Avaya Aura® Communication Manager Hardware Description and Reference

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Notice

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

Warranty

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

Preventing Toll Fraud

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

Avaya Fraud Intervention

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

Disclaimer

Avaya is not responsible for any modifications, additions or deletions to the original published version of this documentation unless such modifications, additions or deletions were performed by Avaya. Customer and/or End User agree to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation to the extent made by the Customer or End User.

How to Get Help

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

- Within the United States, click the *Escalation Contacts* link that is located under the *Support Tools* heading. Then click the appropriate link for the type of support that you need.
- Outside the United States, click the *Escalation Contacts* link that is located under the *Support Tools* heading. Then click the *International Services* link that includes telephone numbers for the international Centers of Excellence.

Providing Telecommunications Security

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

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

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

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

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

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

Responsibility for Your Company's Telecommunications Security

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

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

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

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

TCP/IP Facilities

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

Product Safety Standards

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

- IEC 60950-1 latest edition, including all relevant national deviations as listed in the IECEE Bulletin—Product Category OFF: IT and Office Equipment.
- CAN/CSA-C22.2 No. 60950-1 / UL 60950-1 latest edition.

This product may contain Class 1 laser devices.

- Class 1 Laser Product
- Luokan 1 Laserlaite
- Klass 1 Laser Apparat

Electromagnetic Compatibility (EMC) Standards

This product complies with and conforms to the following international EMC standards, as applicable:

- CISPR 22, including all national standards based on CISPR 22.
- CISPR 24, including all national standards based on CISPR 24.
- IEC 61000-3-2 and IEC 61000-3-3.

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.

Federal Communications Commission Part 15 Statement:

For a Class A digital device or peripheral:

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

For a Class B digital device or peripheral:

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

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

Equipment With Direct Inward Dialing ("DID"):

Allowing this equipment to be operated in such a manner as to not provide proper answer supervision is a violation of Part 68 of the FCC's rules.

Proper Answer Supervision is when:

A. This equipment returns answer supervision to the public switched telephone network (PSTN) when DID calls are:

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

B. This equipment returns answer supervision signals on all (DID) calls forwarded back to the PSTN.

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.

Automatic Dialers:

When programming emergency numbers and (or) making test calls to emergency numbers:

- Remain on the line and briefly explain to the dispatcher the reason for the call.
- Perform such activities in the off-peak hours, such as early morning or late evenings.

Toll Restriction and least Cost Routing Equipment:

The software contained in this equipment to allow user access to the network must be upgraded to recognize newly established network area codes and exchange codes as they are placed into service.

Failure to upgrade the premises systems or peripheral equipment to recognize the new codes as they are established will restrict the customer and the customer's employees from gaining access to the network and to these codes.

For equipment approved prior to July 23, 2001:

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

For equipment approved after July 23, 2001:

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the Administrative Council on Terminal Attachments (ACTA). On the rear of this equipment is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXX. If requested, this number must be provided to the telephone company.

The REN is used to determine the quantity of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0.
L'indice d'équivalence de la sonnerie (IES) sert à indiquer le nombre maximal de terminaux qui peuvent être raccordés à une interface téléphonique. La terminaison d'une interface peut consister en une combinaison quelconque de dispositifs, à la seule condition que la somme d'indices d'équivalence de la sonnerie de tous les dispositifs n'excède pas cinq.
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. For products approved after July 23, 2001, the REN for this product is part of the product identifier that has the format US:AAAEQ##TXXX. The digits represented by ## are the REN without a decimal point (for example, 03 is a REN of 0.3). For earlier products, the REN is separately shown on the label.

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 premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2.T	AS.2	RJ2GX, RJ21X, RJ11C
CO trunk	02GS2	0.3A	RJ21X, RJ11C
	02LS2	0.3A	RJ21X, RJ11C
Tie trunk	TL31M	9.0F	RJ2GX
Basic Rate Interface	02IS5	6.0F, 6.0Y	RJ49C
1.544 digital interface	04DU9.BN	6.0F	RJ48C, RJ48M
	04DU9.1KN	6.0F	RJ48C, RJ48M
	04DU9.1SN	6.0F	RJ48C, RJ48M
120A4 channel service unit	04DU9.DN	6.0Y	RJ48C

If this 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.

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

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

Installation and Repairs

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. It is recommended that repairs be performed by Avaya certified technicians.

FCC Part 68 Supplier's Declarations of Conformity

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/DoC.

Canadian Conformity Information

This Class A (or B) digital apparatus complies with Canadian ICES-003.

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

This product meets the applicable Industry Canada technical specifications/Le présent materiel est conforme aux specifications techniques applicables d'Industrie Canada.

European Union Declarations of Conformity

Avaya Inc. declares that the equipment specified in this document bearing the "CE" (Conformité Europeénne) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (2004/108/EC) and Low Voltage Directive (2006/95/EC).

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

European Union Battery Directive

Avaya Inc. supports European Union Battery Directive 2006/66/EC. Certain Avaya Inc. products contain lithium batteries. These batteries are not customer or field replaceable parts. Do not disassemble. Batteries may pose a hazard if mishandled.

Japan

The power cord set included in the shipment or associated with the product is meant to be used with the said product only. Do not use the cord set for any other purpose. Any non-recommended usage could lead to hazardous incidents like fire disaster, electric shock, and faulty operation.

本製品に同梱または付属している電源コードセットは、本製品専用です。本製品以外の製品ならびに他の用途で使用しないでください。火災、感電、故障の原因となります。

If this is a Class A device:

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。

If this is a Class B device:

This is a Class B product based on the standard of the Voluntary Control Council for Interference from Information Technology Equipment (VCCI). If this is used near a radio or television receiver in a domestic environment, it may cause radio interference. Install and use the equipment according to the instruction manual.

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 Ｂ 情報技術装置です。この装置は，家庭環境で使用することを目的としていますが，この装置がラジオやテレビジョン受信機に近接して使用されると，受信障害を引き起こすことがあります。取扱説明書に従って正しい取り扱いをして下さい。

Downloading documents

For the most current versions of documentation, see the Avaya Support Web site:

http://www.avaya.com/support

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Overview

This Hardware Description and Reference provides information on hardware currently supported by Avaya Aura® Communication Manager. Use this book to find information on Avaya servers and Media Gateways, as well as circuit packs, media modules, telephones, and other hardware used with Communication Manager.

This book contains information on the following hardware:

- Linux-based servers
- Other servers
- Media gateways integrated gateways and trunk gateways
- Circuit packs, channel service units, and power supplies
- Media modules
- Telephones and speakerphones
- UPS units
- Ethernet switches

For each hardware component, an overview and description is provided. Where appropriate, information is also provided on models, configurations, components, LEDs, specifications, supported and related hardware, reliability and survivability, and high-level capacities.

Communication Manager

Communication Manager is an open, scalable, highly reliable, and secure telephony application. Communication Manager provides call processing solutions for large and small customer environments. It also provides user and system-management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking. The standard edition Avaya Aura® Communication Manager also uses H.248 for gateway control.

Communication Manager offers various features in the following categories:

- Call center
- Telephony
- Localization
- Collaboration
- Mobility

- Messaging
- Telecommuting
- System management
- Reliability
- Security, privacy, and safety
- Hospitality
- Attendant features
- Networking
- Intelligent call routing
- Application programming interfaces

For more information about these solutions, see *Avaya Aura® Communication Manager Overview*, 03-300468.

Communication Manager runs on the following Linux-based servers:

- S8300D Server
- S8510 Server
- S8800 Server

Avaya servers and Media Gateways

Avaya servers and Media Gateways provide smart ways to rethink networking. They add top-tier scalability and reliability, while supporting critical applications in a distributed, yet secure, multivendor environment. To provide businesses with maximum flexibility, the server and gateway components in Communication Manager follow a modular mix-and-match approach. A wide range of custom configurations can be deployed to meet a broad spectrum of business needs:

- From a single location, upgrading to a converged IP network for 200 employees
- To a complex multinational converged network that is capable of supporting 10,000-plus voice and data users

Note:

Some of Avaya servers and gateways were tested against extreme physical and environmental requirements such as shock, vibration, and Electromagnetic Interference (EMI). These tests were performed by the United States Navy for server and gateway use on their ships. The Navy uses specialized racks and reinforcements although no physical changes have been made to the servers and gateways themselves. For more information on design and implementation of a similar ruggedized solution, contact the Avaya Custom Engineering Group.

Servers

Avaya line of servers provides a robust application platform based on industry-standard operating systems. This platform supports distributed IP networking and centralized call processing across multiprotocol networks. These servers are available as an integrated solution with other servers or independently.

Avaya servers have the following features and benefits:

- Redundant, survivable call processing and media processing supports crucial business continuity.
- Standard-based computing supports Linux operating system.
- Distributed survivable IP networking supports campus, global multisite, and branch environments.

Media gateways

Avaya Media Gateways connect to an Avaya server, either directly or indirectly through other Media Gateways. Media gateways are the stackable and modular hardware elements of your communication system. They deliver connectivity to a variety of endpoint and trunk types to provide data, voice, fax, video, and messaging capabilities on your network. The connections between Media Gateways that allow the passage of these media types are called the bearer networks. The connections between the server and the Media Gateways for call control signaling are called the control networks.

Avaya Media Gateways support both bearer and signaling traffic that is routed between packet-switched and circuit-switched networks. Avaya Media Gateways provide a variety of flexible deployment options. These options include 100% Internet Protocol (IP) environments and blended environments such as IP and Time Division Multiplexing (TDM).

Avaya Media Gateways have the following benefits:

- Interoperable with standard-based data networks
- Stackable, modular, and configurable component solutions

- Provision of redundant equipment and capabilities
- Provision of distributed networking
- Compatible with cabinets in traditional Avaya systems

Categories of Media Gateways

There are two primary categories of Media Gateways:

- Those that use media modules to connect to endpoints and trunks. These H.248 Media Gateways are usually used at branch and smaller locations and include:
  - G700 Media Gateway
  - G450 Media Gateway
  - G430 Media Gateway
  - G350 Media Gateway
  - G250 Media Gateway
- Those that use circuit packs to connect to endpoints and trunks. The Avaya G650 Media Gateway is usually used at central and large locations.

G650 Media Gateway

The G650 Media Gateway provides card slots for up to 14 TN-type circuit packs, redundant, hot-swappable power supplies, and AC or DC power. The backplane can support 14 circuit packs and 2 power supplies and provides monitoring of system fans, power supplies, and temperature. Up to five G650 Media Gateways can be mounted in an EIA-310 standard 19-inch (48 cm) rack.

Common architectural aspects of the 650 Media Gateway

A Media Gateway consists of the following architectural components:

- **TDM bus.** The TDM bus has 512 time slots. The TDM bus runs internally throughout each Media Gateway and terminates on each end. The TDM bus consists of two 8-bit parallel buses, bus A and bus B. Bus A and bus B carry circuit-switched digitized voice and data signals. Bus A and Bus B can also carry control signals to all port circuits and between port circuits and the server. The port circuits place digitized voice signals and data signals on a TDM bus. Bus A and bus B are usually active simultaneously. However, only one bus is active at any one time for control signaling.

- **Packet bus.** The packet bus runs internally throughout each Media Gateway and terminates on each end. The packet bus carries logical links and control messages from the server. The links and messages are carried 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. These circuits include, for example, IPSI, expansion interface, and IP Media Resource 320 circuit packs, control D-channels, and remote management terminals.
- **Port circuits.** The port circuits form analog or digital interfaces between the Media Gateway and external trunks and linking devices. These linking devices provide links between the gateway and external trunk and the TDM bus and the 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.
- **Service circuits.** For traditional servers, S8300D Servers, S8510 Servers, and S8800 Servers, service circuits provide tone production and detection, call classification, recorded announcements, and speech synthesis. The embedded 8300 Server uses built-in service circuits in the G250, G350, G430, G450, and G700 Media Gateways.

Port networks

The architectures for the S8510 Server and S8800 Server use an entity called a port network (PN). A PN uses combinations of Media Gateways to provide physical ports and interfaces for handling calls. A port network can be one of the following:

- One G650 Media Gateway
- A stack of G650 Media Gateways that is connected with a TDM bus cable and shares connections with the server or port circuit packs

Note:

The G700, G450, G430, G350, and G250 Media Gateways are controlled by a Communication Manager server through H.248 and are not considered port networks. However, they may reside within a configuration including port networks.

For information on port network connectivity, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

System Management

Avaya Integrated Management

Avaya Integrated Management offers a comprehensive set of Web-based network management solutions and system management solutions that support the Avaya converged voice solutions. Integrated Management combines individual applications into the following offers:

- Administration Tools
- System Management
- Enterprise Network Management

For detailed information on Avaya Integrated Management suite, see Products and Services on Avaya Web site.

System Management Interface

Using the System Management Interface (SMI), you can perform the server administration tasks, such as:

- Viewing current alarms
- Maintaining the server including:
  - Checking the servers status
  - Busying out and releasing busy out the server
  - Shutting down the server
- Executing security commands to:
  - enable and disable the modem
  - start and stop the FTP server
  - view the license
- Accessing SNMP to configure trap destinations and to stop and start the master agent
- Accessing the server to acquire configuration information

The SMI contains an extensive Help system that describes each Web screen and the procedures associated with the screen.

Avaya communications devices

Avaya provides new mobility opportunities and devices that are innovative and standards based. Avaya offers a wide selection 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. The highlights of the portfolio include:

- Avaya Softconsole: A software attendant console that brings the features and functionality of a high-end attendant console to your converged network.
- Avaya IP Softphone: A collection of computer telephony integration (CTI) applications. With this you can control telephone calls, both incoming and outgoing, directly from your personal computer (PC).
- Avaya IP Agent: An advanced PC-based application. With IP agent you can access the contact center agent functionality of Communication Manager over the private network or public network. You can also use IP Agent to handle calls associated with an IP telephone or Callmaster VI telephone.
- Avaya 4630 Screenphone: A full-color touch-screen phone with Web access.
- Avaya IP Wireless Phones: Provides access to conferencing and corporate directories.
- Avaya Conference Phone: Provides full-duplex technology to enhance sound quality.
- Avaya IP Deskphone: Designed for various business communication needs.

Avaya IP communication devices are supported without special power requirements.

For more information about communication devices, see the www.avaya.com/support.

Adjuncts

The following list contains some of the adjuncts from Avaya that the Avaya servers support:

- Call Detail Recording (CDR) when a terminal server is used
- Avaya Aura® Messaging
- Modular Messaging system
- Avaya Basic Call Management System (BCMS)
- Avaya Call Management System, which is available in three packages:
  - Avaya Call Center Basic
  - Avaya Call Center Deluxe
  - Avaya Call Center Elite

- Avaya Interactive Response system
- Call Accounting Systems supported with the United States of a terminal server.

Linux-based servers

Avaya S8300 Server

An S8300 Server is an Intel Celeron-based processor that runs on the Linux operating system. It resides in one of the following Media Gateways: G250, G350, G430, G450, G650, or G700.

Detailed description

S8300D Server

The S8300D Server is supported by Communication Manager Release 5.2 and later.

An S8300D Server is an Intel Core 2 Duo U5700 processor that runs the Linux operating system. The S8300D Server resides in Slot V1 of a Media Gateway and includes:

- 80-GB hard disk
- 4-GB DRAM (with one 1 GB DIMM)
- 8-GB Internal Solid State Drive (SSD)
- Three USB ports and a 10/100 Base-T port
    - One USB port supports a readable DVD/CD-ROM drive, which is used for system installations and upgrades.
    - Another USB port can be used for a USB modem.
    - A third USB port can be used for a Compact Flash drive.
- One services port
- One internal Compact Flash drive which is used as the primary reboot device
- Modem support for alarming

Software

In addition to Communication Manager software for applications, the S8300D Server runs the following software:

- A Web server that is used for:
    - Backing up and restoring customer data
    - Viewing current alarms
    - Server maintenance, including busy out, shutdown, and status of an S8300D Server

- Security commands to enable and disable the modem
- Security commands to start and stop the FTP server
- Security commands to view the software license
- SNMP access to configure trap destinations and to stop and start the master agent
- Configuration information about the S8300D Server
- Upgrading access to the S8300D Server

- Maintenance software
- Linux Red Hat operating system
- Trivial File Transfer Protocol (TFTP) server
- Secure HTTP server for IP phone file downloads
- H.248 Media Gateway Signaling Protocol
- Control messages tunneled over H.323 Signaling Protocol

Configurations

The Avaya S8300D Server has the following basic hardware configurations:

- S8300D Server/G700 Media Gateway configuration
- S8300D Server/G430 Media Gateway configuration
- S8300D Server/G450 Media Gateway configuration
- S8300D Server/G350 Media Gateway configuration
- S8300D Server/G250 Media Gateway configuration

An Avaya S8300D Server with a Media Gateway and the gateways media modules converge voice and data into one infrastructure. The S8300D Server is an Intel Celeron-based processor that resides in the Media Gateway. The server has the same dimensions and shape as a media module.

In addition, an S8300D Server can serve as a survivable remote server (Local Survivable Server). See S8300D Server in a Survivable Remote Server configuration.

Note:

The S8300D Server must be version D to operate Communication Manager Release 6.0 software.

S8300D Server/G700 Media Gateway configuration

The S8300D Server resides in Slot V1 of a G700 Media Gateway.

A G700 Media Gateway, which is architecturally based on the Avaya C360 switches, contains VoIP resources and modular interface connectivity. The media modules provide analog, digital, T1/E1, BRI, and additional VoIP capabilities.

An S8300D Server with a G700 Media Gateway (Figure 1) has the following components:

- Survivability
- Avaya G700 Media Gateway, which can include:
  - Media modules
  - X330 WAN Access routing module
- S8300D Server in a Survivable Remote Server configuration
- System Management

For more details on the G700 Media Gateway, see Avaya G700 Media Gateway. For more details on the S8300D Server, see Survivability.

S8300D Server in a G700 Media Gateway

Figure 1: S8300D Server in a G700 Media Gateway

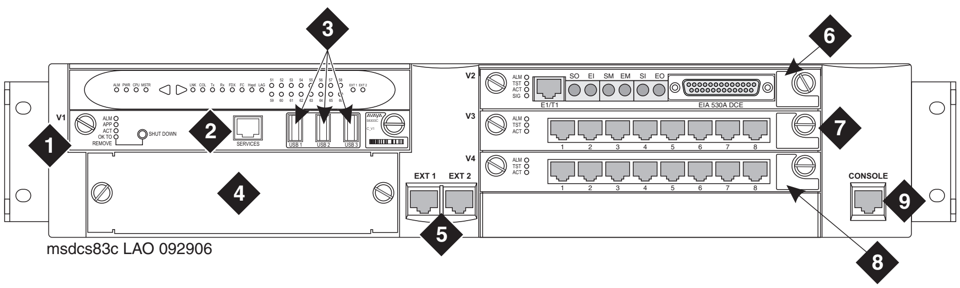


Figure notes:

Number	Description
1.	S8300D Server in Slot V1
2.	Services port
3.	USB ports
4.	Slot
5.	Dual 10/100 Base-T Ethernet switch ports
6.	Media module, Slot V2
7.	Media module, Slot V3
8.	Media module, Slot V4
9.	Console connection for on-site administration

S8300D Server/G450 Media Gateway configuration

The G450 Media Gateway features a VoIP engine, an optional WAN router, and Ethernet LAN connectivity. G450 provides full support for Avaya IP and digital telephones, as well as analog devices such as modems, fax machines, and telephones. The media modules in a G450 Media Gateway provide analog, digital, T1/E1, BRI, and additional VoIP capabilities.

G450 supports the S8300D from version S8300B onward. The S8300D runs Communication Manager to provide call control services to the G450. G450 is compatible with Communication Manager starting with version 5.0.

The S8300D server resides in slot V1. See G450 physical description for the configuration of an S8300D Server in a G450 Media Gateway.

S8300D Server/G430 Media Gateway configuration

The G430 Media Gateway features a VoIP engine, an optional WAN router, and Ethernet LAN connectivity. The G430 provides full support for Avaya IP and digital telephones, as well as analog devices such as modems, fax machines, and telephones.

The G430 supports the S8300D from version S8300C onwards. The S8300D runs Communication Manager to provide call control services to the G430. G430 is compatible with Avaya Communication Manager from version 5.2.

The S8300D server resides in slot V1. See G430 physical description for the configuration of an S8300D Server in a G430 Media Gateway.

S8300D Server/G350 Media Gateway configuration

The G350 Media Gateway features a VoIP engine and WAN router and provides full support for legacy digital and analog telephones. Like the G700 Media Gateway, the media modules in a G350 Media Gateway provide analog, digital, T1/E1, BRI, and additional VoIP capabilities. The following figure shows an S8300D Server and media modules in a G350 Media Gateway.

An S8300D Server and a G350 Media Gateway configuration has the following components:

- Survivability
- Avaya G350 Media Gateway, which includes Related hardware
- Communication Manager
- System Management

For more information on the G350 Media Gateway, see Avaya G350 Media Gateway. For more information on the S8300D Server, see Survivability.

S8300D Server in a G350 Media Gateway

Figure 2: S8300D Server in a G350 Media Gateway

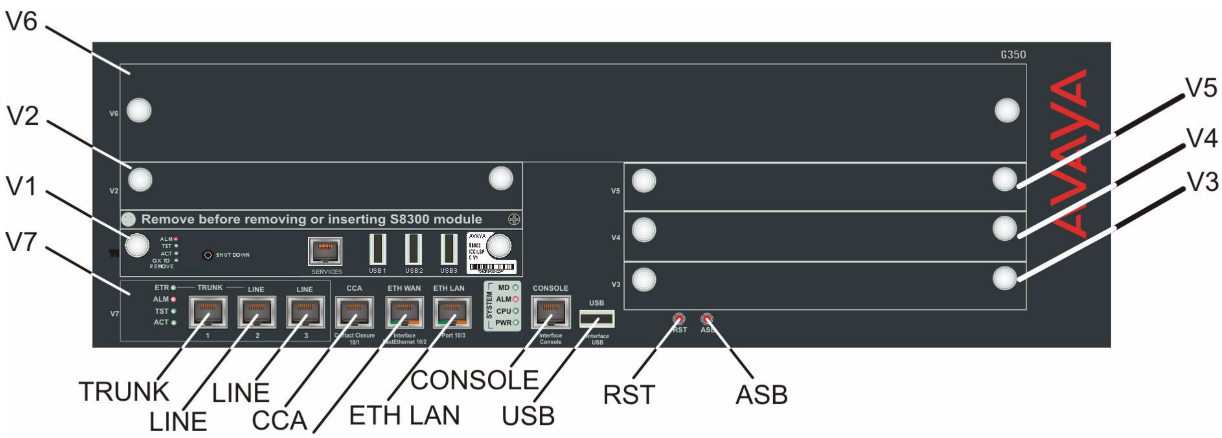


Figure notes:

Port	Description
TRK	An analog trunk port. Part of an integrated analog media module.
LINE 1, LINE 2	Analog telephone ports of the integrated analog media module. An analog relay between TRK and LINE 1 provides Emergency Transfer Relay (ETR) feature.
CCA	RJ-45 port for ACS (308) contact closure adjunct box.
WAN 1	RJ-45 10/100 Base TX Ethernet port.
LAN 1	RJ-45 Ethernet LAN switch port.
CON	Console port for direct connection of CLI console. RJ-45s connector.
USB	USB port for remote access modem.
RST	Reset button. Resets chassis configuration.
ASB	Alternate Software Bank button. Reboots the G350 with the software image in the alternate bank.

