals or groups and associates the calls to the corporate organization hierarchy for simplified charge-back.
- Call Search reports allow call records to be analyzed through broad or narrow search criteria.
- Exception reports highlight longest and most expensive calls as well as trunks without call activity.
- Traffic reports provide a network-wide view of usage by call type, trunk, duration, area code, cost and other variables.
- Account Code reports track calls made on-behalf of clients or projects and can be used for associated billing.

BCMSVu™

BCMSVu is a software application that works in conjunction with MultiVantage Software. This product will provide Call Center Managers with the ability to store data for greater then seven days, a graphical view of real time reports in a Windows environment and a driver to “push” data to wallboards. In addition, BCMS-Vu provides “point and click” historical report data transfer to third-party spreadsheet applications such as Microsoft Excel and Lotus 1-2-3. The application will reside on a server with a Microsoft operating system. The amount of data that can be stored is determined by the capacity of the server.

IP Agent

IP Agent is a softphone agent solution that accommodates all the MultiVantage Call Center agent features and capabilities for agents working remotely or in an office location. Agents ahve access to the full range of MultiVantage agent capabilities using a intuitive graphical user interface (GUI) that works via standard Microsoft Windows drag and drop conventions. The IP Agent solutions includes an interface to access existing cororate database information via LDAP (Lightweight Directory Access Protocol) and an integrated call history feature that allows agents a detailed view of the calls made and received. In addition, contact center managers can administer screen pops based on commonly used triggers, such as dialed number identification services (DNIS), automatic number identification (ANI) and prompted digits.

OCTEL® Voice Messaging

The OCTEL Voice Messaging design performs as well in small voice messaging environments as it does in high-capacity voice, fax and data processing environments. The servers provide high-performance and feature-rich processing solutions over a wide range of configurations.

Reasons to invest in OCTEL:

- Proven server performance and reliability
- Ease of administration
- Smooth growth and expansion through scaleable system architecture
- Comprehensive integration capabilities
- Intuitive and feature-rich user interface options
- Strong international messaging solution with comprehensive service and support
- Powerful applications
- Muti-user interface options
- Digital and analog networking options
- IP-enabled capabilities

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