S8300D Server/G250 Media Gateway configuration

The G250 Media Gateway features a VoIP engine, WAN router, and Power over Ethernet switch. The G250 Media Gateway is available in four models; analog, BRI, DCP, and 1. The G250 Media Gateway supports analog and IP telephones. The G250 Media Gateway has built-in media modules. The G250 Media Gateway has two slots available for optional modules; slot V1 houses an optional S8300D Server and slot V2 houses one of two optional WAN media modules.

An S8300D Server and a G250 Media Gateway configuration has the following components:

- Survivability
- Media gateways and integrated gateways
- Communication Manager
- System Management

For more information on the G250 Media Gateway, see Media gateways and integrated gateways. For information on the S8300D Server, see Survivability.

S8300D Server in a G250 Media Gateway (analog version)

Figure 3: S8300D Server in a G250 Media Gateway (analog version)

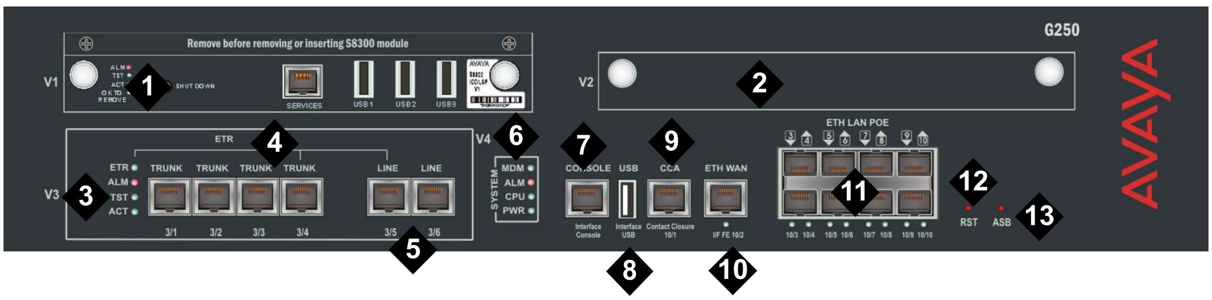


Figure notes:

1. **V1: S8300D/Survivable Remote Server Slot**
2. **V2: WAN Media Module Slot**
3. **Analog port LEDs**
4. **Analog trunks**
5. **Analog line ports**
6. **System LEDs**
7. **Console port**
8. **USB port**
9. **Contact Closure (CCA) port**
10. **Ethernet WAN (ETH WAN) port**
11. **PoE LAN (ETH LAN PoE) ports**
12. **Reset (RST) button**
13. **Alternate Software Bank (ASB) button**

Components

For a list of S8300D components used in each S8300D configuration, see Configurations.

UPS or power backup

For the S8300D Server, any of the available UPS units can instantly supply power during a power outage.

RAM disk

RAM disk is a portion of memory used as a disk partition. In the event of a hard disk failure, the S8300D Server uses only RAM disk to provide call processing for up to 72 hours. Administration and backups are prohibited. Also, Communication Manager Messaging is unavailable when operating in RAM disk mode, so secondary call coverage points for users should be administered even with RAM disk enabled.

Note:

The S8300D server does not support RAM Disk.

Related hardware and adjuncts

Communication Manager Messaging

Note:

The IA770 INTUITY AUDIX messaging is called Communication Manager Messaging, starting with Communication Manager Release 5.2 and later releases.

Communication Manager Messaging is an optional voice mail system used with an S8300D Server. Communication Manager Messaging is a software-only version of messaging that uses a QSIG-MWI H.323 virtual trunk for communication between Communication Manager and Communication Manager Messaging software. This version is available on the G700, G450, G430, G350, and G250 Media Gateway configurations. Without the need for additional hardware, Communication Manager Messaging software processes touchtone signals, converts messages to the G.711 format, and converts text to speech.

Note:

The Communication Manager Messaging application is included with Communication Manager Release 6.0 with many of Communication Manager templates.

The Communication Manager Messaging system can be a solution for one location in a stand-alone S8300D configuration. The system can also be networked with other voice mail systems using TCP/IP and Avaya Message Networking.

Communication Manager Messaging uses many resources of the S8300D Server and the Media Gateway where it resides. The following list outlines the shared resources of S8300 used by Communication Manager Messaging system:

- Hardware for data storage and retrieval
- TFTP server for:
  - Downloading and updating the license file for feature activation

  - Backing up and restoring data over a LAN or a WAN, including translations and messages
  - Updating and upgrading software
- IP address for administration access
- General Alarm Manager for alarm display
- Web interface to start and stop the system

The Communication Manager Messaging system also shares the same switch-tone parameters established for the S8300D Server. The S8300D Server handles switch tones on behalf of Communication Manager Messaging system and passes on the control information to Communication Manager Messaging system using QSIG signaling.

Call center

An S8300D Server provides an excellent solution for a small call center by offering the following call center capabilities:

- A maximum of 16 ASAI links
- Announcement software

G430 supports call center features with large announcement storage, including optional compact flash, large voice trunk capacity, and 16 announcement ports for announcement record and playback.

Printers

The S8300D Server is connected to the customer LAN. Therefore, you can send print requests to any printer within the LAN and IP region of the S8300D Server.

A system printer is supported when a terminal server is used. In this case, the printer is connected to an adjunct PC such as a CDR system, CMS, or Call Accounting System.

A journal printer is supported when a terminal server is used.

Survivability

S8300D Server in a Survivable Remote Server configuration

An S8300D Server in a Survivable Remote Server (Local Survivable Processor) configuration uses the S8300D hardware component and a software license to activate a standby feature. This software allows the Survivable Remote Server with a Media Gateway to be a survivable call-processing server for remote locations and branch locations.

The branch locations can have the following servers as their primary controller:

- S8300D
- S8510
- S8800

An S8300D Server and the Survivable Remote Server cannot reside in the same Media Gateway.

If for any reason communication between a Media Gateway and its primary controller stops, a Survivable Remote Server activates. This "failover" from the primary controller to the Survivable Remote Server is an automatic process without human intervention. The Survivable Remote Server assumes control of any IP telephone provided that telephone has the Survivable Remote Server in its list of controllers.

The Survivable Remote Server can continue to support calls as the primary controller for 30 days. The Survivable Remote Server is in "license-error" mode when it is supporting calls. After 30 days in license-error mode, the Survivable Remote Server administration is blocked and display telephones show **License Error** in their display windows. However, even after 30 days, telephone operations can continue.

Automatic fallback to primary controller

Based on administration of Communication Manager, the G250/G350/G430/G450/G700 Survivable Remote Server can return control of the G250/G350/G430/G450/G700 Media Gateway to the primary controller (server) automatically when the connection is restored between the Media Gateway and the primary controller. By returning control of the Media Gateways to the primary controller automatically, Communication Manager software easily and quickly eliminates the fragmentation between remote gateways in the network created by LAN/ WAN communication failures with the primary controller.

The Media Gateway preserves stable calls when control changes from the Survivable Remote Server to the primary controller. Stable calls are calls that are carrying active two-way or multiparty conversations. Other calls such as those that are on hold are not preserved.

Note:

The fall-back from the Survivable Remote Server to the primary controller may also be manual using a reset on the Survivable Remote Server. This reset breaks the communication between the Survivable Remote Server and each registered endpoint. This break causes the endpoints to register with the primary controller. However, most active calls are preserved.

Number of Survivable Remote Servers supported

The number of Survivable Remote Servers that a configuration can support depends on the controlling server. An S8510 and S8800 Server can support up to 250 Survivable Remote Servers. An S8300D Server can support up to 50 Survivable Remote Servers.

Translations

An automatic process copies translation changes when customers make changes on the primary controller to each Survivable Remote Server.

Hardware Requirement

The hardware for the S8300D Server as primary controller is identical to the hardware for the S8300D Server as Survivable Remote Server. The difference between the two configurations is entirely in software.

IP addressing of the primary controller, the Survivable Remote Server, and IP telephones

A Survivable Remote Server is administered with a different IP address than the IP address of the primary controller. In addition, IP telephones obtain their own IP address from a DHCP server. The DHCP server also sends a list of controllers, Survivable Remote Servers, and their associated IP addresses. The IP telephone then registers with the controller corresponding 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 can be administered for telephones on the DHCP server.

High-level capacities

The S8300D Server supports:

- 900 ports by a combination of trunks and stations
  - 450 IP stations, 450 non-IP stations, or a combination of 450 IP and non-IP stations
  - 450 trunks
- 50 G250/G350/G430/G450/G650/G700 Media Gateways

Table 1: High-level capabilities

Capability	S8300D Server
Call processing feature set	Communication Manager 3.0
Maximum number of stations	450 (IP or TDM)
Maximum number of trunks	450
Reliability options	Single server
Port-network connectivity	Not applicable

Table 1: High-level capabilities

Capability	S8300D Server
Supported Media Gateways	G250, G350, G430, G450, G650, G700
Maximum number of supported gateways	50 (supported by one S8300D Server)
Survivability options	G250, G350, G430, G450, G650, and G700 with S8300D Survivable Remote Server
Number of Survivable Remote Servers in one configuration	Maximum of 50 when supported by an S8300D. Maximum of 250 when supported by an S8510 or S8800 Servers
Port networks	Not applicable

For more detailed system capacity information, see *Avaya Aura® Communication Manager System Capacities Table,* 03-300511.

Avaya S8510 Server

The Avaya S8510 Server is a single server that runs on the Linux operating System and features Communication Manager. The S8510 Server supports Internet Protocol (IP), Session Initiation Protocol (SIP), and traditional endpoints. This tri-level support enables new technology and eases migration from legacy Avaya systems. The S8510 Server is a perfect solution for mid-sized customers, with growth of up to 3200 ports.

Detailed description

An S8510 Server configuration includes the following:

- S8510 Server
- Media gateways for main locations which individually or as stacks connect to port networks through Avaya G650 Media Gateway, which is always sold with new systems.

  Note:

  If used as a survivable remote server, the Avaya G700 Media Gateway, the Avaya G430 Media Gateway, the Avaya G450 Media Gateway, the Avaya G350 Media Gateway, and the Media gateways and integrated gateways are supported through the processor ethernet interface.

  Note:

  Media Gateway types cannot be mixed within the same port network (PN).
- TN2312BP IP server interface, which provides control signaling between the server and the port networks (PNs). In an IP-connect configuration, each PN must contain one TN2312BP circuit pack.
- TN2302AP IP media processor or TN2602AP IP Media Resource 320, which provides TDM-to-IP conversions of audio signals. At least one of these circuit packs is required in each IP-connected PN.
- Communication Manager
- System Management

The S8510 Server supports secure HTTP server for IP phone file downloads.

The S8510 Server supports IP port network connections: single control network (IP-PNC).

S8510 Server

The Avaya S8510 Server uses the Linux operating system and supports several Avaya software applications. It is generally used in single server mode, but in some circumstances can be duplexed. The Avaya S8510 server is targeted for the mid-sized customer.

The major architectural and functional differences between the S8510 and the S8500C are:

- The S8510 hardware platform is a Dell multi-core CPU platform.
- The S8510 does not support the RAMDisk feature but instead supports the hardware version of RAID (Redundant Array of Independent Disks) Level 1 industry standard feature with Dual Hard Disk Drives (HDD).

The S8510 server hardware system comes equipped with the hardware version of the RAID Level 1 feature. This feature employs the disk mirroring method which creates a set of data on two or more disks. A general RAID 1 mirrored pair contains two disks which increases the reliability of the system. Each of the disks is independent of the other and contains a complete copy of the data.

The default S8510 server configuration has a single Power Supply. However, the S8510 server supports the redundant power supply configuration, and a customer can choose to order an extra Power Supply.

For examples of the front and back of the S8510 Server, see Figure 4: S8510 Server (front view) on page 42 and Figure 5: S8510 Server (back view) on page 43.

Front view of S8510 Server

Figure 4: S8510 Server (front view)

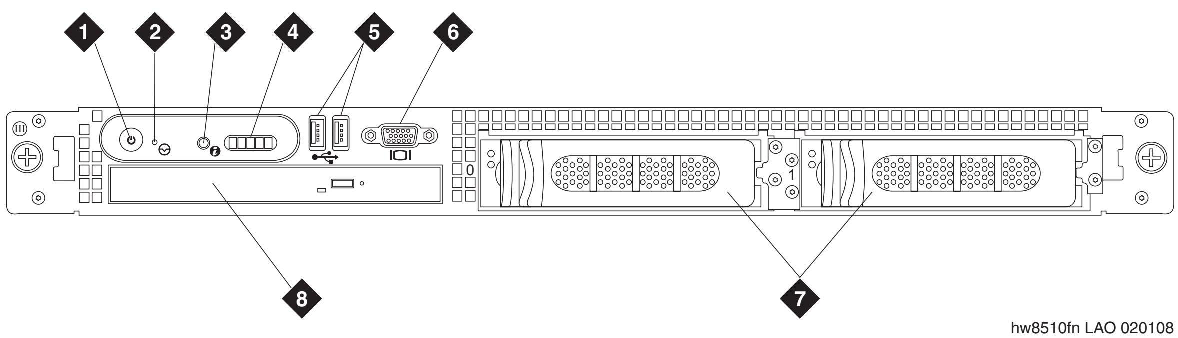


Figure notes:

1. **Power-on LED**
2. **NMI button (not used)**
3. **System ID button**
4. **LCD display**
5. **USB ports**
6. **Video connector (not used)**
7. **Hard disk drives**
8. **Optical DVD/CD drive**

Back view of S8510 Server

Figure 5: S8510 Server (back view)

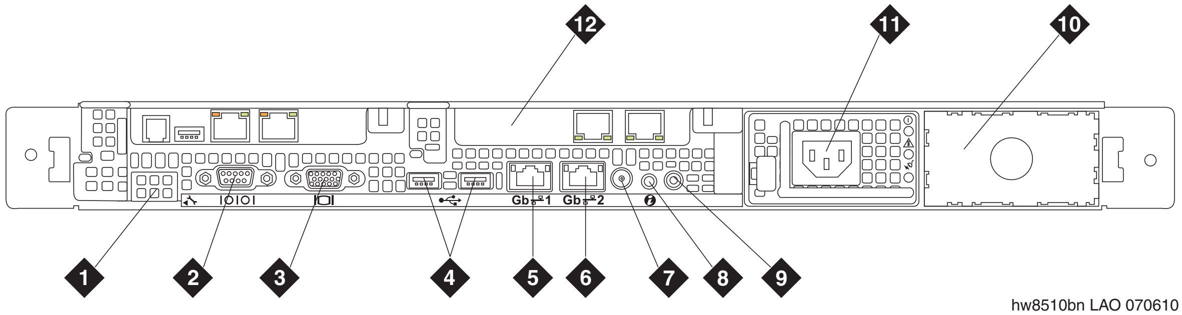


Figure notes:

1. Remote access controller (not used)
2. Serial connector (not used)
3. Video connector (not used)
4. USB ports (not used)
5. GB-1 (Eth0)
6. Services port - GB-2 (Eth1)
7. Services status indicator connector
8. System ID button
9. System Status LED
10. Bay for optional redundant power supply
11. Power supply
12. Dual NIC

S8510 LED Indicators

Figure 6 shows the drive status/activity LEDs.

Figure 7 show the status LEDs on the back of the S8510.

Table 2 describes the LED indicator conditions for power, power supply, AC line status, and drive status.

Figure 6: S8510 Server (drive status/activity)

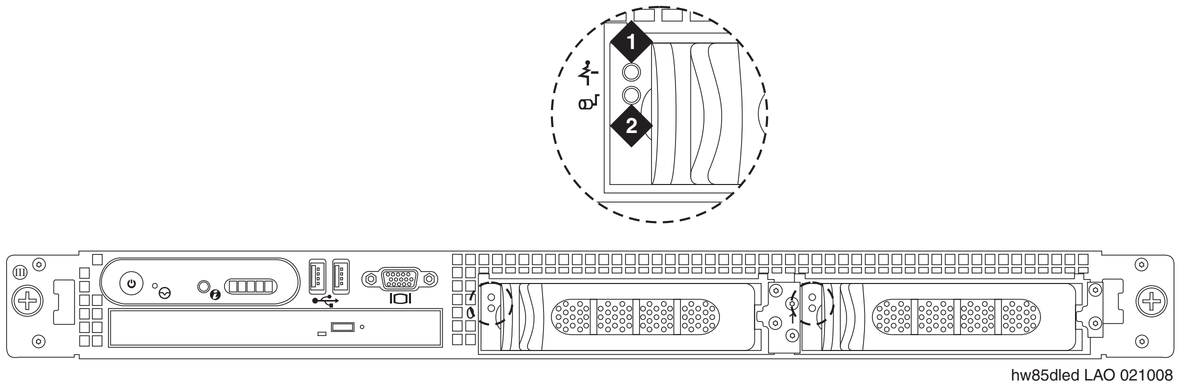


Figure notes:

1. **Drive status**
2. **Drive activity**

Figure 7: S8510 Power Supply/AC Line LEDs

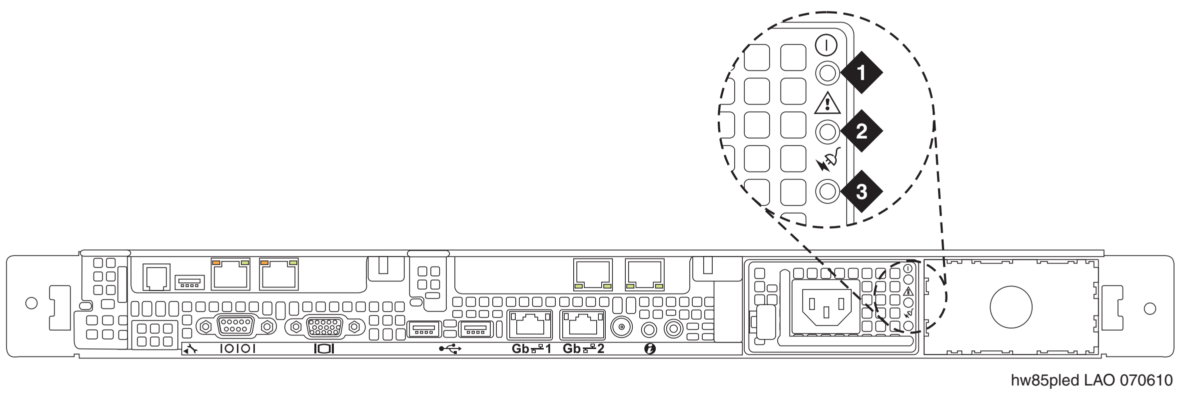


Figure notes:

1. **Power supply status**
2. **Power supply fault**
3. **AC line status**

Table 2: S8510 LED indicator conditions *1 of 2*

LED	Indicator/Pattern	Function/Condition
Power Button	On	System has power and is operational
	Off	System has no power

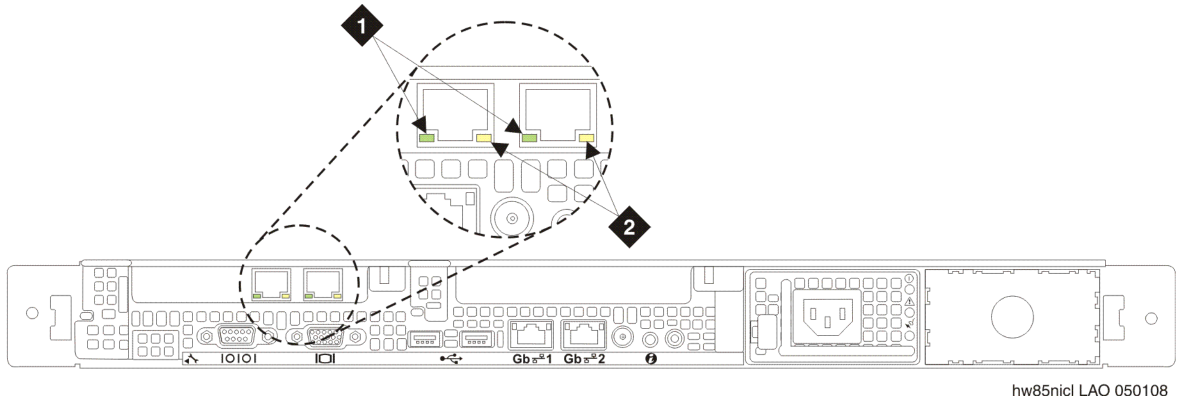
Table 2: S8510 LED indicator conditions *2 of 2*

LED	Indicator/Pattern	Function/Condition
Power Supply Status	Green	Power supply is operational
Power Supply Fault	Amber	There is a problem with the power supply
AC line status	Green	Power supply is connected to a valid AC power source
Drive status	Off	Drive ready for insertion or removal
	Steady green	Drive online
	Blinks green, amber, off	Drive predicted failure
	Blinks amber 4 times per second	Drive failed
	Blinks green 2 times per second	Identify drive/preparing for removal
	Blinks green slowly	Drive rebuilding
	Blinks green 3 seconds, amber 3 seconds, off 6 seconds	Rebuild aborted
Drive activity	Blinks green	Drive has activity

NIC Indicator Codes

Each NIC on the back panel has an indicator which provides information on network activity and link status. Figure 8 describes the NIC indicator codes.

Figure 8: NIC Indicator Codes



1	Network activity (TX/RX)
2	Connection rate • Off: 10BaseT active link • Green: 100BaseT active link • Amber: 1000BaseT active link

Configurations

S8510 Server components

The S8510 comes with the following components:

- One Quad Core Intel® Xeon® E5410 Processor 5000 Sequence.
- A minimum of 2 GB (2 x 1 GB) of 667 MHz, fully buffered DIMMs (FBD), upgradable to a maximum of 32 GB by installing combinations of 1-GB, 2-GB, or 4-GB memory modules in the eight memory module sockets on the system board.

Servers dedicated to Release 6.0 Communication Manager must have 8 GB of physical memory.

- Support for two 3.5-inch, internal hot-pluggable SATA (7200 rpm) hard drives.
- An optional slimline DVD-ROM/ CD-RW drive.
- One hot-pluggable, 670-W power supply with an option of installing a second power supply in a 1 + 1 redundant (optional) configuration.

- Four fan modules, each comprises two dual-rotor fans, for a total of eight cooling fans.

The system board includes the following features:

- One dual network interface card (NIC)
- Two integrated gigabit Ethernet NICs capable of supporting 10-mbps, 100-mbps, and 1000-mbps data rates.
- Four USB 2.0-compliant connectors. Two on the front support an optional mouse and keyboard.
- An integrated VGA-compatible video subsystem with an ATI ES1000, 33-MHz PCI video controller.
- Back-panel connectors include serial, video, two USB connectors, and two NIC connectors.
- Front-panel connectors include a video and two USB connectors.
- Front-panel 1x5 LCD for system ID and error messaging.

RAID

The S8510 supports the hardware version of RAID (Redundant Array of Independent Disks) Level 1 industry standard feature with Dual Hard Disk Drives (HDD). A general RAID 1 mirrored pair contains two disks which increases the reliability of the system.

Each disk is independent of each other and contains a complete copy of the data. The primary HDD is mirrored onto the secondary HDD. If either drive fails, the other continues to function without service interruption. If both disks fail, the server is out of service.

Replacement of a failed HDD requires no service interruption.

CAUTION:

The firmware associates HDDs with the server. HDDs can be moved as a pair to another server, but in order to do this, you must import the other servers configuration in the RAID-BIOS. This must be done at boot time on a keyboard and a monitor, which have to be connected directly to the server. HDD slot numbering must be maintained when moving drives between systems.

The **`raid_status`** bash command displays the server RAID controller status.

There is a new web interface *RAID Status* option under **Diagnostics** on the Main. *RAID HDD Status* is displayed as part of the *Server->Status Summary*.

RAID HDD Status is displayed as part of the **`server`** bash command.

Value	Description
1	1 HDD operational

Value	Description
2	2 HDD operational
-1	1 or 2 failed HDD; error in acquiring HDD information

S8510 Server specifications

The following table outlines the specifications of the S8510 Server.

Type	Description
Processor	
Processor type	1 Quad-core Intel Xeon E5410 processor 5000 sequence
Expansion bus	
Bus type	PCIe
Memory	
Architecture	PC2-4100 667 MHz fully buffered DIMMs with ECC protection
Memory module sockets	8 240-pin
Memory module capacities	1 GB, 2 GB, 4 GB
Min/Max RAM	1GB/32 GB
Drives	
SATA hard drives	2 3.5 in., internal hot-pluggable with backplane support
Optical drive	1 DVD-ROM/CD-RW combination
Connectors (front)	
USB	2 4-pin, USB 2.0 compliant
Video	15-pin VGA
Connectors (back)	
NIC	2 RJ-45
Serial	9-pin, DTE, 16550-compatible
USB	2 4-pin, USB 2.0 compliant
Video	15-pin VGA

Type	Description
AC power supply Wattage Voltage Heat dissipation Maximum inrush current	 670 W 90-264 VAC, autoranging, 47-63 Hz, 10.0 A (at 90 VAC) 2697 BTU/h (maximum) Under typical line conditions and over the entire system ambient operating range, the inrush current may reach 55 A per power supply for 10 ms or less.
System battery	CR 2032 3.0-V lithium ion coin cell
Dimensions (HxWxD/Us) Weight	1.7 x 19 x 30 in. (4.3 x 48.3 x 7.26 cm)/1 U 39 lb (17.7 kg)

DANGER:

The Avaya S8510 Server contains lithium batteries. These batteries are not customer field replaceable parts. Do not disassemble. Batteries may pose a hazard if mishandled.

Environmental requirements

Type	Description
Temperature Operating Storage	 10° to 35°C (50° to 95°F) –40° to 65°C (–40° to 149°F)
Relative humidity Operating Storage	 8% to 85% (non-condensing) with a maximum humidity gradation of 10% per hour 5% to 95% (non-condensing)
Maximum vibration Operating Storage	 0.25 G at 3–200 Hz for 15 min 0.5 G at 3–200 Hz for 15 min

Type	Description
Maximum shock	
Operating	One shock pulse in the positive z axis (one pulse on each side of the system) of 41 G for up to 2 ms
Storage	Six consecutively executed shock pulses in the positive and negative x, y, and z axes (one pulse on each side of the system) of 71 G for up to 2 ms
Altitude	
Operating	–16 to 3048 m (–50 to 10,000 ft)
Storage	–16 to 10,600 m (–50 to 35,000 ft)

Related hardware

Table 3: S8510 Server Hardware Specifications

Specification	S8510 Server
Processor	2 GHz Quad Core
SDRAM Memory	8GB
Hard Drives	250GB SATA RAID Level 1 Duplicated Redundant disk drive
On-Board NICs	Dual Gigabit Ethernet 10/100/100
Optical Drive	CD RW/DVD read-only drive
NIC Card	1 X Dual NIC PCI-e Card (100/1000)
Diagnostic Indicators	LEDs and Alphanumeric Display
Form Factor (HxWxD) Weight	1U High: 1.7" x 19" x 30" 39lb (17.7kg)
Fans	Redundant Fans
Power Supply	Dual (Optional) Redundant Hot Pluggable

Survivability for the S8510 server

Recovery capability is embedded in Communication Manager software that resides on the S8510 Server. Thus, the servers can use the following recovery options:

- Servers, port networks, and gateways supported by S8510 Survivable Core Server
- S8300 Server in a Survivable Remote mode

S8510 Server as a Survivable Core Server

A Communication Manager configuration may use the S8510 Server as a Survivable Core Server (Enterprise Survivable Server). The Survivable Core Server option provides survivability to a configuration by allowing backup servers to be placed in various locations in the customer's network. A Survivable Core Server assumes call processing control of all or part of the configuration in case the main server, either S8510 or S8800 Server, fails or network connections to the main server fail.

A main server may have up to 63 Survivable Core Server available to provide backup service. The placement of the Survivable Core Server or Survivable Core Servers in the configuration is typically targeted at ensuring that port networks that are configured in different segments of the customer's LAN/WAN can receive service even when LAN/WAN connections are lost.

Once the communication failure to the main server has been corrected, control of call processing may be returned from the Survivable Core Server to the main server either manually port network by port network or automatically for all port networks at once.

Note:

In the transition of control from the main server to a Survivable Core Server, all calls are dropped while the Media Gateways carrying the calls reset to connect to the Survivable Core Server.

Servers, port networks, and gateways supported by S8510 Survivable Core Server

The S8510 Server may serve as the Survivable Core Server for either an S8510 or an S8800 main server. If the main server is a S8510 Server, any and all Survivable Core Servers in the configuration must also be S8510 Server. The S8510 Server Survivable Core Server can maintain the duplication when it takes call processing control from the main server. To support duplication, an S8510 Server Survivable Core Server must also contain a dual NIC card. Note that when the S8510 Server is used as a Survivable Core Server for the S8800 main, the S8510 Server has the same capacities as the S8800 main.

Note:

A Survivable Core Server may support a G250, G350, G430, G450, G650, or G700 Media Gateway through the C-LAN connection of the Survivable Core Server-connected port network.

S8300 Server in a Survivable Remote mode

The S8300 Survivable Remote Server is located in the G700/G450/G430/G350/G250 Media Gateway and provides survivability when the S8510 Server is inaccessible. Each S8510 Server can have up to 250 Survivable Remote Servers. The S8300D Survivable Remote Server can support up to 50 H.248 Media Gateways. The Survivable Remote Server has a copy of the S8510 Server customer translations.

Power outages

In most cases, an Avaya solution can 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.

S8510 port connections

Use standard CAT5 cables with RJ45 connectors on each end to connect to the various ports. If the S8510 Server has only one port network, connect that port network through the dual NIC. Figure 9: S8510 Server connectivity guide on page 52 shows typical connectivity for the S8510 Server.

Figure 9: S8510 Server connectivity guide

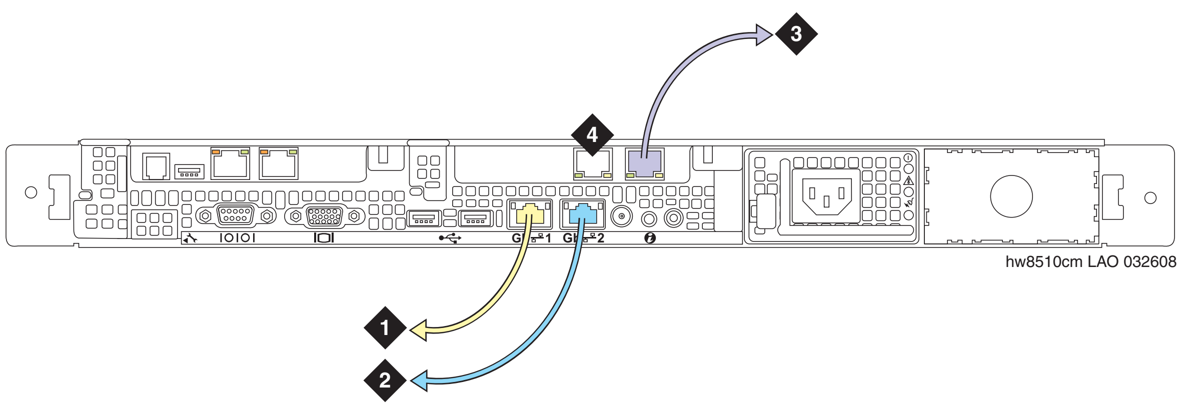


Figure notes:

1. **Eth0: To the customer network if the control network is nondedicated. Or, to the control-network Ethernet switch if the control network is dedicated (straight-through CAT5 cable)**
2. **Eth1: To the Services laptop computer (crossover CAT5 cable)**
3. **Eth3: To the customer network if the control network is dedicated (straight-through CAT5 cable)**
4. **Eth4: Not used**

S8510 BIOS Upgrade Feature

The **BIOS Upgrade** feature is provided in the **Server Upgrades** section on the System Management Interface under.

Field Replaceable Units

Field Replaceable Unit
S8510 Server
Hard Disk Drive(s) Redundant Disk Hot Pluggable
Power Supply(s) (Optional) Redundant Power Supply Hot Pluggable
Memory
Dual NIC

High-level capacities

Table 4: High-level capacities

Capability	S8510 Server
Call processing feature set	Communication Manager 3.1. The S8510 Server is supported by Communication Manager releases 5.1 and later.
Reliability options	Single server control and duplicated bearer. Note that Communication Manager 5.1 is the minimum load for the S8510.
Port network connectivity	IP-connect and direct-connect
Supported port network Media Gateways	Voice bearer over IP: G650
	1 of 2

Table 4: High-level capacities *(continued)*

Capability	S8510 Server
Maximum number of supported Media Gateways for branch offices	250 (includes G700, G650, G450, G430, G350, and G250 Media Gateways in any combination)
Maximum locations	64 port networks, plus up to 250 G700/G650/G450/G430/G350/G250 Media Gateways
Survivability options	G250, G350, G430, G450, G650,and G700 Media Gateways with S8300D Survivable Remote Server S8510 Survivable Core Server or Survivable Remote Server
Number of Survivable Remote Servers in one configuration	Maximum 250 Survivable Remote Servers
Number of Survivable Core Servers in one configuration	Maximum 63 Survivable Core Servers
Port networks per IPSI	One with IP-connect port networks. Three with direct-connect port networks.
	2 of 2

For more detailed system capacity information, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

In addition to voice calls, the S8510 Server, through Communication Manager and the United States of an appropriate media processor (T2302AP or TN2602AP), supports transport of the following messages:

- Fax, Teletypewriter device (TTY), and modem calls using pass-through mode
- Fax, V.32 modem, and TTY calls using proprietary relay mode

SECURITY ALERT:
Faxes sent to nonAvaya endpoints cannot be encrypted.

Note:
V.32 modem relay is needed primarily for secure SCIP telephones formerly known as Future Narrowband Digital Terminal (FNBDT) telephones and STE BRI telephones.

- T.38 Fax over the Internet, including endpoints connected to nonAvaya systems
- 64kbps clear channel transport in support of firmware downloads, BRI secure telephones, and data appliances

Note:

The path between endpoints for modem tone transmissions must use Avaya telecommunications and networking equipment.

For more information, see TN2302AP IP media processor or TN2602AP IP Media Resource 320.

For more information, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

Related Documents

LED Descriptions for Avaya Aura® Communication Manager Hardware Components, 03-602804

Installing the Avaya S8510 Server Family and Its Components, 03-602918

Server Availability Management Processor: Avaya S8510 Server, 03-602923

Avaya S8800 Servers

The Avaya S8800 Server supports several Avaya software applications. The server is available in a 1U model or 2U model and with various hardware components. The server model and specific hardware components in your server depend on the requirements of the software application that will run on the server.

Communication Manager supports the 1U model of the S8800 Server. While installing Communication Manager in simplex mode on the S8800 Server, you only use one S8800 Server, whereas installing Communication Manager in duplex mode requires you to have two S8800 Servers.

S8800 Server

Front view of S8800 Server

Figure 10: S8800 Server (front view)

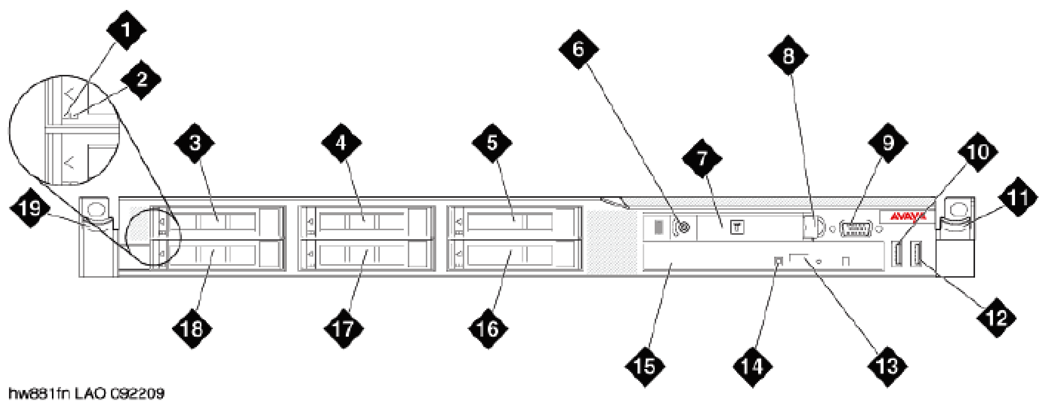


Figure notes:

Number	Description of Device
1.	Hard disk drive activity LED (green)
2.	Hard disk drive status LED (amber)
3.	Drive bay 0
4.	Drive bay 2
5.	Drive bay 4 (unused for Communication Manager)
6.	Power control button and LED

Number	Description of Device
7.	Operator information panel **Note:** The operator information panel is shown in the pushed in position.
8.	Operator information panel release latch
9.	Video connector
10.	USB connector 1
11.	Rack release latch
12.	USB connector 2
13.	DVD eject button
14.	DVD drive activity LED
15.	DVD drive
16.	Drive bay 5 (Unused for Communication Manager)
17.	Drive bay 3 (Unused for Communication Manager)
18.	Drive bay 1
19.	Rack release latch

Back view of S8800 Server

Figure 11: S8800 Server (back view)

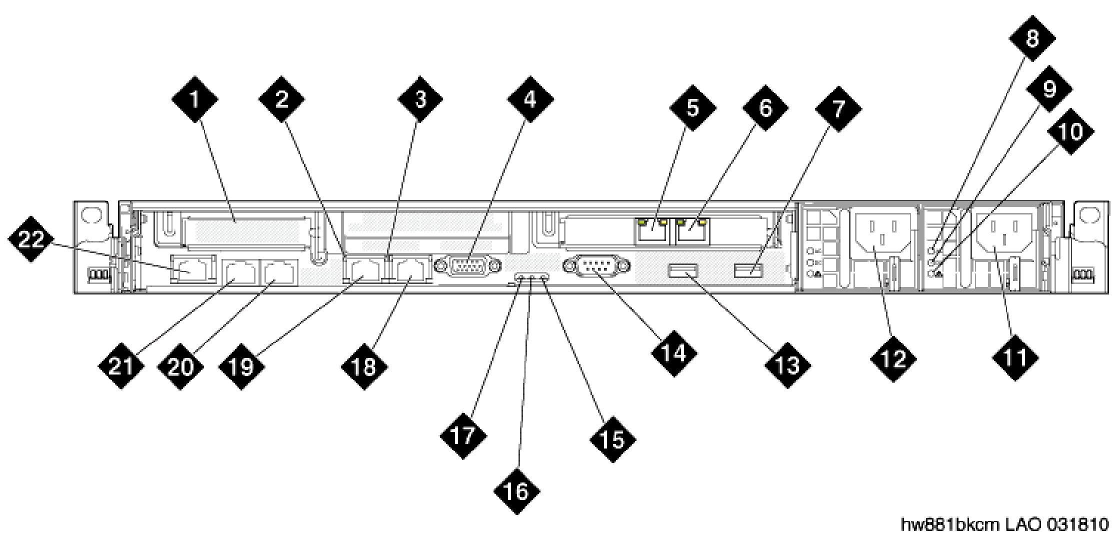


Figure notes:

Number	Description of Device
1.	PCIe slot 1 (unused)
2.	Ethernet activity LED
3.	Ethernet link LED
4.	Video connector
5.	Ethernet connector 6 (eth5) (unused)
6.	Ethernet connector 5 (eth4) (second NIC for bonding to eth0 for redundancy)
7.	USB connector 4
8.	AC power LED (green)
9.	DC power LED (green)
10.	Power supply error LED (amber)
11.	Power supply 2 (optional redundant power supply)
12.	Power supply 1 (primary power supply)
13.	USB connector 3
14.	Serial connector
15.	System error LED (amber)
16.	System locator LED (blue)
17.	Power LED (green)
18.	Ethernet connector 2 (eth 1) (Services port)
19.	Ethernet connector 1 (eth 0) (Corporate LAN / processor Ethernet)
20.	Ethernet connectors 4 (eth 3) (Duplication link if configuration is duplicated server)
21.	Ethernet connector 3 (eth 2) (not used)
22.	System management Ethernet connector (IMM)

Note:

Hardware label for Ethernet ports on the server is called Ethernet connectors. Communication Manager software refers to Ethernet ports as eth.

S8800 Server components

Component	Minimum Specification	Upgrade options based on product requirements
Microprocessor	One Intel E5520 quad core, 2.26 GHZ processor	No additional options
Memory	S8510 Server: 8 GB of DRAM (Install an additional 1 GB DRAM DIMMS to achieve 8GB of DRAM) S8800 Server: 12GB DRAM S8330D: 8GB DRAM	No additional options
Media drive	DVD-R/W SATA slimline	No additional options
Hard disk drive expansion bays	Six 2.5-inch hot-swap SAS hard disk drive bays	No additional options

Component	Minimum Specification	Upgrade options based on product requirements
Hard disk drive	S8510 Server: 2 drives each 3.5” 250GB SATA drives configured as a RAID 1 array yielding 232 GB of usable disk space or 3 drives each 2.5” 146 GA SAS drives configured as a RAID 5 array yielding 272 GB of usable disk space S8800 Server: 3 each 2.5” 146 GB SAS hard drives configured in a RAID 5 array yielding 272 GB of usable disk space **Note:** For an upgrade from Communication Manager Release 5.2.1 on an S8800 server to Communication Manager Release 6.0 on an S8800 Server, you need to install four additional 2 GB DRAM DIMMS. Thus there is a total of 12 GB of DRAM installed in the server. When installing additional DRAM, the DIMMs are to be installed in accordance with the instructions found in the “Sequence for populating DIMM connectors” section on page 20 of the Maintaining the *Avaya S8800 Server for Avaya Aura® Communication Manager*, 03-603446. S8300D Server: Single 250 GB drive	No additional options
RAID controllers	ServeRAID-MR10i RAID SAS adapter that provides RAID level 1 or 5. Includes 256 MB cache module and battery for write cache	No additional options
PCI expansion slots	Two PCI Express x16 Gen 2 slots: • Slot 1 supports a low profile DUAL NIC card (half height, half-length cards) • Slot 2 supports full height, half-length cards	No additional options
Hot-swap fans	Six	No additional options

Component	Minimum Specification	Upgrade options based on product requirements
Power supply	One 675W, 12V AC power supply **Note:** The second (redundant) power supply is required if you desire to have Communication Manager system meet or exceed an availability requirement of 99.99%.	Redundant 675W, 12V AC power supply
Video controller	Integrated Matrox G200 (two analog ports, one front and one back, that can be connected at the same time) The maximum video resolution is 1280 x 1024 at 75 Hz. • SVGA compatible video controller • DDR2 250 MHz SDRAM video memory controller • Avocent Digital Video Compression • Video memory is not expandable	No additional options

S8800 Server specifications

The following table outlines the specifications of the S8800 Server.

Type	Description
Dimensions	Height: 43 mm (1.69-inches, 1U) Depth: 711 mm (28-inches) Width: 440 mm (17.3-inches)
Weight	Maximum weight: 15.4 kg (34 lb) when fully configured.
Heat output	Approximate heat output: • Minimum configuration: 662 Btu per hour (194 watts) • Maximum configuration: 1400 Btu per hour (400 watts) Heat output varies depending on the number and type of optional features that are installed and the power management optional features that are in use.

Type	Description
Acoustic noise emissions	Declared sound power, operating: 6.1 bel The sound levels were measured in controlled acoustical environments according to the procedures specified by the American National Standards Institute (ANSI) S12.10 and ISO 7779 and are reported in accordance with ISO 9296. Actual sound-pressure levels in a given location might exceed the average values stated because of room reflections and other nearby noise sources. The declared sound power levels indicate an upper limit, below which a large number of computers will operate.

Electrical input requirements	• Sine-wave input (47–63 Hz) required
	• Input voltage low range:
	  - Minimum: 100 V AC
	  - Maximum: 127 V AC
	• Input voltage high range:
	  - Minimum: 200 V AC
	  - Maximum: 240 V AC
	• Input kilovolt-amperes (kVA), approximately:
	  - Minimum: 0.194 kVA
	  - Maximum: 0.700 kVA

Front connectors	• Two USB • Video
Back connectors	• Six Ethernet (RJ— 45 connectors). • Serial • Two USB • Video • Systems management Ethernet (IMM)

Environmental requirements

Server status	Air temperature	Maximum Altitude	Relative humidity
Server on	• 10º C to 35º C (50º F to 95º F) at altitude of up to 914.4 m (3,000 feet) • 10º C to 32º C (50 to 90º F) at altitude of 914.4 m to 2,133 m (3,000 to 7,000 feet)	2,133 m (7,000 feet)	8% to 80%
Server off	10°C to 43°C (50.0°F to 109.4°F)	2,133 m (7,000 feet)	8% to 80%

Media gateways and integrated gateways

Avaya G250 Media Gateway

The Avaya G250 Media Gateway is an H.248 Media Gateway managed by a server that has Communication Manager software installed on it. The Communication Manager Media Gateways form part of Avaya solution for extending communication capabilities from the headquarters of an organization to all collaborative branch locations. The Communication Manager Branch Gateways help you provide the same high quality services to all organization members, regardless of their location.

Note:
The G250 Media Gateway is no longer being sold.

Detailed description

The G250 Media Gateway is a high-performance converged telephony and networking device that is located in small branch locations, providing all infrastructure needs in one box such as telephone exchange and data networking. G250 is designed for used in a two to 12 user environment, aimed at small branch offices with two to eight stations. The G250 Media Gateway features a VoIP engine, WAN router, and Power over Ethernet LAN connectivity. The G250 supports legacy IP and analog telephones. In addition, the G250-DCP model supports DCP telephones.

The G250 Media Gateway integrates seamlessly with the following Avaya servers:

- S8800
- S8510
- S8300D

These servers run the Communication Manager call processing software to provide the same top quality telephony services to the small branch office as to the headquarters of the organization. The server can be located at the headquarters and serve the G250 remotely.

The G250 Media Gateway can optionally house an internal Avaya S8300D Server as a survivable remote server (Enhanced Local Survivability) or as the main server for stand-alone deployment. As a survivable remote server, the S8300D is capable of providing full Communication Manager functionality in the event that the connection with the server is lost.

As an alternative to the survivable remote server, G250 can be configured for Standard Local Survivability (SLS). See Survivability on page 72.

The G250 supports the connection of personal computers, LAN switches, IP phones, analog telephones, and trunks through fixed analog and PoE ports on the chassis. A media module slot supports either of two WAN media modules for connection to a WAN.

There are four models of G250 with various port combinations for support of analog, BRI, or T1/E1 trunks or DCP telephones, as described in Models on page 66.

For more information on the features of the G250 Media Gateway, see *Overview of the Avaya G250 and G350 Media Gateways*, 03-300435.

Models

The G250 Media Gateway is available in the following models:

- Analog model (G250-Analog). The G250-Analog includes four analog trunk ports, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.
- BRI model (G250-BRI). The G250-BRI includes two ISDN BRI trunk ports, one analog trunk port, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.
- DCP model (G250-DCP). The G250-DCP provides 12 DCP (Digital Communications Protocol) ports, as well as four analog trunk ports, two analog line ports, a Fast Ethernet WAN port, and two LAN ports.

  CAUTION:

  The DCP ports on the G250-DCP are intended for in-building use only. Phone lines connected to DCP ports are not to be routed out of the building. Failure to comply with this restriction could cause harm to personnel or equipment.
- DS-1 model (G250-DS-1). The G250-DS-1 provides a T1/E1 and a PRI trunk port that supports fractional T1/E1 and PRI. The G250-DS-1 also includes one analog trunk port, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.

Components

Chassis

Figure 12 shows the G250-Analog Media Gateway chassis. Figure 13 shows the G250-BRI Media Gateway chassis. Figure 14 shows the G250-DCP Media Gateway chassis. Figure 15 shows the G250-DS-1 Media Gateway chassis.

Figure 12: The Avaya G250-Analog Media Gateway Chassis

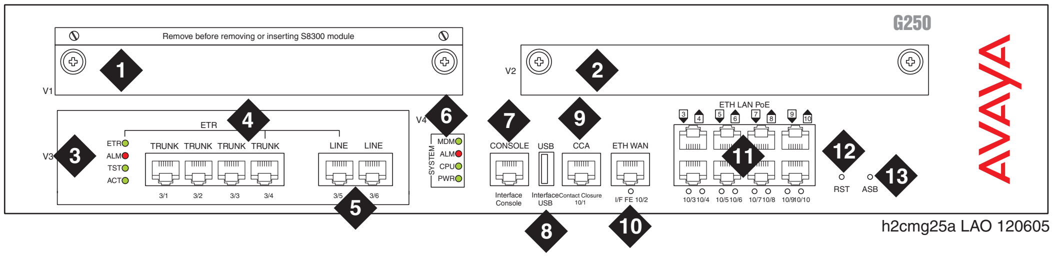


Figure notes:

1. V1: ICC/Survivable Remote Server Slot
2. V2: WAN Media Module Slot
3. Analog port LEDs
4. Analog trunks
5. Analog line ports
6. System LEDs
7. Console port
8. USB port
9. Contact Closure (CCA) port
10. Ethernet WAN (ETH WAN) port
11. PoE LAN (ETH LAN PoE) ports
12. Reset (RST) button
13. Alternate Software Bank (ASB) button

Figure 13: The Avaya G250-BRI Media Gateway Chassis

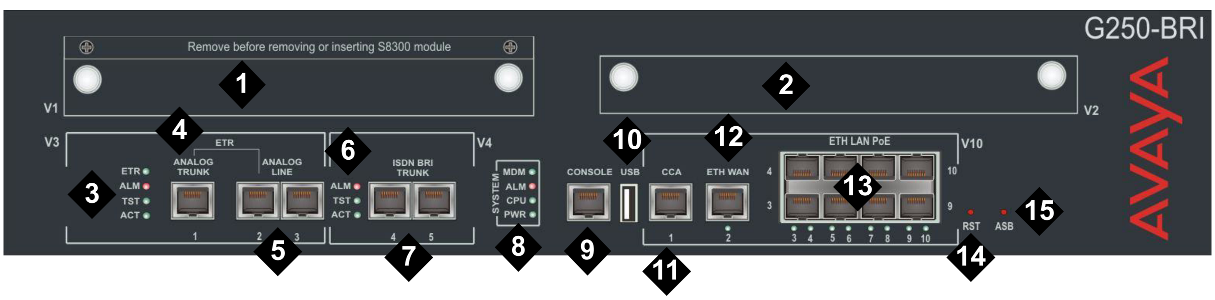


Figure notes:

1. V1: ICC/Survivable Remote Server Slot
2. V2: WAN Media Module Slot
3. Analog port LEDs
4. Analog trunk
5. Analog line ports
6. ISDN BRI LEDs
7. ISDN BRI trunks
8. System LEDs
9. Console port
10. USB port
11. Contact Closure (CCA) port
12. Ethernet WAN (ETH WAN) port
13. PoE LAN (ETH LAN PoE) ports
14. Reset (RST) button
15. Alternate Software Bank (ASB) button

Figure 14: The Avaya G250-DCP Media Gateway Chassis

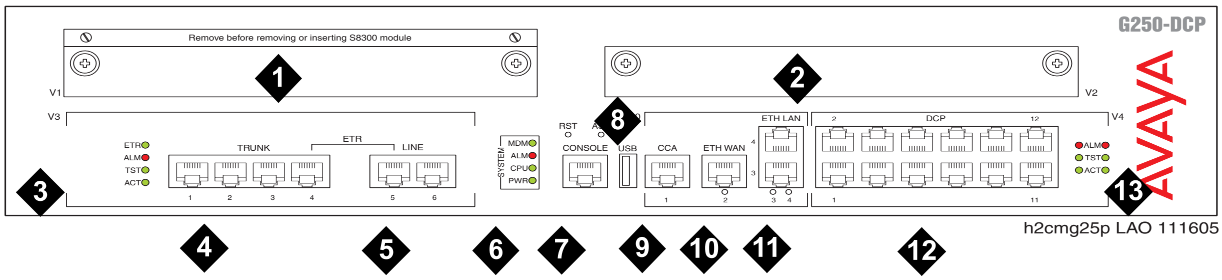


Figure notes:

1. V1: ICC/Survivable Remote Server Slot
2. V2: WAN Media Module Slot
3. Analog port LEDs
4. Analog trunks
5. Analog line ports
6. System LEDs
7. Console port
8. USB port
9. Contact Closure (CCA) port
10. Ethernet WAN (ETH WAN) port
11. ETH LAN ports
12. DCP ports
13. DCP port LEDs

Figure 15: The Avaya G250-DS-1 Media Gateway Chassis

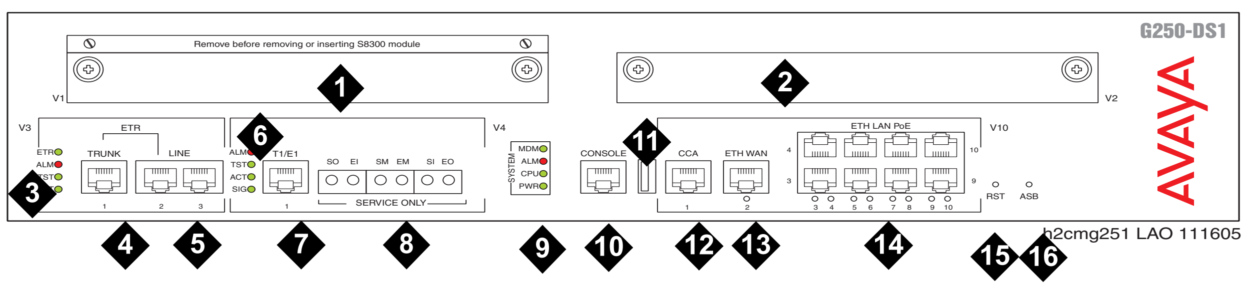


Figure notes:

1. V1: ICC/Survivable Remote Server Slot
2. V2: WAN Media Module Slot
3. Analog port LEDs
4. Analog trunk
5. Analog line ports
6. T1/E1/PRI trunk interface LEDs
7. T1/E1 interface
8. Service
9. System LEDs
10. Console port
11. USB port
12. Contact Closure (CCA) port
13. Ethernet WAN (ETH WAN) port
14. PoE LAN (ETH LAN PoE) ports
15. Reset (RST) button
16. Alternate Software Bank (ASB) button

Ports and buttons

Table 5 describes the functions of the fixed ports and buttons on the front panel of G250.

Table 5: Fixed ports and buttons on the G250 front panel

Port	Description
TRUNK	Four analog trunk ports (G250-Analog Media Gateway, G250-DCP Media Gateway) or one analog trunk port (G250-BRI Media Gateway, G250-DS-1 Media Gateway). These fixed trunk ports support loop-start, DIOD (for Japan only) trunks and caller ID detection.
LINE	Two analog telephone ports. An analog relay provides Emergency Transfer Relay (ETR) feature. For the G250-Analog and G250-DCP, the relay is between TRUNK port 3/4 and LINE port 3/5. For the G250-BRI and G250-DS-1, the relay is between TRUNK port 3/1 and LINE port 3/2. Also used for incoming analog DID trunks with either wink-start or immediate start. The G250 integrated analog line ports support three ringer loads, which is the ringer equivalency number (REN), for the following loop lengths: - 20,000 feet (6096 meters) over 0.65 mm (.025 in.) wire (22 AWG) - 16,000 feet (4877 meters) over 0.5 mm (.02 in.) wire (24 AWG) - 10,000 feet (3048 meters) over 0.4 mm (.016 in.) wire (26 AWG) At .1 or less REN ringer loads, the supported loop length is 20,000 feet (6096 meters) at 22, 24, and 26 AWG.
ISDN BRI TRUNK (G250-BRI Media Gateway)	Two 4 wire S/T ISDN BRI (Basic Rate Interface) 2B+D access ports with RJ-45 jacks. Each port interfaces to the central office at the ISDN T reference point. The ISDN BRI trunk ports do not support: • BRI stations • Combining both B channels together to form a 128-kbps channel
CONSOLE	Console RS-232 interface port for direct connection of CLI console. RJ-45 connector.
	1 of 2

Table 5: Fixed ports and buttons on the G250 front panel *(continued)*

Port	Description
USB	USB port. Supports the connection of • USB flash drive • USB externally powered hub • The Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem.
CCA	RJ-45 port for ACS (308) contact closure adjunct box.
ETH WAN	RJ-45 10/100 Base TX Ethernet port for connection to a cable or DSL broadband modem/router.
ETH LAN POE (G250-Analog, G250-BRI, and G250-DS-1)	Eight Power over Ethernet (PoE) LAN ports with 80 watts (aggregated for all ports) for connecting IP phones or any Ethernet devices, such as PCs.
RST	Reset button. Resets chassis configuration.
ASB	Alternate Software Bank button. Reboots the G250 with the software image in the alternate bank.
DCP (G250-DCP)	12 DCP ports. These DCP ports are intended for in-building use only. The G250-DCP ports support a loop length as follows: • 5500 feet (1676 meters) over 0.65 mm (.025 in.) wire (22 AWG) • 3500 feet (1067 meters) over 0.5 mm (.02 in.) wire (24 AWG) • 2200 (671 meters) over 0.4 mm (.016 in.) wire (26 AWG)
T1/E1 port (G250-DS-1)	For T1, this port is capable of supporting inband signalling across all 24 channels (supports a maximum bandwidth of 1.536 mbps). For E1, this port is capable of supporting R2MFC signalling across all 30 channels (supports a maximum bandwidth of 1.92 mbps).
PRI ports (G250-DS-1)	The PRI ports are capable of supporting PRI signalling for 23 or 30 bearer channels. NFAS signalling is not supported.
	2 of 2

Specifications

Dimensions and site requirements

The following table lists the physical dimensions and site requirements of the G250 Media Gateway.

Description	Value
Height	2U (3.5 in., 88.9 mm)
Width	17.3 in. (440 mm)
Depth	13.375 in. (340 mm)
Weight of empty chassis	22 lb (10 kg)
Ambient working temperature	32 °F-104 °F (0 °C-40 °C)
Operation altitude	up to 10,000 ft. (3,048 m)
Front clearance	2.5 in. (6.4 cm) (minimum clearance for ventilation)
Rear clearance	2.5 in. (6.4 cm) (minimum clearance for ventilation)
Humidity	10-90% non-condensing
Power rating	90-240 V~, 47-63 Hz
BTU	823 BTU/h
Max current	2.4A

Power cord specifications

The specifications for power cords suitable for use with G250 are as follows:

For North America: The cordset must be UL Listed/CSA Certified, 16 AWG, 3-conductor (3rd wire ground), type SJT. One end terminates to an IEC 60320, sheet C13 type connector rated 10A, 250V. The other end terminates to either a NEMA 5-15P attachment plug for nominal 125V applications or a NEMA 6-15P attachment plug for nominal 250V applications.

For Outside North America: The cord must be VDE Certified or Harmonized (HAR), rated 250V, 3- conductor (3rd wire ground), 1.0 mm2 minimum conductor size. The cord terminates at one end to a VDE Certified/CE Marked IEC 60320, sheet C13 type connector rated 10A, 250V. The other end of the cord terminates to a 3-conductor grounding type attachment plug rated at a minimum of 10A, 250V. The configuration is specific for the region/country in which the cord is used. The attachment plug must bear the safety agency certifications mark(s) for the region/ country of installation.

Note that the G250 Media Gateway relies on two ground connections. These connections are a mains plug with an earth contact and a permanent Supplementary Ground Conductor. Because of unreliable earthing concerns in Finland, Norway, and Sweden, the G250 Media Gateway must be installed in a Restricted Access Location (RAL). Only trained service personnel or customers can access the RAL. Trained service personnel are aware of the reasons for the restricted access and any safety precautions that must be taken. In these scenarios, the personnel or customers must use a lock and key or other means of security when they access the G250 Media Gateway.

Related hardware

The media modules reside in the G250 Media Gateway and interact with the motherboard and backplane.

Note:

For stand-alone mode, the S8300D Server is inserted into slot 1. See Avaya S8300 Server.

There are two WAN media modules:

- MM340 T1/E1 data WAN – For information, see MM340 E1/T1 data WAN Media Module.
- MM342 USP data WAN – For information, see MM342 USP data WAN Media Module.

Survivability

The G250 Media Gateway supports Standard Local Survivability (SLS) that is a configurable software module. With SLS, a local G250 can provide a core set of Media Gateway Controller functions when no link is available to the server, a Survivable Remote Server, or a Survivable Core Server (Enterprise Survivable Server). SLS is configured on a system-wide basis using the new Provisioning and Installation Manager (PIM), or an individual G250 using the command line interface (CLI).

SLS is supported as follows on the G250 Media Gateway:

- G250-Analog: SLS supported for all analog interfaces, IP phone, and IP Softphone
- G250-BRI: SLS supported for all analog interfaces, ISDN BRI trunk interfaces, IP phone, and IP Softphone
- G250-DCP: SLS supported for all analog and DCP interfaces, IP phone, IP Softphone, and DCP phone
- G250-DS-1: SLS supported for all analog interfaces, ISDN PRI trunk interfaces, non-ISDN digital DS-1 trunk interfaces, IP phone, and IP Softphone

High-level capacities

The following table outlines the capacities of various G250 services.

Note:

Some capacities might change. For the most up-to-date list, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Table 6: G250 capacities

Description	Capacity	Comments
Media Gateway Limits		
Maximum number of G250 Media Gateways controlled by an external S8510 or S8800 Server	250	This number also applies if a combination of Avaya G250/G350/G430/G450/G650/G700 Media Gateways are controlled by the same external S8xxx server.
Maximum number of G250 Media Gateways controlled by an external S8300D Server housed in a G430/G450/G650/ G700 Media Gateway	50	
Servers registered as Media Gateway Controllers. If an MGC becomes unavailable, the G250 uses the next MGC on the list.	4	The built-in SLS module can be considered a fifth MGC, although its functionality is more limited than that of a full scale server.
Media module slots	2	One S8300D Server slot (V1) for insertion of S8300D only. One WAN media module slot (V2) for insertion of a WAN media module only.
Maximum number of WAN media modules	1	Always in slot V2.
Maximum number of voice media modules	0	
Maximum total number of telephones supported by the G250	14	
Maximum number of IP phones	12	Limited by the number of VoIP resources used and the calling patterns (VoIP to VoIP conferencing, VoIP to non-VoIP etc.)
		1 of 2

Table 6: G250 capacities *(continued)*

Description	Capacity	Comments
Maximum number of analog phones	2	
Maximum number of DCP phones	12	G250-DCP only. None in the other G250 models.
Maximum number of BRI endpoints	0	
1 facilities	1 T1/E1	G250 DS-1 only. None in the other G250 models.
Maximum number of all trunks of any type	4 (5 on G250-BRI, 10 on G250-DS-1)	
Maximum number of G250 analog trunks	4 (Analog, G250-DCP) 1 (G250-BRI, G250-DS-1)	All ports are fixed.
Maximum number of BRI trunks	2 (G-250 BRI only)	Four voice channels, two D-channels.
Maximum number of E1/T1 voice trunks	1	G250-DS-1 only. None in the other G250 models.
Simultaneous two-way conversations from IP phone to legacy telephone or trunk	10 (Analog, G250-BRI) 16 (G250-DCP, G250-DS-1)	True for all codecs (G.711, G.729a, G.726, G.723), and all encryption combinations.
Miscellaneous		
Fax capacity	4	Simultaneous fax transmissions using VoIP resources
Touch-tone recognition (TTR)	8	Receivers
Tone Generation	As much as necessary for all TDM calls	
Announcements (VAL)	6 playback channels for playing announcements. 15 minutes for either G711-quality stored announcements or music-on-hold.	
		2 of 2

Avaya G350 Media Gateway

The Avaya G350 Media Gateway is an Avaya solution for extending high-quality communication capabilities from the headquarters of an organization to all collaborative branch locations.

Note:

The G350 Media Gateway is no longer being sold.

Detailed description

The G350 Media Gateway is a high-performance converged telephony and networking device located at a small branch and providing all infrastructure needs in one box including telephone exchange and data networking. The G350 Media Gateway is designed for use in an eight to 72 user environment, aimed at branch offices with 16 to 40 stations. The G350 Media Gateway features a VoIP engine, WAN router, and Power over Ethernet LAN switch, and provides full support for IP, DCP, and analog telephones.

The G350 Media Gateway integrates seamlessly with the following Avaya servers:

- S8800
- S8510
- S8300D

These servers run Communication Manager call processing software to provide the same top quality telephony services to the small branch office as to the headquarters of the organization. The server can be located at the headquarters and serve the G350 Media Gateway remotely.

The G350 Media Gateway can optionally house an internal Avaya S8300D Server as a survivable remote server or as the main server for stand-alone deployment. As an alternative to the survivable remote server, the G350 Media Gateway can instead be configured for Standard Local Survivability (SLS). See Survivability on page 82.

In addition to advanced and comprehensive telephony services, the G350 Media Gateway provides full data networking services, precluding the need for a WAN router or LAN switch.

The G350 Media Gateway is a modular device, adaptable to support different combinations of endpoint devices. Pluggable media modules provide interfaces for different types of telephones and trunks. A combination is selected to suit the needs of the branch.

A LAN media module with Ethernet ports that are PoE standard compliant provides support for IP telephones as well as all other types of data devices. A range of telephony modules provides full support for legacy equipment such as analog and digital telephones.

For more information about features of the G350 Media Gateway, see *Overview of the Avaya G250 and G350 Media Gateways*, 03-300435.

Configurations

Deployment modes

The G350 Media Gateway is a modular device with multiple configuration possibilities to meet specific individual needs. Six slots in the G350 Media Gateway chassis house a customized selection of media modules. These media modules connect to different types of circuit switched phones, trunks, and data devices. One of the slots can house an internal server. A major configuration option is of which type of server to deploy. The server can be a media module or a stand-alone device.

The G350 Media Gateway can be deployed in one of two basic working modes:

- Distributed Communication Manager Branch Gateways. In this mode, an external server controls the G350 Media Gateway. This can be a stand-alone server, such as the S8510 or the S8800 server, or a separate Media Gateway in a stand-alone configuration.

  The G350 Media Gateway can also house an S8300D Server module to function as a Survivable Remote Server (Local Survivable Processor). This Survivable Remote Server can take over control of the G350 Media Gateway if the external server stops serving the G350 Media Gateway. For a summary of how the Survivable Remote Server in a G350 works, see S8300D Server in a Survivable Remote Server configuration.

- Stand-alone. In this mode, an internally housed S8300D Server module controls the G350 Media Gateway. See Avaya S8300 Server.

Multiple G350 Media Gateways can be deployed in many remote branches of a large organization. Large branches or main offices can deploy an Avaya G450 or G700 Media Gateway, which provides similar functionality to the G350 for a larger number of users. Up to 50 G350, G430, G450, and G700 Media Gateways can be controlled by a single S8300D Server housed in a G450 or G700 Media Gateway. Up to 250 G250, G350, G430, G450, and G700 Media Gateways can be controlled by a single S8510 or S8800 Server.

Expanded capacity and multiple G350 gateways in a branch

You can deploy multiple G350 Media Gateways in branch offices and benefit from increased capacities and additional configuration options. Beginning with Communication Manager Release 3.1.X, the advanced mode in Avaya Solution Designer reflects these additional capacities. Using the Solution Designer, you can build a G350 configuration and verify that it meets system resource limitations.

You can use the G350 Media Gateway with a S8300B as a primary server for up to five G250 or G350 Media Gateways. You can install any combination of media modules. These configurations are subject to traffic engineering rules. See Table 8: G350 capacities on page 82 for more information.

Note:

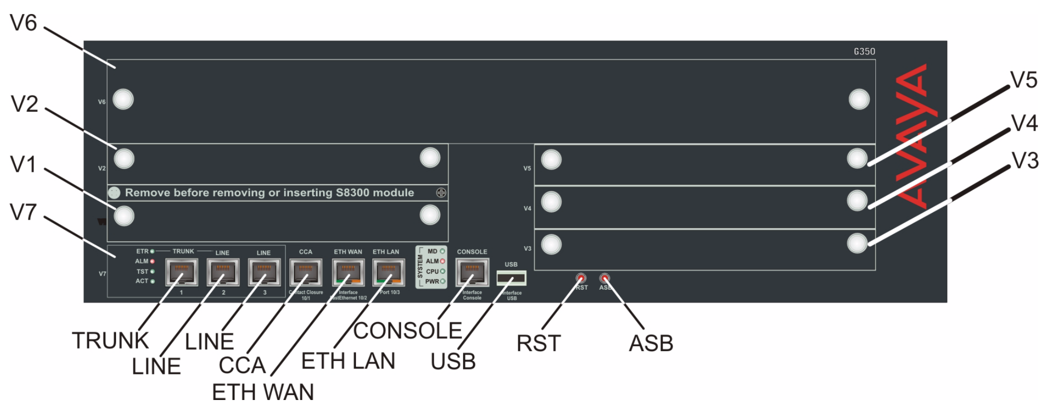
Maximum capacities depend on the specific configuration of the branch gateway. Verify your planned configuration with Avaya Solution Designer.

Components

Chassis

The following figure shows the G350 chassis.

Figure 16: G350 chassis



The chassis features:

- Six media module slots, V1 to V6.
- Fixed ports and buttons, including embedded analog media module V7.

Ports and buttons

Table 7 describes the functions of the fixed ports and buttons on the front panel of G350.

Table 7: Fixed ports and buttons on the G350 chassis

Port/Button	Description
TRUNK	An analog trunk port. Part of an integrated analog media module. The fixed trunk port supports loop-start, ground-start, CAMA, and DIOD (for Japan only) trunks.
LINE	Two analog telephone ports of the integrated analog media module. An analog relay between TRUNK port 7/1 and the furthest left LINE port 7/2 provides Emergency Transfer Relay (ETR) feature. Also used for incoming analog DID trunks. The G350 integrated analog line ports support three ringer loads, which is the ringer equivalency number (REN), for the following loop lengths: - 20,000 feet (6096 meters) over 0.65 mm (.025 in.) wire (22 AWG) - 16,000 feet (4877 meters) over 0.5 mm (.02 in.) wire (24 AWG) - 10,000 feet (3048 meters) over 0.4 mm (.016 in.) wire (26 AWG) At .1 or less REN ringer loads, the supported loop length is 20,000 feet (6096 meters) at 22, 24, and 26 AWG.
CCA	RJ-45 port for ACS (308) contact closure adjunct box.
ETH WAN 1	RJ-45 10/100 Base TX Ethernet WAN port.
ETH LAN 1	RJ-45 10/100 Base TX Ethernet LAN port.
CONSOLE	Console port for direct connection of CLI console. RJ-45 connector.
USB	USB port. Supports the connection of • USB flash drive • USB externally powered hub • The Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem.
RST	Reset button. Resets chassis.
ASB	Alternate Software Bank button. Reboots the G350 with the software image in the alternate bank.

Specifications

Dimensions and site requirements

The following table lists the physical dimensions and site requirements of the G350 Media Gateway.

Description	Value
Height	5.25 in. (133.3 mm)
Width	19 in. (482.6 cm)
Depth	15.75 in. (400 mm)
Weight of empty chassis	18.8 - 22 lb (9-10 kg)
Ambient working temperature	32°F-104°F (0-40°C)
Operation altitude	up to 6,561 ft. (2000 m)
Front clearance	12 in. (30 cm.)
Rear clearance	18 in. (45 cm.)
Humidity	10-90% non-condensing
Power rating	90-264 V~, 47-63 Hz
BTU	1038 BTU/h
Max current	6A

Power cord specifications

The specifications for power cords suitable for use with G350 are as follows:

For North America: The cordset must be UL Listed/CSA Certified, 16 AWG, 3-conductor (3rd wire ground), type SJT. One end is to be terminated to an IEC 60320, sheet C13 type connector rated 10A, 250V. The other end terminates to either a NEMA 5-15P attachment plug for nominal 125V applications or a NEMA 6-15P attachment plug for nominal 250V applications.

For Outside North America: The cord must be VDE Certified or Harmonized (HAR), rated 250V, 3- conductor (3rd wire ground), 1.0 mm2 minimum conductor size. The cord terminates at one end to a VDE Certified/CE Marked IEC 60320, sheet C13 type connector rated 10A, 250V. The cord's other end terminates to a 3-conductor grounding type attachment plug rated at a minimum of 10A, 250V. The configuration is specific for the region/country in which the cord is used. The attachment plug must bear the safety agency certifications mark(s) for the region/ country of installation.

The G350 Media Gateway relies on two ground connections. These connections are a mains plug with an earth contact and a permanent Supplementary Ground Conductor. Because of unreliable earthing concerns in Finland, Norway, and Sweden, the G350 Media Gateway must be installed in a Restricted Access Location (RAL). Only trained service personnel or customers can access the RAL. They know the reasons for the restricted access and any safety precautions that must be taken. In these cases, these personnel or customers must use a lock and key or other means of security when they access the G350 Media Gateway.

Related hardware

Media modules

Avaya media modules convert the voice path of the traditional circuits, such as analog trunk, T1/E1, and DCP to a TDM bus. The VOIP engine then converts the voice path from the TDM bus to packetized VoIP, compressed or uncompressed, on an Ethernet connection.

The media modules reside in the G350 Media Gateway and interact with the motherboard and backplane.

Note:

For stand-alone mode, the S8300D Server is inserted into slot V1. See Avaya S8300 Server on page 29.

There are nine telephony media modules:

- MM710 T1/E1 ISDN PRI – For information, see MM710 T1/E1 Media Module on page 212.
- MM711 Analog – For information, see MM711 Analog Media Module on page 215.
- MM712 DCP – For information, see MM712 DCP Media Module on page 218.
- MM714 Analog – For information, see MM714 Analog Media Module on page 219.
- MM716 Analog – For information, see MM716 Analog Media Module on page 222
- MM717 DCP – For information, see MM717 DCP Media Module on page 223.
- MM720 BRI – For information, see MM720 BRI Media Module on page 224.
- MM722 BRI – For information, see MM722 BRI Media Module on page 225.
- MM312 DCP – For information, see MM312 DCP Media Module on page 209.

There are two WAN media modules:

- MM340 T1/E1 WAN – For information, see MM340 E1/T1 data WAN Media Module on page 226.
- MM342 USP WAN – For information, see MM342 USP data WAN Media Module on page 227.

There are two LAN media modules:

- MM314 – For information, see MM314 LAN Media Module on page 210.
- MM316 – For information, see MM316 LAN Media Module on page 211

For more information about the G350 Media Gateway, see *Overview of the Avaya G250 and G350 Media Gateways*, 03-300435.

Survivability

The G350 Media Gateway supports Standard Local Survivability (SLS) that is a configurable software module. With SLS, a local G350 can provide a core set of Media Gateway Controller functions when no link is available to the server, a Survivable Remote Server, or a Survivable Core Server (Enterprise Survivable Server). SLS is configured on a system-wide basis using the new Provisioning and Installation Manager (PIM), or an individual G350 using the command line interface (CLI).

SLS is supported as follows on the G350 Media Gateway:

- G350 with C/S (hardware vintage) 3.0 and up: SLS supported for all analog interfaces, ISDN BRI/PRI trunk interfaces, non-ISDN digital DS-1 trunk interfaces, IP phone, IP Softphone, and DCP phone.

High-level capacities

The following table outlines the capacities of various G350 services.

Note:

Some capacities might change. For the most up-to-date list, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Table 8: G350 capacities

Description	Standard Configuration	Enhanced Configuration	Comments
Media Gateway Limits			
Maximum number of G350 Media Gateways controlled by an S8510 or S8800 Server	250		This number also applies if the same external S8*xxx* server controls a combination of Avaya G700/G650/G450/G430/G350/G250 Media Gateways.
			1 of 3

Table 8: G350 capacities ***(continued)***

Description	Standard Configuration	Enhanced Configuration	Comments
Maximum number of G350 Media Gateways controlled by an S8300D Server housed in a G450/ G700 Media Gateway	50		
Maximum number of G350 or G250 Media Gateways controlled by an S8300D Server housed in a G350 Media Gateway.	5		An S8300D housed in a G350 can also control Multitech gateways.
Maximum number of telephones supported by the G350	40	72	Limited by the physical hardware resources and what is supported in Avaya Solution Designer
Maximum number of IP telephones per G350 Media Gateway	40	72 (using an external switch)	Limited by the physical hardware resources and what is supported in Avaya Solution Designer
Maximum number of analog phones per G350 Media Gateway	40	72	
Maximum number of DCP phones per G350 Media Gateway	40	72	.
Maximum number of BRI endpoints per G350 Media Gateway	16	64	Up to three MM720 BRI Media Modules can be inserted in any standard media module slots.
Simultaneous two-way conversations from IP phone to legacy telephone or trunk.	32 – G.711 16 – G.729a, G.726		Simultaneous two-way conversations limited by the VoIP engine, including call progress tones.
Transcoding from G.711 to G.729 IP phones	16		Simultaneous two-way conversations.
			2 of 3

Table 8: G350 capacities ***(continued)***

Description	Standard Configuration	Enhanced Configuration	Comments
Transcoding from TDM phones to G.729 IP phones	16		Simultaneous 2-way conversations. For TDM transcoding, the number of 16 applies to conversations where one end of each conversation is on a G350 and transcoding occurs for that endpoint on the G350. If transcoding must occur on both ends of the conversation, the number of conversations is 10.
Maximum number of BRI trunks	16	32	Up to three MM720 BRI Media Modules can be inserted in any G350 media module slots.
Maximum number of PSTN trunks	24 (T1) 30 (E1)	48 (T1) 60 (E1)	Up to three MM711 Media Modules can be inserted into standard media module slots and used as trunks. The base unit has one analog trunk port. A full E1/T1 trunk group is supported for PSTN. An additional 15 IP trunks are also supported.
Miscellaneous			
Fax capacity	8		Simultaneous fax transmissions using VoIP resources
Touch-tone recognition (TTR)	15		
Tone Generation	15		
Announcements (VAL)	6 Playback 1 Record		
			3 of 3

Avaya G430 Media Gateway

The Avaya G430 Media Gateway is a multipurpose Media Gateway targeting small and medium branches of 1 to 150 users. The G430 Media Gateway supports two expansion modules to support varying branch office sizes. It works in conjunction with Communication Manager IP telephony software running on Avaya S8xxx Servers to help deliver intelligent communications to enterprises of all sizes.

The G430 Media Gateway combines telephone exchange and data networking by providing PSTN toll bypass and routing data and VoIP traffic over WAN. The G430 Media Gateway features a VoIP engine, an optional WAN router, and Ethernet LAN connectivity. The G430 Media Gateway provides full support for Avaya IP and digital telephones, as well as analog devices such as modems, fax machines, and telephones.

Detailed description

G430 can support up to 150 users when deployed as a branch gateway in a mid-to-large branch office of a large enterprise or a call center. The configuration requires Communication Manager IP telephony software running on one or more Avaya S8xxx Servers. The 150 user capacity is reached when the Avaya S8300D server is used.

Telephone services on a G430 are controlled by an Avaya S8xxx Server operating either as an External Call Controller (ECC) or as an Internal Call Controller (ICC). The G430 Media Gateway supports the Avaya S8300D Server as an ICC or as an ECC when the S8300D is installed in another Media Gateway. G430 also supports the Avaya S8800, duplex, and S8510 Servers as ECCs.

You can use an ICC in addition to an ECC with the ICC installed as a Survivable Remote Server (Local Survivable Processor) designed to take over call control in the event that the ECC fails or the WAN link between the branch office and main location breaks. The Survivable Remote Server provides full featured telephone service survivability for the branch office. G430 itself also features Standard Local Survivability (SLS), which provides basic telephone services in the event that the connection with the primary ECC is lost.

G430 is a scalable device with a basic configuration consisting of one power supply unit (PSU), 256 MB RAM, and a single on-board DSP supporting 20 VoIP channels. This configuration can be enhanced by adding a DSP board supporting either 10, 20, or 80 VoIP channels. You can also replace the 256 MB RAM with 512 MB RAM and use an external compact flash to increase the number of announcement files from 256 to 1024.

The G430 Media Gateway is a modular device, adaptable to support different combinations of endpoint devices. While fixed front panel ports support the connection of external LAN switches, network data ports, Ethernet WAN lines, and external routers, three slots are provided for plugging in optional media modules. Two EM200 expansion modules can be connected to the G430 Media Gateway, providing two media module slots each, bringing the total number of available media module slots to seven.

Pluggable media modules provide interfaces for different types of telephones and trunks. A combination is selected to suit the needs of the branch. A range of telephony modules provides full support for legacy equipment such as analog and digital telephones. IP phones are supported through an external LAN switch.

The G430 Media Gateway features a field replaceable RAM memory card and a DSP childboard.

The G430 Media Gateway chassis features field replaceable RAM, DSPs, PSUs, fan tray, and main board module for enhanced reliability.

For more information on the features of the G430 Media Gateway, see *Overview for the Avaya G430 Media Gateway*, 03-603235.

G430 physical description

Figure 17: Avaya G430 Media Gateway Chassis

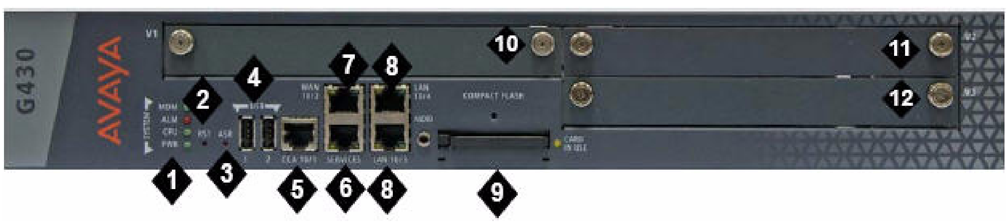


Figure notes:

1. **System LEDs**
2. **RST button**
3. **ASB button**
4. **USB ports**
5. **CCA (Contact Closure) port**
6. **Services port**
7. **ETH WAN port**
8. **ETH LAN ports**
9. **Compact Flash slot**
10. **V1 — slot for standard media module or S8300D Server**
11. **V2 — slot for standard media module**
12. **V3 — slot for standard media module**

Ports and buttons

Table 9: Fixed ports and buttons on the G430 front panel

Port/Button	Description
CCA	RJ-45 port for ACS (308) contact closure adjunct box.
ETH WAN	One 10/100 Base TX Ethernet WAN port. RJ-45 connector.
ETH LAN	Two 10/100 Base TX Ethernet LAN ports. RJ-45 connectors.
SERVICES	Ethernet 10/100 port for services and maintenance access. RJ-45 connector.
USB	Two USB ports with USB connectors. Supports the connection of: • USB flash drive (no more than one USB flash drive can be connected) • The Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem (no more than one USB modem can be connected)
RST	Reset button. Resets chassis configuration.
ASB	Alternate Software Bank button. Reboots the G430 with the software image in the alternate bank.

EM200 physical description

Figure 18: Avaya EM200 Media Gateway Chassis

Specifications

Physical dimensions and site requirements

The following table lists the physical dimensions and site requirements of the G430 Media Gateway.

Table 10: Avaya G430 Media Gateway specifications

Description	Value
Height	2.62 in. (66.5 mm)
Width	19 in. (482.6 mm)
Depth	12.8 in. (325 mm)
Weight of empty chassis	under 11 pounds (under 5 Kg)
Weight of chassis with basic configuration	13-14 pounds (6-7 Kg)
Ambient working temperature	32°F to 104°F (0°C to 40°C)
Operation altitude	up to 10,000 ft (3048 m)
Front Clearance	12 in. (30 cm)
Rear Clearance	18 in. (45 cm)
Humidity	10-90%, non-condensing
Power rating	90-264V AC, 47-63 Hz
BTU	800 BTU/h
Max current	2.4 A

G430 Power cord specifications

The specifications for power cords suitable for use with G430 are as follows:

For North America: The cord set must be UL Listed/CSA Certified, 16 AWG, 3-conductor (3rd wire ground), type SJT. The other end is to be terminated to an IEC 60320, sheet C13 type connector rated 10A, 250V. The other end is to be terminated to either a NEMA 5-15P attachment plug for nominal 125V applications or a NEMA 6-15P attachment plug for nominal 250V applications.

For Outside North America: The cord must be VDE Certified or Harmonized (HAR), rated 250V, 3-conductor (3rd wire ground), 1.0 mm2 minimum conductor size. The cord is to be terminated at one end to a VDE Certified/CE Marked IEC 60320, sheet C13 type connector rated 10A, 250V and the other end to a 3-conductor grounding type attachment plug rated at a minimum of 10A, 250V and a configuration specific for the region or country in which it will be used. The attachment plug must bear the safety agency certifications mark for the region or country of installation.

G430 Media module specifications

Table 11: Media modules

Description	Value
Height	0.79 in. (2 cm)
Width	6.69 in. (17 cm)
Depth	12.20 in. (31 cm)
Weight	0.7-0.9 (300-400 grams)

Supported media modules in the G430

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

The media modules reside in the G430 Media Gateway and interact with the motherboard and backplane.

Note:

For stand-alone mode, the S8300D Server is inserted into slot V1. See Avaya S8300 Server on page 29.

The following telephony media modules are supported by the G430 Media Gateway:

Table 12: Media Modules in the G430 ***1 of 2***

Media Module	See
MM710 T1/E1 ISDN PRI	MM710 T1/E1 Media Module
MM711 Analog	MM711 Analog Media Module
MM712 DCP	MM712 DCP Media Module

Table 12: Media Modules in the G430 ***2 of 2***

Media Module	See
MM714 Analog	MM714 Analog Media Module
MM714B	MM714B Analog Media Module
MM716 Analog	MM716 Analog Media Module
MM717 DCP	MM717 DCP Media Module
MM720 BRI	MM720 BRI Media Module
MM722 BRI	MM722 BRI Media Module

Media module slot configurations in G430

When choosing a combination of media modules to install in the G430 chassis and EM200 expansion modules, consider the slots in which each module type can be inserted and the limitations and recommendations regarding combinations of media modules.

The G430 chassis has three media module slots marked V1, V2, and V3 (see G430 physical description). The two optional EM200 expansion modules have two media module slots each (see EM200 physical description). The slots of the EM200 connected to the EXPANSION OUT 1 connector on the rear of the G430 are slots V5 and V6, and the slots of the EM200 connected to the EXPANSION OUT 2 connector on the rear of the G430 are slots V7 and V8. Each media module is restricted to certain slots:

Table 13: Permitted slots for media modules

Media Module	Permitted slots
MM710	Any media module slot V1-V3, V4-V8
MM711	Any media module slot V1-V3, V4-V8
MM712	Any media module slot V1-V3, V4-V8
MM714	Any media module slot V1-V3, V4-V8
MM714B	Any media module slot V1-V3, V4-V8
MM716	Any media module slot V1-V3, V4-V8
MM717	Any media module slot V1-V3, V4-V8
MM720	Any media module slot V1-V3, V4-V8

Table 13: Permitted slots for media modules *(continued)*

Media Module	Permitted slots
MM722	Any media module slot V1-V3, V4-V8
S8300D	V1

Survivability

You can configure Standard Local Survivability (SLS) to enable a local G430 to provide a degree of MGC functionality when no link is available to an external MGC. SLS is configured from the individual G430 itself using the command line interface. SLS is supported for all analog interfaces, ISDN BRI/PRI trunk interfaces, non-ISDN digital DS-1 trunk interfaces (T1 Robbed Bit and E1-CAS), IP phones, IP softphones, and DCP phones.

You can configure Enhanced Local Survivability (ELS) by installing an S8300D as a Survivable Remote Server (Local Survivable Processor). In this configuration, the S8300D is not the primary MGC but takes over to provide continuous telephone service if all external MGCs become unavailable. Calls in progress continue without interruption when the S8300D takes over.

High-level capacities

The following table outlines the capacities of various G430 services.

Note:

Some capacities might change. For the most up-to-date list, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Table 14: G430 capacities

Description	Capacity	Comments
Media Gateway Limits		
Maximum number of G430 Media Gateways controlled by an S8510 or S8800 Server	250	This number also applies if the same external server controls a combination of Avaya G430, G450, G350, G250, G650, and G700 Media Gateways.
		1 of 3

Table 14: G430 capacities *(continued)*

Description	Capacity	Comments
Maximum number of G430 Media Gateways controlled by an S8300D Server housed in another G430 (G450 or G700) Media Gateway	50	This number also applies if the same external server controls a combination of Avaya G430, G450, G350, G250, G650, and G700 Media Gateways.
Maximum total number of telephones supported by the G430	150	Assumes that the MGC is an S8300D installed in the G430 as an ICC. Otherwise, the capacity is greater.
Maximum number of IP telephones per G430 Media Gateway	150	Assumes that the MGC is an S8300D installed in the G430 as an ICC. Otherwise, the capacity is greater.
Maximum number of analog phones per G430 Media Gateway	56 104 for a G430 with one EM200 152 for a G430 with two EM200s	
Maximum number of DCP phones per G430 Media Gateway	56 104 for a G430 with one EM200 152 for a G430 with two EM200s	.
Maximum number of BRI endpoints per G430 Media Gateway	48 80 for a G430 with one EM200 112 for a G430 with two EM200s	
Simultaneous two-way conversations with TDM transcoding from IP phone to legacy telephone or trunk.	100	
		2 of 3

Table 14: G430 capacities *(continued)*

Description	Capacity	Comments
Simultaneous two-way conversations with TDM transcoding from TDM phones to IP phones	100	
Maximum number of BRI trunks	24 40 for a G430 with one EM200 56 for a G430 with two EM200s	
Maximum number of PSTN trunks	4 (T1) 3 (E1)	For E1/T1 trunks: 7 channels are supported in Tandem mode.
Miscellaneous		
Simultaneous fax transmissions	100	Fax transmissions using VoIP resources
Touch-tone recognition (TTR)	32	
Tone Generation	unlimited	
Announcements ports	15 ports for playback 1 for record	
		3 of 3

Avaya G450 Media Gateway

The Avaya G450 Media Gateway is a multipurpose Media Gateway that can be deployed in medium to large sized branch locations or in wiring-closets servicing buildings and floors in a campus environment. It works in conjunction with Communication Manager IP telephony software running on Avaya S8xxx Servers to help deliver intelligent communications to enterprises of all sizes.

The G450 Media Gateway combines telephone exchange and data networking by providing PSTN toll bypass and routing data and VoIP traffic over WAN. The G450 Media Gateway features a VoIP engine, an optional WAN router, and Ethernet LAN connectivity. The G450 Media Gateway provides full support for Avaya IP and digital telephones, as well as analog devices such as modems, fax machines, and telephones.

Detailed description

The G450 can support up to 450 users when deployed as a branch gateway in a mid-to-large branch office of a large enterprise or a call center and can serve up to 2400 users when deployed as a campus gateway. Both configurations require Communication Manager IP telephony software running on one or more Avaya S8xxx Servers. The Avaya S8300D server provides a capacity of 450 users, and the Avaya S8510 Server provides a capacity of 2400 users

Telephone services on a G450 are controlled by an Avaya S8xxx Server operating either as an External Call Controller (ECC) or as an Internal Call Controller (ICC). The G450 supports the Avaya S8300D Server as an ICC or as an ECC when the S8300D is installed in another Media Gateway. The G450 also supports the Avaya 8800, duplex, and S8510 Servers as ECCs.

In addition to an ECC, an ICC can be installed as a Survivable Remote Server (Local Survivable Processor) designed to take over call control when the ECC fails or WAN link between the branch office and main location breaks. The Survivable Remote Server provides full featured telephone service survivability for the branch office. G450 also features Standard Local Survivability (SLS) which provides basic telephone services when the connection with the primary ECC is lost.

The G450 is a scalable device with a basic configuration consisting of one power supply unit (PSU), 256 MB RAM, and a single DSP childboard supporting either 20 or 80 VoIP channels. This configuration can be enhanced by adding a redundant PSU, up to two RAM modules of 1 GB each, and up to three additional DSP childboards, increasing the number of VoIP channels to 240 channels.

The G450 Media Gateway is a modular device, adaptable to support different combinations of endpoint devices. While fixed front panel ports support the connection of external LAN switches, network data ports, Ethernet WAN lines and external routers, eight slots are provided for plugging in optional media modules. Pluggable media modules provide interfaces for different types of telephones, trunks, and WAN links. A combination is selected to suit the needs of the branch. A range of telephony modules provides full support for legacy equipment such as analog and digital telephones. A range of WAN modules provide support for Universal Serial Port and E1/T1 WAN links. IP phones are supported through an external LAN switch.

The G450 chassis features field replaceable RAM, DSPs, PSUs, fan tray, and main board module for enhanced reliability.

For more information about features of the G450 Media Gateway, see *Overview for the Avaya G450 Media Gateway*, 03-602058.

G450 physical description

Figure 19: Avaya G450 Media Gateway Chassis

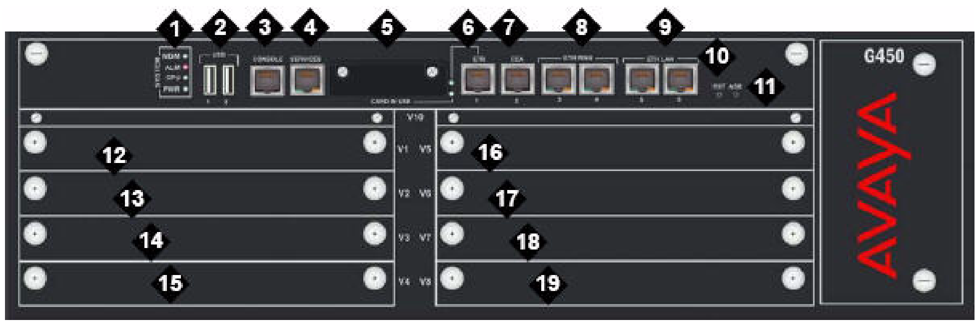


Figure notes:

1. **System LEDs**
2. **USB ports**
3. **Console port**
4. **Services port**
5. **Compact flash slot**
6. **ETR (Emergency Transfer Relay) port**
7. **CCA (Contact Closure Adjunct) port**
8. **ETH WAN ports**
9. **ETH LAN ports**
10. **RST button**
11. **ASB button**
12. **V1 — slot for standard media module or S8300D Server**
13. **V2 — standard media module slot**
14. **V3 — standard media module slot**
15. **V4 — standard media module slot**
16. **V5 — standard media module slot**
17. **V6 — standard media module slot**
18. **V7 — standard media module slot**
19. **V8 — standard media module slot**

Ports and buttons

Table 15: Fixed ports and buttons on the G450 front panel

Port/Button	Description
CCA	RJ-45 port for ACS (308) contact closure adjunct box.
ETH WAN	Two 10/100 Base TX Ethernet WAN port. RJ-45 connectors.
ETH LAN	Two 10/100/1000 Base TX Ethernet LAN ports. RJ-45 connectors.
CONSOLE	RS-232 port for services and maintenance access. RJ-45 connector.
SERVICES	Ethernet 10/100 port for services and maintenance access. RJ-45 connector.
ETR	Emergency Transfer Relay port. Controls two external 808A emergency transfer panels. RJ-45 connector.
USB	Two USB ports with USB connectors. Supports the connection of: • USB flash drive (no more than one USB flash drive can be connected) • The Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem (no more than one USB modem can be connected)
RST	Reset button. Resets chassis configuration.
ASB	Alternate Software Bank button. Reboots the G450 with the software image in the alternate bank.

Specifications

Physical dimensions and site requirements

The following table lists the physical dimensions and site requirements of the G450 Media Gateway.

Table 16: Avaya G450 Media Gateway specifications

Description	Value
Height	3U 5.25 in. (133.35 mm)
Width	19 in. (482.6 mm)
Depth	18 in. (456 mm)
Weight of empty chassis	16.5 pounds (7.4 Kg) 21 pounds (9.5 kg) with blank plates
Weight of chassis with basic configuration, including main board, power supply unit, fan tray, one DSP, and blank panels on the media module slots	31 pounds (14 Kg)*
Ambient working temperature	32° to 104°F (0° to 40°C)
Operation altitude	up to 10,000 ft (3048 m)
Front Clearance	12 in. (30 cm)
Rear Clearance	18 in. (45 cm)
Humidity	10-90%, non-condensing
Power rating	90-264V AC, 47-63 Hz
BTU	1,780 BTU/h
Max current	6 A

G450 power cord specifications

The specifications for power cords suitable for use with G450 are as follows:

For North America: The cord set must be UL Listed/CSA Certified, 16 AWG, 3-conductor (3rd wire ground), type SJT. One end is to be terminated to an IEC 60320, sheet C13 type connector rated 10A, 250V. The other end is to be terminated to either a NEMA 5-15P attachment plug for nominal 125V applications or a NEMA 6-15P attachment plug for nominal 250V applications.

For Outside North America: The cord must be VDE Certified or Harmonized (HAR), rated 250V, 3-conductor (3rd wire ground), 1.0 mm2 minimum conductor size. The cord is to be terminated at one end to a VDE Certified/CE Marked IEC 60320, sheet C13 type connector rated 10A, 250V and the other end to a 3-conductor grounding type attachment plug rated at a minimum of 10A, 250V and a configuration specific for the region/country in which it will be used. The attachment plug must bear the safety agency certifications mark(s) for the region/ country of installation.

G450 Media module specifications

Table 17: Media modules

Description	Value
Height	0.79 in. (2 cm)
Width	6.69 in. (17 cm)
Depth	12.20 in. (31 cm)
Weight	0.7-0.9 (300-400 grams)

Supported media modules in the G450

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

The media modules reside in the G450 Media Gateway and interact with the motherboard and backplane.

Note:

For stand-alone mode, the S8300D Server is inserted into slot V1. See Avaya S8300 Server on page 29.

The following telephony media modules are supported by the G450 Media Gateway:

Table 18: Media Modules in the G450 ***1 of 2***

Media Module	See
MM710 T1/E1 ISDN PRI	MM710 T1/E1 Media Module
MM711 Analog	MM711 Analog Media Module
MM712 DCP	MM712 DCP Media Module
MM714 Analog	MM714 Analog Media Module
MM716 Analog	MM716 Analog Media Module
MM717 DCP	MM717 DCP Media Module
MM720 BRI	MM720 BRI Media Module

Table 18: Media Modules in the G450 ***2 of 2***

Media Module	See
MM722 BRI	MM722 BRI Media Module
MM340 T1/E1 WAN	MM340 E1/T1 data WAN Media Module
MM342 USP WAN	MM342 USP data WAN Media Module

Media Module slot configurations in the G450

When choosing a combination of media modules to install in G450 chassis, consider the slots in which each module type can be inserted, and the limitations and recommendations regarding combinations of media modules.

The G450 chassis has eight media module slots marked V1, V2, V3, V4, V5, V6, V7, and V8 (see G450 physical description). Each medial module is restricted to certain slots:

Table 19: Permitted slots for media modules

Media Module	Permitted slots
MM340	V3, V4, V8
MM342	V3, V4, V8
MM710	Any media module slot V1-V8
MM711	Any media module slot V1-V8
MM712	Any media module slot V1-V8
MM714	Any media module slot V1-V8
MM716	Any media module slot V1-V8
MM717	Any media module slot V1-V8
MM720	Any media module slot V1-V8
MM722	Any media module slot V1-V8
S8300	V1

Survivability

You can configure Standard Local Survivability (SLS) to enable a local G450 to provide a degree of MGC functionality when no link is available to an external MGC. SLS is configured from the individual G450 itself using the command line interface. SLS is supported for all analog interfaces, ISDN BRI/PRI trunk interfaces, non-ISDN digital DS-1 trunk interfaces (T1 Robbed Bit and E1-CAS), IP phones, IP softphones, and DCP phones.

You can configure Enhanced Local Survivability (ELS) by installing an S8300D as a Survivable Remote Server (Local Survivable Processor). In this configuration, the S8300D is not the primary MGC but takes over to provide continuous telephone service if all external MGCs become unavailable. Calls in progress continue without interruption when the S8300D takes over.

High-level capacities

The following table outlines the capacities of various G450 services.

Note:

Some capacities might change. For the most up-to-date list, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Table 20: G450 capacities

Description	Capacity	Comments
Media Gateway Limits		
Maximum number of G450 Media Gateways controlled by an S8510 or S8800 Server	250	This number also applies if the same external server controls a combination of Avaya G450, G430, G350, G250, and G700 Media Gateways.
Maximum number of G450 Media Gateways controlled by an S8300D Server housed in another G450 Media Gateway	50	This number also applies if the same external server controls a combination of Avaya G450, G430, G350, G250, and G700 Media Gateways.
Maximum number of G450 Media Gateways controlled by an S8300D Server housed in a G700 Media Gateway.	50	This number also applies if the same external server controls a combination of Avaya G450, G430, G350, G250, and G700 Media Gateways.
		1 of 3

Table 20: G450 capacities ***(continued)***

Description	Capacity	Comments
Maximum total number of telephones supported by the G450	450	Assumes that the MGC is an S8300D installed in the G450 as an ICC. Otherwise, the capacity is greater.
Maximum number of IP telephones per G450 Media Gateway	450	Assumes that the MGC is an S8300D installed in the G450 as an ICC. Otherwise, the capacity is greater.
Maximum number of analog phones per G450 Media Gateway	192	
Maximum number of DCP phones per G450 Media Gateway	192	.
Maximum number of BRI endpoints per G450 Media Gateway	128	
Simultaneous two-way conversations with TDM transcoding from IP phone to legacy telephone or trunk.	206	
Simultaneous two-way conversations with TDM transcoding from TDM phones to IP phones	206	
Maximum number of BRI trunks	64	
Maximum number of PSTN trunks	184 (T1) 240 (E1)	For E1 trunks: 240 channels are supported in Tandem mode. 206 channels are supported for IP to PSTN.
Miscellaneous		
Simultaneous fax transmissions	240	Fax transmissions using VoIP resources
Touch-tone recognition (TTR)	64	
		2 of 3

Table 20: G450 capacities *(continued)*

Description	Capacity	Comments
Tone Generation	unlimited	
Announcements ports	63 ports for playback 1 for record	
		3 of 3

IG550 Integrated Gateway

The IG550 Integrated Gateway is a part of Avaya growing solutions for extending Communication Manager communication capabilities from the headquarters of an organization to all collaborative branch locations. The IG550 Integrated Gateway is an H.248 Media Gateway that combines Avaya high-performance telephony and Voice over IP (VoIP) communications with the sophisticated routing capabilities of the Juniper J-Series Services Routers.

Detailed description

The IG550 consists of the TGM550 Telephony Gateway Module (TGM550) and Telephony Interface Modules (TIMs). The IG550 is inserted into a Juniper J2320, J2350, J4350, or J6350 Services Router. The IG550 is also connected over a LAN or WAN to an Avaya server running Communication Manager. Therefore, Avaya S8800, S8510, and S8300D Servers are able to provide the same top quality telephony services to the small branch office as to the headquarters of the organization. As a result, IG550 provides full feature support for IP and analog telephones.

The IG550 is designed for use in a 2-to-100 user environment. The IG550 can be appropriately configured and priced to more precisely match the number of users.

The IG550 features Standard Local Survivability (SLS). SLS provides partial backup Media Gateway controller (MGC) functionality in the event that the connection with the primary MGC is lost.

In addition to advanced and comprehensive telephony services that are provided by the TGM550, the Juniper J-series Router, the J2320, J2350, J4350 or J6350 provides full data networking services, precluding the need for a WAN router. The J-series routers use Juniper Physical Interface Modules (PIMs) for the hardware components to support network and routing features. The J-series routers also provide Ethernet connections to a separate Ethernet switch that IP phones connect to.

For more information on the features of IG550 Integrated Gateway and the J-series Service Routers, see *Overview of the Avaya IG550 Integrated Gateway*, 03-601548.

Configurations

The IG550 Integrated Gateway is available with three capacity levels depending on which version of the TGM550 is used. The versions of the TGM550 are:

- TGM550 MP20 supports up to 20 concurrent VoIP calls, depending on the types of calls
- TGM550 MP80 supports up to 80 concurrent VoIP calls
- TGM550 MP10 supports up to 10 concurrent VoIP calls

Any J-series router can house a single TGM550 of any of the three versions.

Components

IG550 and J4350 Services Router

Figure 20: The IG550 Integrated Gateway in a J4350 Services Router

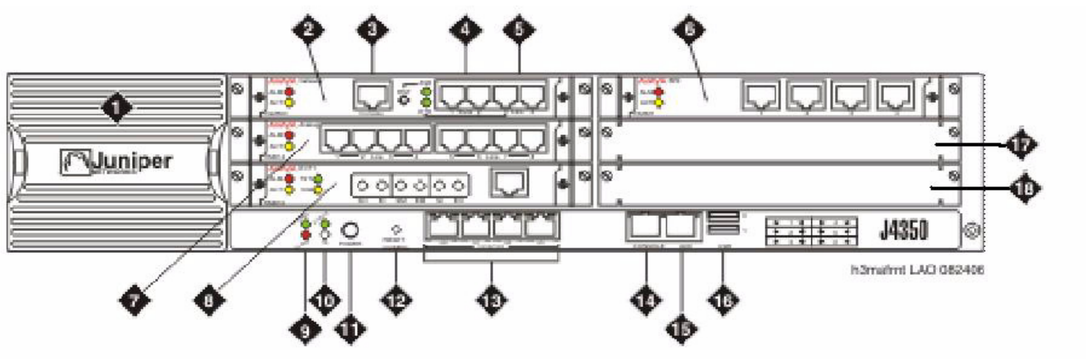


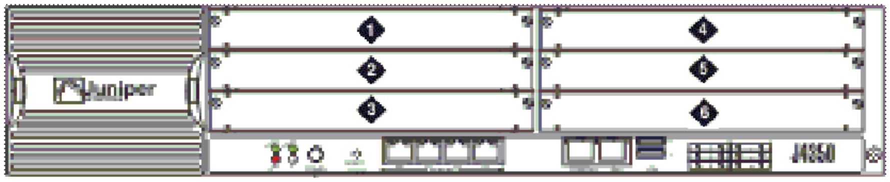
Figure notes:

1. **Juniper Services Router, J4350 shown**
2. **TGM550 Telephony Gateway Module (in slot V1)**
3. **TGM550 console port**
4. **TGM550 analog trunk ports**
5. **TGM550 analog line ports**
6. **TIM521 BRI telephony interface module (in slot V4)**
7. **TIM514 analog telephony interface module (in slot V2)**
8. **TIM510 E1/T1 telephony interface module (in slot V3)**
9. **J-series Router Alarm LEDs**
10. **J-series Router Power LEDs**
11. **Power button**
12. **Reset button**
13. **Gigabit Ethernet ports**
14. **Console port**
15. **Aux port**
16. **USB ports**
17. **Slot V5 (empty in illustration)**
18. **Slot V6 (empty in illustration)**

Slot locations on J4350 Services Router

The slots on the J4350 Services Router are identified as follows:

Figure 21: Slot numbers on the Juniper J4350 Services Router



The J4350 Services router chassis has six slots. Modules can be inserted into the slots according to the following guidelines:

- The TGM550 and TIMs can be housed in any of the six router slots.
- Fast Ethernet and Gigabit Ethernet ePIMs and the 16-port GigaE uPIM can be housed only in slots 3 or 6.
- The 16-port GigaE uPIM can be housed in slot 2, 3, 5 or 6.
- Other PIMs, including all other uPIMs, can be housed in any slots.

Fixed ports and buttons on the Juniper J4350 Services Router

Table 21: Fixed ports and buttons on the Juniper J4350 Services Router

Port/Button	Description
Gigabit Ethernet	Four Gigabit Ethernet ports. The JUNOS software identifies the port locations, from left to right, as ge-0/0/0, ge-0/0/1, ge-0/0/2, and ge-0/0/3. One port can serve as a management interface, typically ge-0/0/0.
Alarm LED	Lights yellow for a minor alarm condition, red for a major alarm condition, or is off when no alarm conditions exist. Alarm notification applies only to the J-series router, not to the TGM550.
Power LED	Green light that lights steadily, blinks, or is off to show power on/off status.
Status LED	Blinks to show startup of the router, lights steadily to show normal operation after startup, and red to indicate an error condition upon startup.
Console	Console RS-232 interface port for direct connection of CLI console. RJ-45 connector.
USB	Two USB ports. Support the connection of • Disk on Key USB memory stick • USB flash drive • The Multitech MT5634ZBA-USB-V92 USB modem.
	1 of 2

Table 21: Fixed ports and buttons on the Juniper J4350 Services Router *(continued)*

Port/Button	Description
Power button	Turns on power to the router and TGM550.
Reset button	Resets chassis configuration to either rescue configuration or factory default, if rescue not available. Resends configuration data to the TGM550. If the button is held 12 or more seconds, the root password is also reset.
Aux	Not activated.
	2 of 2

IG550 and J6350 Services Router

Figure 22: The IG550 Integrated Gateway in a J6350 Services Router

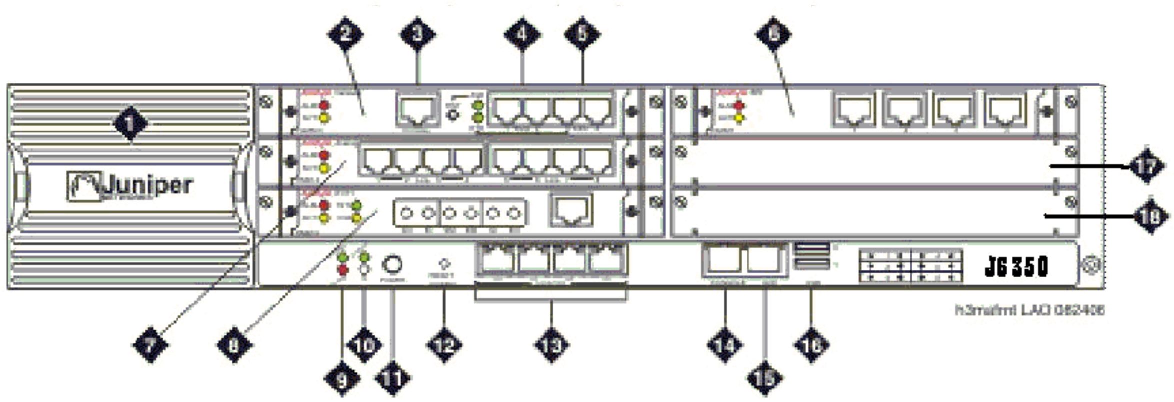


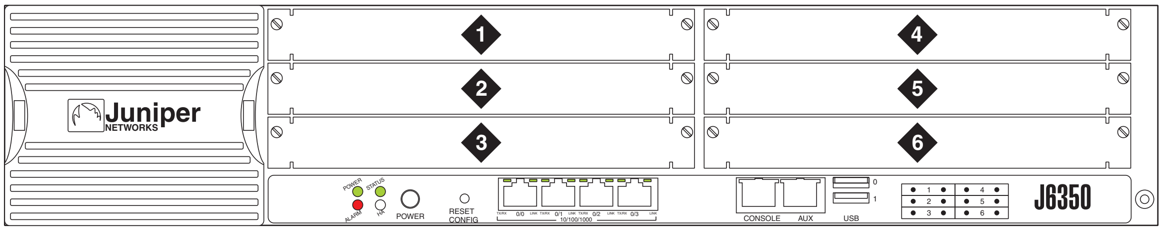
Figure notes:

1. **Juniper Services Router, J4350 shown**
2. **TGM550 Telephone Gateway Module (in slot V1)**
3. **TGM550 console port**
4. **TGM550 analog trunk ports**
5. **TGM550 analog line ports**
6. **TIM521 BRI telephony interface module (in slot V2)**
7. **TIM514 analog telephony interface module (in slot V2)**
8. **TIM510 E1/T1 telephony interface module (in slot V4)**
9. **J-series Router Alarm LEDs**
10. **J-series Router Power LEDs**
11. **Power button**
12. **Reset button**
13. **Gigabit Ethernet ports**
14. **Console port**
15. **Aux port**
16. **USB ports**
17. **Slot V5 (empty)**
18. **Slot V6 (empty)**

Slot locations on J6350 Services Router

The slots on the J6350 Services Router are identified as follows:

Figure 23: Slot numbers on the Juniper J6350 Services Router



The J6350 Services router chassis has six slots. Modules can be inserted into the slots according to the following guidelines:

- The TGM550 and TIMs can be housed in any of the six router slots.
- Fast Ethernet and Gigabit Ethernet ePIMs and the 16-port GigaE uPIM can be housed only in slots 2, 3, 5, or 6.

Other PIMs, including all other uPIMs, can be housed in any slot.

Fixed ports and buttons on the Juniper J6350 Services Router

Table 22: Fixed ports and buttons on the Juniper J6350 Services Router

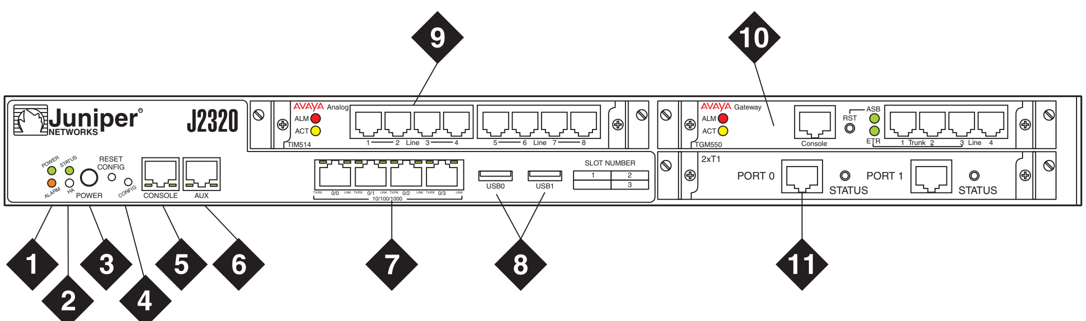
Port/Button	Description
Gigabit Ethernet	Four Gigabit Ethernet ports. The JUNOS software identifies the port locations, from left to right, as ge-0/0/0, ge-0/0/1, ge-0/0/2, and ge-0/0/3. One port can serve as a management interface, typically ge-0/0/0.
Alarm LED	Lights yellow for a minor alarm condition, red for a major alarm condition, or is off when no alarm conditions exist. Alarm notification applies only to the J-series router, not to the TGM550.
Power LED	Green light that lights steadily, blinks, or is off to show power on/off status.
Status LED	Blinks to show startup of the router, lights steadily to show normal operation after startup, and red to indicate an error condition upon startup.
Console	Console RS-232 interface port for direct connection of CLI console. RJ-45 connector.
	1 of 2

Table 22: Fixed ports and buttons on the Juniper J6350 Services Router *(continued)*

Port/Button	Description
USB	Two USB ports. Supports the connection of • Disk-on-Key USB memory stick • USB flash drive • Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem.
Power button	Turns on power to the router and TGM550.
Reset button	Resets chassis configuration to either rescue configuration or factory default, if rescue not available. Resends configuration data to TGM550. If the button is held 12 or more seconds, the root password is also reset.
Aux	Not activated.
	2 of 2

IG550 and J2320 Services Router

Figure 24: Example of IG550 Integrated Gateway in a J2320 Services Router



hwma232c LAO 070507

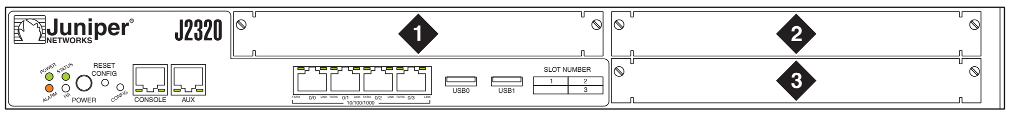
Figure notes:

1. **J-series Router Alarm LEDs**
2. **J-series Router Power LEDs**
3. **Power button**
4. **Reset button**
5. **Console port**
6. **Aux port**
7. **Gigabit Ethernet ports**
8. **USB ports**
9. **TIM514 analog telephony interface module (in slot V1)**
10. **TGM550 Telephony Gateway Module (in slot V2)**
11. **Dual port T1 PIM (in slot V3)**

Slot locations on J2320 Services Router

The slots on the J2320 Services Router are identified as follows:

Figure 25: Slot numbers on the Juniper J2320 Services Router



hwma232s LAO 061407

The J2320 Services router chassis has three slots. Modules can be inserted into the slots according to the following guidelines:

- The TGM550 and TIMs can be housed in any of the three router slots.
- The 16-port GigaE uPIM must be inserted into slot 3.
- All other supported PIMs, including all other uPIMs, can be housed in any slots.

Note:

J2320 does *not* support the following PIMs:

- Any of the ePIMs
- T3/E3 PIMs
- The four-port fast Ethernet PIM

Fixed ports and buttons on the Juniper J2320 Services Router

Table 23: Fixed ports and buttons on the Juniper J2320 Services Router

Port/Button	Description
Gigabit Ethernet	Four Gigabit Ethernet ports. The JUNOS software identifies the port locations, from left to right, as ge-0/0/0, ge-0/0/1, ge-0/0/2, and ge-0/0/3. One port can serve as a management interface, typically ge-0/0/0.
Alarm LED	Lights yellow for a minor alarm condition, red for a major alarm condition, or is off when no alarm conditions exist. Alarm notification applies only to the J-series router, not to the TGM550.
Power LED	Green light that lights steadily, blinks, or is off to show power on/off status.
Status LED	Blinks to show startup of the router, lights steadily to show normal operation after startup, and red to indicate an error condition upon startup.
	1 of 2

Table 23: Fixed ports and buttons on the Juniper J2320 Services Router *(continued)*

Port/Button	Description
Console	Console RS-232 interface port for direct connection of CLI console. RJ-45 connector.
USB	Two USB ports. Supports the connection of • Disk on Key USB memory stick • USB flash drive • Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem.
Power button	Turns on power to the router and TGM550.
Reset button	Resets chassis configuration to either rescue configuration or factory default, if rescue not available. Resends configuration data to TGM550. If the button is held 12 or more seconds, the root password is also reset.
Aux	Not activated.
	2 of 2

IG550 and J2350 Services Router

Figure 26: Example of IG550 Integrated Gateway in a J2350 Services Router

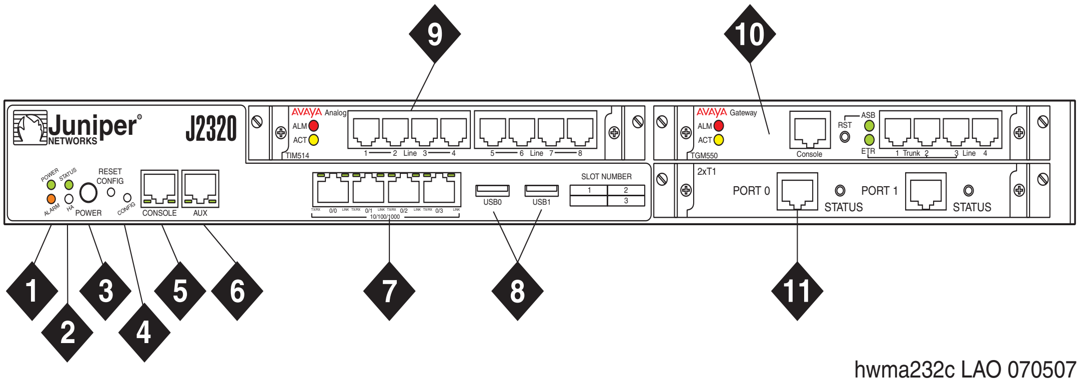


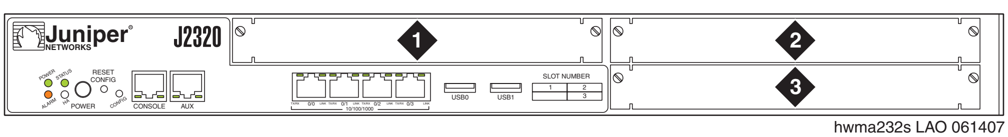
Figure notes:

1. **J-series Router Alarm LEDs**
2. **J-series Router Power LEDs**
3. **Power button**
4. **Reset button**
5. **Console port**
6. **Aux port**
7. **Gigabit Ethernet ports**
8. **USB ports**
9. **TIM514 analog telephony interface module (in slot V1)**
10. **TIM508 (in slot V2**
11. **TGM550 Telephony Gateway Module (in slot V3)**
12. **Dual port T1 PIM (in slot V4)**
13. **TIM510 (in slot V5)**

Slot locations on J2350 Services Router

The slots on the J2350 Services Router are identified as follows:

Figure 27: Slot numbers on the Juniper J2350 Services Router



The J2350 Services router chassis has five slots. Modules can be inserted into the slots according to the following guidelines:

- The TGM550 and TIMs can be housed in any of the five router slots.
- The 16-port GigaE uPIM must be inserted into slot 2, 4, or 5.
- All other supported PIMs, including all other uPIMs, can be housed in any slots.

The J2350 does *not* support the following PIMs:

- Any of the ePIMs
- T3/E3 PIMs
- The Dual-port fast Ethernet PIM

Fixed ports and buttons on the Juniper J2350 Services Router

Figure 28: Fixed ports and buttons on the Juniper J2350 Services Router

Port/Button	Description
Gigabit Ethernet	Four Gigabit Ethernet ports. The JUNOS software identifies the port locations, from left to right, as ge-0/0/0, ge-0/0/1, ge-0/0/2, and ge-0/0/3. One port can serve as a management interface, typically ge-0/0/0.
Alarm LED	Lights yellow for a minor alarm condition, red for a major alarm condition, or is off when no alarm conditions exist. Alarm notification applies only to the J-series router, not to the TGM550.
Power LED	Green light that lights steadily, blinks, or is off to show power on/off status.
Status LED	Blinks to show startup of the router, lights steadily to show normal operation after startup, and red to indicate an error condition upon startup.
Console	Console RS-232 interface port for direct connection of CLI console. RJ-45 connector.
USB	Two USB ports. Support the connection of • Disk on Key USB memory stick • USB flash drive • The Multitech MultiModemUSB MT5634ZBA-USB-V92 USB modem.
Power button	Turns on power to the router and TGM550.
Reset button	Resets chassis configuration to either rescue configuration or factory default, if rescue not available. Resends configuration data to TGM550. If the button is held 12 or more seconds, the root password is also reset.
Aux	Not activated.

TGM550 Gateway Module

All versions of the TGM550, including MP20, MP80, and MP10, have the same faceplate, ports, buttons, and LEDs. The customer can upgrade the capacity of the TGM550 by ordering a field replacement of the Digital Signal Processor (DSP), versions of which are identified as MP20, MP80, and MP10.

Figure 29: The TGM550 Gateway Module

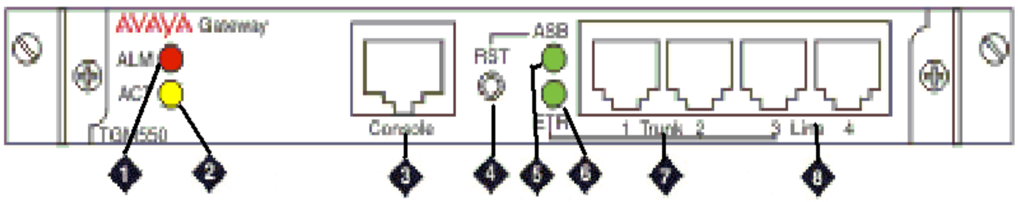


Figure notes:

1. Alarm LED
2. ACT LED
3. Console port
4. RST button
5. ASB LED
6. ETR LED
7. Analog trunk ports
8. Analog line ports

Fixed ports and buttons on the TGM550 Gateway Module

Table 24: Fixed ports and buttons on the TGM550

Port/Button	Description
ALM LED	Lights red to indicate an alarm on the TGM550 or a reboot.
ACT LED	Lights yellow to show activity of trunk or line ports. Also lights yellow during a reboot.
CONSOLE	Console port for direct connection of TGM550 CLI console. RJ-45 connector.
RST	Reset button. Resets the TGM550 configuration. It also reboots the TGM550 with the software image in the alternate bank.
ASB	Alternate Software Bank LED. Lights green if the software is not running from the selected boot bank.
	1 of 2

Table 24: Fixed ports and buttons on the TGM550 *(continued)*

Port/Button	Description
ETR	Lights green if the Emergency Transfer Relay is active or the TGM550 reboots. ETR uses trunk port 2 and line port 3.
Analog Trunk	Two analog trunk ports
Analog Line	Two analog trunk lines
	2 of 2

Specifications

The IG550 technical specifications include physical dimensions and tolerances of the Juniper J-series Services Router, power cord specifications, and TGM550 Gateway Module specifications.

J2320 Services Router specifications

Table 25: J2320 Services Router specifications

Description	Value
Height	1.75 in. (44.45 mm)
Width	17.5 in. (44.5 cm)
Depth	15.1 in. (38.4 cm)
Weight of empty chassis	14.8 lb (6.7 kg)
Ambient working temperature	32°F to 104°F (0° to 40°C)
Operation altitude	up to 10,000 ft. (3,048 m)
Front Clearance	6 in. (15 cm)
Rear Clearance	6 in. (15 cm)
Humidity	5-90% relative humidity
Power rating	AC: 100-240 VAC, 50 to 60 Hz, 6 to 8 A, 350 Watts; DC: -48 to -60 VDC, 420 Watts

J2350 Services Router specifications

Table 26: J2350 Services Router specifications

Description	Value
Height	2.61 in. (66.22 mm)
Width	17.5 in. (44.5 cm)
Depth	15.1 in. (38.4 cm)
Weight of empty chassis	16.3 lb (7.4 kg) 0
Ambient working temperature	32° to 104°F (0° to 40°C)
Operation altitude	up to 10,000 ft. (3,048 m)
Front Clearance	6 in. (15 cm)
Rear Clearance	6 in. (15 cm)
Humidity	5-90% relative humidity
Power rating	AC: 100-240 VAC, 50 to 60 Hz, 6 to 8 A, 350 Watts; DC: -48 to -60 VDC, 420 Watts

J4350/J6350 Services Router specifications

The table of technical specifications provides detailed information on the physical dimensions and tolerances of the J4350/J6350 Services Router:

Table 27: J4350/J6350 Services Router specifications

Description	Value
Height	3.5 in. (8.9 cm)
Width	17.5 in. (44.5 cm)
Depth	21.5 in. (54.6 cm)
Weight of empty chassis	23.0 lb (10.4 kg) — J4350; 25.0 lb (11.3 kg) — J6350
Ambient working temperature	32° to 104°F (0° to 40°C)
Operation altitude	up to 10,000 ft. (3,048 m)
Front Clearance	6 in. (15 cm)
Rear Clearance	6 in. (15 cm)

Table 27: J4350/J6350 Services Router specifications

Description	Value
Humidity	5-90% relative humidity
Power rating	AC: 100-240 VAC, 50 to 60 Hz, 6 to 8 A, 350 Watts; DC: -48 to -60 VDC, 420 watts

J-series Services Router power cord specifications

AC power cord

Detachable AC power cords, each 2.5 m (approximately 8 ft) long, are supplied with the Services Router. The appliance coupler at the female end of the cord inserts into the appliance inlet on the faceplate of the AC power supply. The coupler is type C19 as described by International Electrotechnical Commission (IEC) standard 60320. The plug at the male end of the power cord fits into the power source receptacle that is standard for your geographical location.

Note:

In North America, AC power cords must not exceed 4.5 m (approximately 14.75 ft) in length to comply with National Electrical Code (NEC) Sections 400-8 (NFPA 75, 5-2.2) and 210-52 and Canadian Electrical Code (CEC) Section 4-010(3). The cords supplied with the router are in compliance.

Country	Electrical Specifications	Plug Standards
Australia	250 VAC, 10 A, 50 Hz	AS/NZ 3112 1- 993
China	250 VAC, 10 A, 50 Hz	GB2099.1 1996 and GB1002 1996 (CH1-10P)
Europe (except Italy and United Kingdom)	250 VAC, 10 A, 50 Hz	CEE (7) VII
Italy	250 VAC, 10 A, 50 Hz	CEI 23 - 16/VII
Japan	125 VAC, 12 A, 50 Hz or 60 Hz	JIS 8303
North America	125 VAC, 10 A, 60 Hz	NEMA 5-15
United Kingdom	250 VAC, 10 A, 50 Hz	BS 1363A

DC power cord

Each DC power supply has a single DC input (-48 VDC and return) that requires a dedicated 15 A (-48 VDC) circuit breaker. If the J6350 router contains redundant DC power supplies, one power supply must be powered by a dedicated power feed derived from feed A, and the other power supply must be powered by a dedicated power feed derived from feed B. This configuration provides the commonly deployed A/B feed redundancy for the system.

Most sites distribute DC power through a main conduit that leads to frame-mounted DC power distribution panels, one of which might be located at the top of the rack that houses the router. A pair of cables (one input and one return) connects each set of terminal studs to the power distribution panel.

Each DC power cable (ñ48 VDC and return) must be 14 AWG single-strand wire cable or as permitted by the local code. Each lug attached to the power cables must be a ring-type, vinyl-insulated TV14-6R lug, or equivalent.

TGM 550 Gateway Module specifications

Table 28: TGM550 Gateway Module

Description	Value
Ambient working temperature	32°F to 158°F (0°C to 70°C)
Operation altitude	up to 10,000 ft. (3,048 m)

Grounding cable for IG550

When housing a TGM550, the J-series router must use a grounding cable that meets the following specifications:

- 10 AWG
- Able to handle up to 8 Amp current
- Have a ring-type, vinyl-insulated TV14-6R lug, or equivalent, to accommodate the 10 AWG cable

> **CAUTION:**
> The original grounding cable for Juniper Services Routers is 14 AWG only and must be replaced with a 10 AWG cable.

Related hardware

The IG550 Gateway Module supports a variety of optional internal boards called Telephony Interface Modules (TIMs). In addition, the Juniper J-series Services Routers support swappable internal components called Physical Interface Modules (PIMs).

Supported optional modules in IG550

Note:

The list of PIMs for J-series routers is a sample only. For a complete list of PIMs, see Juniper J-series Router documentation at http://juniper.net.

Table 29: Supported interface modules

Modules	Description
Telephony Interface Modules	
TM508	8 analog line or station ports, which can be administered as DID trunk ports
TIM514	4 analog line or station ports and 4 analog trunk ports
TM516	16 analog line or station ports. Off-Premise Stations are not supported.
TM518	8 analog line or station ports and 8 analog trunks
TIM510	1 E1/T1 trunk port, a DS-1 level port that provides a wide variety of E1 or T1 circuit support. Can provide up to 30 E1 or 24 T1 channels
TIM521	4 ISDN BRI trunk ports providing up to 8 bearer channels
J-series Router Physical Interface Modules	
Dual-Port Serial PIM	2 serial ports
Dual-Port T1 or E1 PIM	2 E1/T1 ports, each providing up to 30 E1 or 24 T1 data channels for WAN connections
Dual-Port Channelized T1 or E1 PIM	2 T1 or E1 ports
T3 or E3 PIM	1 E3/T3 port for WAN connections
Gigabit Ethernet SFP ePIM	One Gigabit port. Supported on the J4350 and J6350 Services routers only.
Gigabit Ethernet copper ePIM	One Gigabit port. Supported on the J4350 and J6350 Services routers only.
Dual-Port Fast Ethernet PIM	2 Fast Ethernet ports. Supported on the J4350 and J6350 Services routers only.
	1 of 2

Table 29: Supported interface modules *(continued)*

Modules	Description
Four-Port Fast Ethernet ePIM	4 Fast Ethernet ports. Supported on the J4350 and J6350 Services routers only.
4-Port ISDN BRI S/T PIM	4 ISDN BRI data-only ports
4-Port ISDN BRI U PIM	4 ISDN BRI data-only ports
1-, 6-, 8-, or 16-Port GigaE uPIM	6-, 8-, or 16-Gigabit Ethernet ports **Note:** The 16-port GigaE uPIM requires two slots in the router.
ADSL PIM (Annex A)	One port for DSL over an analog trunk
ADSL PIM (Annex B)	One port for ADSL over ISDN providing up to 32 virtual channels
G.SHDSL PIM	Two ports for 32 virtual channels of ATM over SHDSL connections
	2 of 2

TIM combination limitations in IG550

This table lists the maximum limits of TIM combination in IG550.

Table 30: TIM combination limitations

	J2320 slots 1-3	J2350 slots 1-5	J4350/J6350 slots 1-6
Maximum number of interface TIMs (excluding TGM)	2	4	4
Maximum number of TIM516s (Analog)	1	2	3
Maximum number of TIM514s (Analog)	2	4	4
Maximum number of TIM508s (Analog)	1	3	3
Maximum number of TIM518s (Analog)	1	3	3
Maximum number of TIM521s (BRI)	2	4	4
Maximum number of TIM510s (E1/T1)	2	4	4

Note:

The limitations listed in this section are recommended maximums. You must also calculate the power requirements and heat generation for the specific TIM and PIM combination the customer wants to ensure the J-series router can support that combination. See the information on limits based on heat and power used by IG550 in *Overview of the Avaya IG550 Integrated Gateway*, 03-601548.

For more information on each of TIMs, see Telephony Interface Modules.

Survivability

You can configure Standard Local Survivability (SLS) to enable a local IG550 to provide a degree of MGC functionality when no link is available to an external MGC. SLS is configured on a system-wide basis using the Provisioning and Installation Manager (PIM). Alternatively, SLS can be configured from the individual IG550 itself using the CLI. SLS supports all analog interfaces, ISDN BRI/PRI trunk interfaces, non-ISDN digital DS-1 trunk interfaces, IP phones, and IP Softphones.

High-level capacities

For information on system capacities of IG550 Integrated Gateway, see *Overview of the Avaya IG550 Integrated Gateway* (03-601548), *Avaya Aura® Communication Manager System Capacities Table* (03-300511), and other related documents at www.avaya.com/support.

Avaya G650 Media Gateway

The Avaya G650 Media Gateway is a 14-slot, rack mounted carrier configured for TN form factor circuit packs. The G650 Media Gateway is used with the S8510 and S8800 Servers.

Detailed description

The G650 Media Gateway is 8U or 14-inche (35.6 centimeters) high can be mounted on a standard 19-inch (48.3 centimeters) data rack. The G650 Media Gateway uses one or two 655A power supply ports, operating on AC and/or DC input power. Either power supply can provide all the power needed by G650 Media Gateway. When two power supplies exist, they share the power load. One power supply can operate on AC power and the other on DC power. However, each power supply has its own AC power cord so that both power sources can supply power to the Media Gateway simultaneously. Both power supplies can take input power from the DC input cable if their AC power fails.

The system will always use AC power if available.

See Figure 30 for an example of the G650 Media Gateway.

Figure 30: G650 Media Gateway

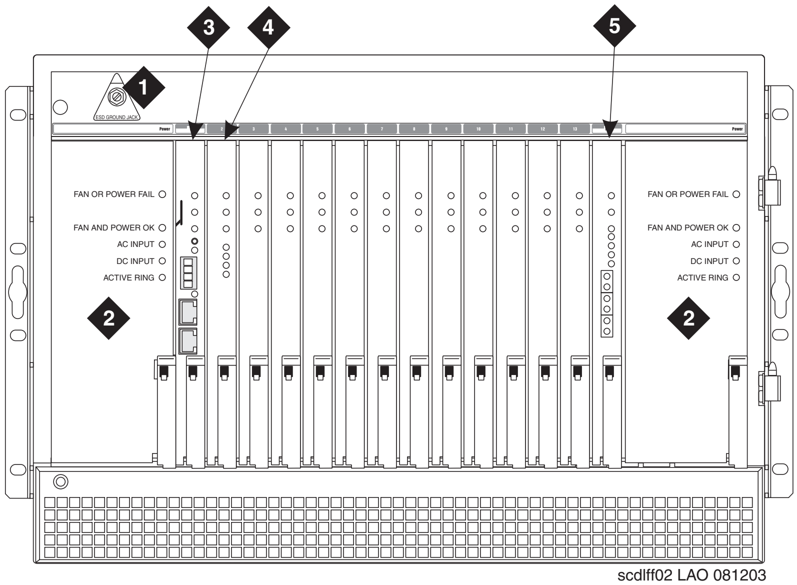


Figure notes:

Number	Description
1.	**Wrist strap for ESD (electrostatic discharge)**
2.	**655A power supply**
3.	**TN2312BP IP server interface (IPSI)**
4.	**TN799DP CLAN**
5.	**TN2302AP IP Media Processor or TN2602AP IP Media Resource 320**

Configurations

The G650 Media Gateway can be rack mounted or, in single G650 configurations, table or floor mounted. Multiple G650 Media Gateways, up to five, can be mounted in a rack and connected by TDM/LAN cables to create a G650 stack.

The G650 Media Gateway is mounted in industry standard EIA-310 19-inch (48.3 centimeters) open racks. The G650 Media Gateway provides options for front or mid mounting. Although the G650 can be mounted in a 19-inch (48.3 centimeters) four-post data rack, the G650 Media Gateway does not mount simultaneously to all four posts. When mounted in a four-post rack, the G650 Media Gateway uses the front mounting position.

Single G650

A single G650, equipped with feet, can be table or floor mounted. Side-by-side G650s, connected by TDM/LAN cables, are not supported. In a single configuration, the G650 always has an A carrier address.

Multiple G650s

Up to five G650 Media Gateways, can be mounted in a rack and connected by TDM/LAN cables to create a G650 stack. Multiple G650 Media Gateways must be vertically adjacent and their front panels must align in the same vertical plane. For example, carrier A is always below carrier B, which is always below carrier C, and so on through carrier E. Note that existing TDM/LAN cables used for the G600 cabinets are not compatible with G650.

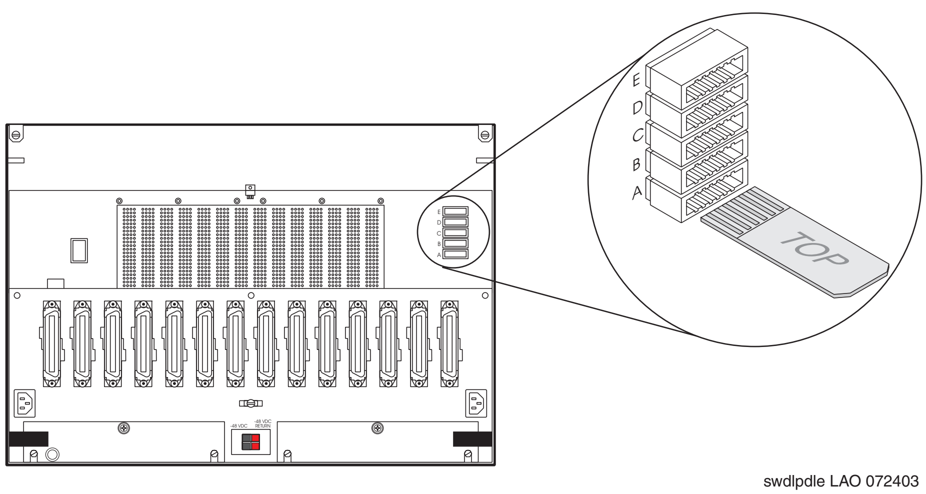
Multiple G650 Media Gateways, up to five, can be mounted in a rack but **not** connected by TDM/LAN cables. In this case, each G650 is defined as a port network. Each gateway requires its own interface hardware (EI, ATM-EI, TN2312BP IPSI). Note that with this configuration, all the G650 Media Gateways have a carrier address of A.

Carrier addressing

The carrier position, A through E, must be set on all G650 Media Gateways. The carrier address is set using a small printed circuit card that is plugged into one of five, A through E, connectors inside the carrier.

For an example and location of the printed circuit card, see Figure 31: Printed circuit card on page 124.

Figure 31: Printed circuit card



Multiple G650 Media Gateways can be rack mounted with some connected by TDM/LAN cables, and others not connected by TDM/LAN cables. For example, a customer can request that the G650 Media Gateway in the bottom of the rack not be connected to another G650. The carrier address of the G650 Media Gateway in the bottom of the rack is A. The customer can request that the next two G650 Media Gateways in the rack be connected together by a TDM/LAN cable. The carrier address of the lower of these two G650 Media Gateways is A, and the address of the upper G650 is B. And the customer can request that two additional G650 Media Gateways be placed in the rack and connected by a TDM/LAN cable. The carrier address of the lower of these two G650 Media Gateways is A and the address of the upper G650 is B. In this example, the G650s in the stack form three independent port networks:

- PN 1 has one G650 with an A carrier address
- PN 2 has two G650s with an A and B carrier address
- PN 3 has two G650s with an A and B carrier address

The carrier address of an individual rack mounted, table mounted, or floor mounted G650 is A.

See Figure 32: G650 stack on page 126 for an example of a G650 stack.

Figure 32: G650 stack

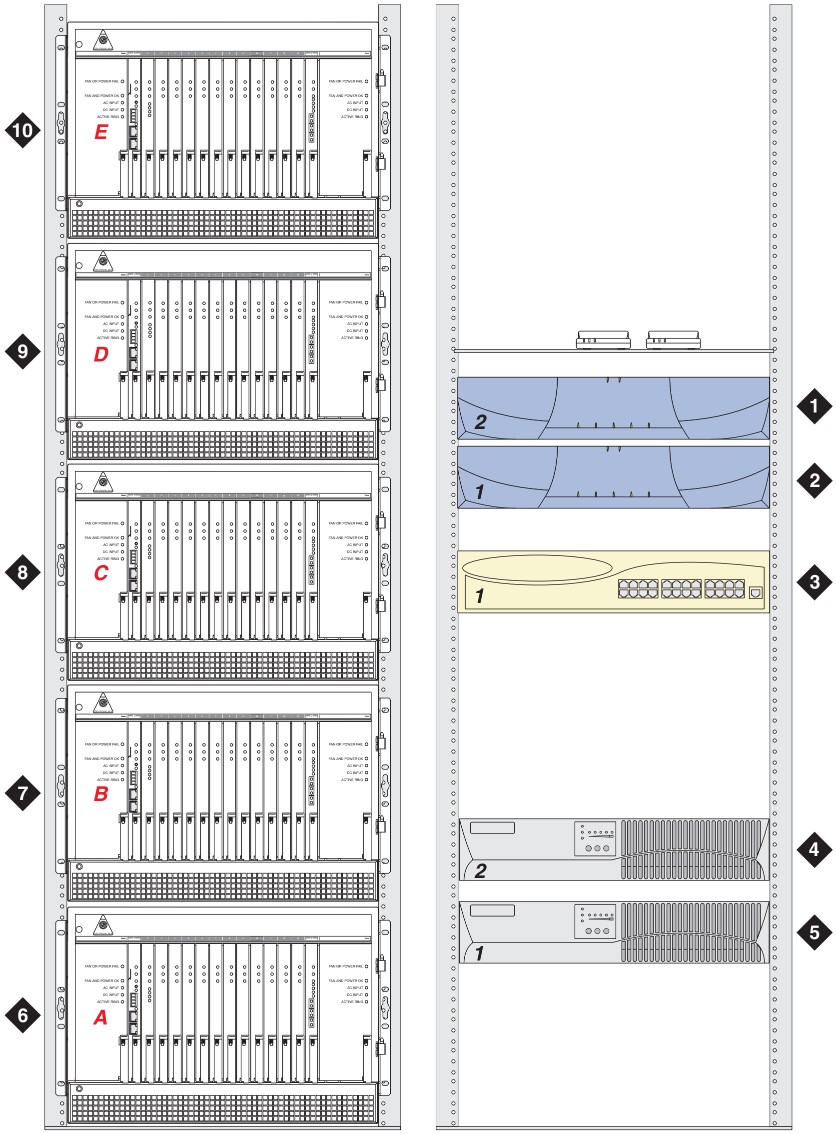


Figure notes:

Number	Description	Number	Description
1 & 2	S8800 Servers	7	G650 Media Gateway: Carrier position "B"
3	Ethernet switch	8	G650 Media Gateway: Carrier position "C"
4 & 5	UPS units: one for each server	9	G650 Media Gateway: Carrier position "D"
6	G650 Media Gateway: Carrier position "A"	10	G650 Media Gateway: Carrier position "E"

Components

Required circuit packs

The G650 Media Gateway requires the following circuit pack:

- 655A power supply

If the G650 Media Gateway or G650 stack is connected as a port network (PN) to the S8800 or S8510 Server for the control network, the following circuit pack is also required in the PN:

- TN2312BP IP server interface

If the G650 PN is connected by fiber to other PNs, one of the following circuit pack type is required:

- For more information about ATM-connect PNs, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

If the G650 PN uses the IP-PNC method for connections to other PNs, connects to branch location Media Gateways in the network, supports IP endpoints, or serves as a gateway between fiber-PNC PNs and IP-PNC PNs in the PN configuration, the following circuit packs are required:

- TN2302AP IP media processor or TN2602AP IP Media Resource 320

If the G650 PN connects to branch location Media Gateways or adjuncts in the network, supports IP endpoints, or uses IP trunks, the following circuit pack is also required:

- TN799DP control LAN (C-LAN) interface

For each physical location of a PN or group of PNs, the following circuit pack is also required:

- TN771DP maintenance and test

Optional circuit packs

Additional circuit packs can be used with the G650 Media Gateway. Their use depends on which server is using the G650 Media Gateway, the S8510 Server or the S8800 Server. See Appendix B: Optional components for servers on page 301.

I/O connections

The 14-slots of the G650 are equipped with twisted pair cables. These cables run from the backplane to the 25-pair type D, metal shelled I/O connector panel mounted on the rear of the carrier. The power supply slots (0 and 15) do not provide external I/O connections.

I/O adapters

You can use any existing adapter for input and output if the associated TN circuit pack is supported in the G650.

Fan assembly

The three-fan unit can operate at two different speeds:

- Mid speed for normal cooling
- High speed when a temperature threshold is exceeded or a fan failure is detected

Specifications

Power requirements

AC power

Commercial AC is the primary input power source. Both, slot 0 and slot 15 have dedicated AC input. The 655A power supply can operate on 90 - 264VAC AC input at 47 - 63Hz. The nominal ranges for AC power are:

- 100 - 120VAC at 50 or 60Hz
- 200 - 240VAC at 50 or 60Hz

DC power

Minus 48VDC power can be supplied simultaneously as backup power. One -48VDC power input point is provided on the G650 backplane and is distributed through the backplane to each power supply.

Power output

Power supply output voltage measurements are +5VDC, -5VDC, and -48VDC

See the following table for power source information.

Chassis style and power-distribution unit	Power source options	Power input receptacles
• AC or DC power supply. Apparatus Code 655A • A 655A power supply is required in slot 0. • A 655A power supply is optional in slot 15.	• Single phase 120 VAC with neutral wire • Single phase 240 VAC with neutral wire • -48VDC	• 120 VAC, 60 Hz NEMA 5-15R • 240 VAC, 50 Hz IEC 320 • When you install G650s in Japan, use country-specific receptacles for 100 and 200 VAC, 50/60 Hz. • When you install G650s in Mexico, use country-specific receptacles for 127 VAC.

See Table 31: Circuit breakers for AC-powered chassis on page 129 for circuit breaker information for AC-powered chassis.

Table 31: Circuit breakers for AC-powered chassis

Chassis type	Circuit breaker size
Rack mount chassis (120 VAC) 60 Hz	15 A
Rack mount chassis (240 VAC) 50 Hz	10 A

Dimensions

The G650 Media Gateway has the following dimensions:

- 14h x 17.5w x 22d (inches)
- 35.6h x 44w x 56d (centimeters)
- height in rack: 8 U
- weight: 35 - 39 pounds or 16 - 18 kilograms

The G650 requires 12-inches or 30 centimeters of clearance in the rear and 18-inches or 45 centimeters of clearance in the front. This clearance allows for adequate ventilation and conforms with standards for the EIA3 10D data rack. In a multiple G650 configuration, the G650s are placed in a rack without any space between them. If G650s are not correctly placed in the rack, the TDM/LAN cables cannot connect them.

Operating conditions

The normal operating conditions for the G650 are:

- 41° Fahrenheit (5° Celsius) to 104° Fahrenheit (40° Celsius)
- 10 percent to 90 percent relative humidity, not condensing below 10,617 feet (3,236 meters.

Avaya G700 Media Gateway

The Avaya G700 Media Gateway is an H.248 Media Gateway. The G700 Media Gateway with a server supports the entire range of adjuncts and peripheral equipment supported by Communication Manager.

Each G700 is associated with a primary call controller. The primary controller may be an S8300D, S8510, or S8800 Server. The S8300D is on a circuit pack that is always installed in slot V1 of a G700. The S8510 or S8800 Server is housed in a separate box that connects to the G700 Media Gateway over a network through a C-LAN circuit pack. Both servers can support multiple G700 Media Gateways.

The S8300D Servers can be configured as either a primary server or a Survivable Remote Server (Local Survivable Processor).

Note:

The G700 Media Gateway is no longer being sold.

Detailed description

The G700 Media Gateway is scalable and offers options. It is functional on its own or with other G700 Media Gateways. The G700 Media Gateway is also functional in a stack that is mixed with Avaya C360 devices.

A maximum of 50 G700 Media Gateways can be supported using the S8300D Server. A maximum of 250 G700 Media Gateways can be supported using the S8800 Server or the S8510 Server.

To power IP telephones without additional cables, stack the G700 Media Gateways with the Avaya C363T-PWR or C364T-PWR.

The following list describes the basic architecture of the G700 Media Gateway:

- Intel i960 controller that hosts all the base switch-control and management software.
- Fits in an EIA-310-D standard 19-inch rack.
- Supports 15 ports of tone detection.
- Contains four media module slots.
- One P330 expansion-module slot.
- One slot for the Octaplane stacking fabric.
- Can sit on a desktop or be rack-mounted.
- Contains an internal motherboard. For more information, see Motherboard on page 135.
- Standard based 10/100 Ethernet Interface connection types. A wall field or breakout panel is not required.

- Internal global AC/DC power supply that provides low-voltage DC power to the fans, motherboard, and media modules.
- Four internal fans that provide cooling for the internal components.
- An LED board that indicates system-level status.
- A serial port for command-line access.
- An eight-port layer-2 switch or two 10/100BaseT external ports.

  Note:

  An expansion module can be ordered for additional 10/100T, 100FX, ATM, or Gigabit Ethernet ports.

- A VoIP engine that supports up to 64 G.711 single-channel calls, or 32 compression codec, G.729, G.726, or G.723, TDM/IP simultaneous calls. In addition to voice calls, it supports transport of the following information:
  - Fax, Teletypewriter device (TTY), and modem calls over a corporate IP intranet using pass-through mode
  - Fax and TTY calls using proprietary relay mode

  Note:

  The path between endpoints for fax transmissions must use Avaya telecommunications and networking equipment.

  SECURITY ALERT:

  Faxes sent to nonAvaya endpoints cannot be encrypted.

  - 64kbps clear channel transport in support of BRI Secure Phone and data appliances (includes support for H.320 video over IP-connected Port Networks)
  - T.38 Fax over the Internet (including endpoints connected to nonAvaya systems)
  - Modem tones over a corporate IP intranet

  Note:

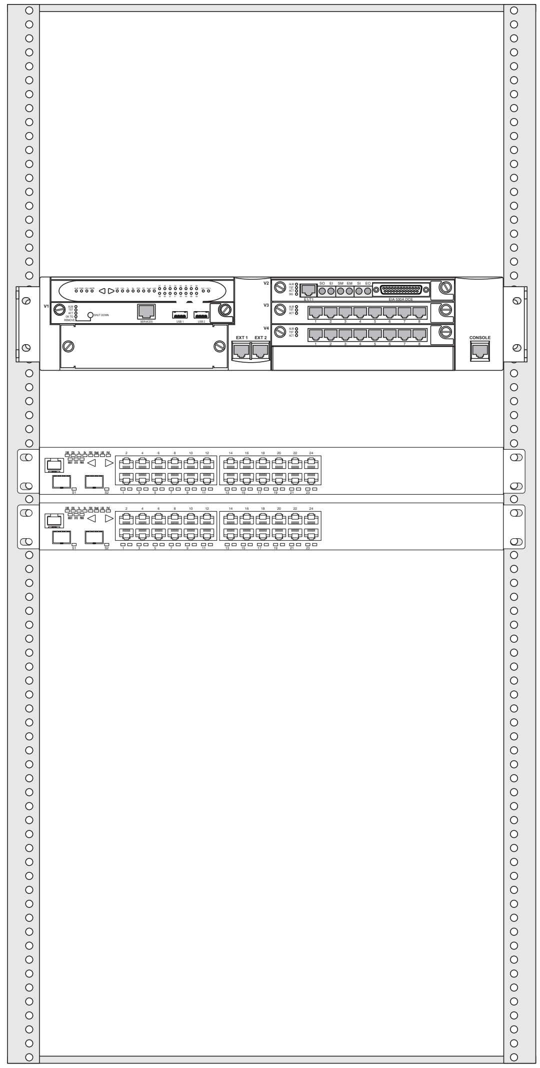
  The path between endpoints for modem tone transmissions must use Avaya telecommunications and networking equipment.

  See *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504, for more information.

The G700 Media Gateway supports SRTP media encryption.

The G700 Media Gateway has an architectural design that is similar to the Avaya stackable switching products. The following figure shows the G700 Media Gateway with two Avaya C360 switches. The G700 Media Gateway is shown at the top of the stack.

Figure 33: G700 Media Gateway with two Avaya C360 switches



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Configurations

G700 Media Gateway with an S8800 Server

The G700 Media Gateway with an S8800 Server as its primary controller connects through a LAN to a TN799DP C-LAN circuit pack. This circuit pack is mounted in a Media Gateway. This configuration is basically the same whether the G700 Media Gateway has an S8300D Server as a Survivable Remote Server or does not have a Survivable Remote Server. The following figure is an example of G700 Media Gateway connectivity.

Figure 34: The G700 Media Gateway connecting to the S8800 Server

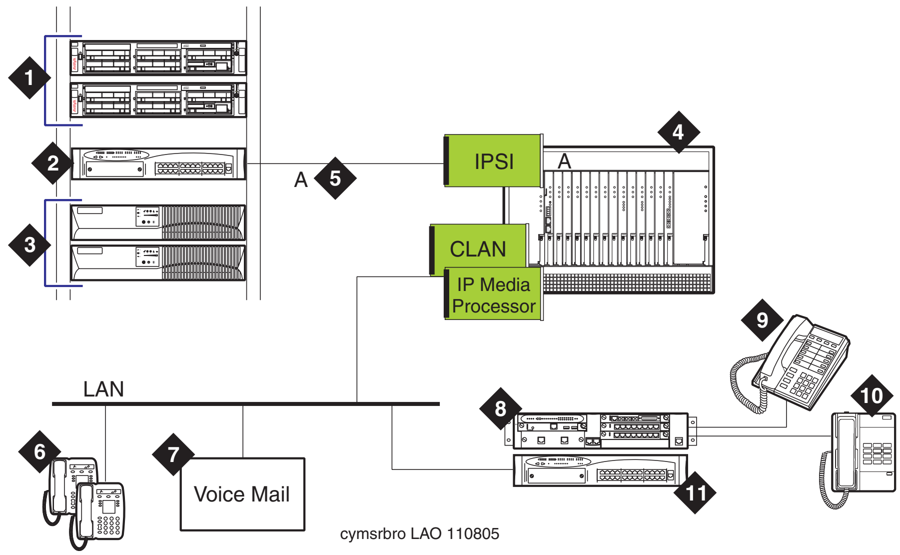

Figure notes:

Number	Description
1.	Two S8800 Servers
2.	An Ethernet switch, must be provided by Avaya
3.	Two uninterruptible power supplies (UPSs), one for each server
4.	G650 Media Gateway
5.	Dedicated LAN connectivity to the Media Gateway's IPSI circuit pack
6.	IP telephones connected through the customer's LAN
7.	Voice mail. INTUITY AUDIX is shown connected through IP
8.	A G700 Media Gateway is connected by the LAN to the C-LAN circuit pack that is located in a G650 Media Gateway. The S8300D Server in a Survivable Remote Server configuration is located in the G700 Media Gateway. In the event of a loss in communication between the S8800 Server and the G700, the Survivable Remote Server provides a backup for its registered endpoints
9.	DCP telephones — Avaya multifunction digital telephones
10.	Analog connectivity, such as analog telephones, lines, and trunks
11.	Ethernet switch (optional)

Components

Octaplane stacking fabric

"Octaplane" is a name for an Avaya hardware capability to bundle stackable components using 4-Gbps communication in each direction. This technology combines separate units into a larger logical switch using different lengths of cables. These cables connect to the expansion slots in the rear of the units. These cables are wired in a ring configuration, which provides redundancy to the stack. If a single unit fails, the stack integrity is maintained. You can remove or replace any single unit without disrupting operation or performing stack-level reconfiguration.

The following table lists the cables available to create an Octaplane stack.

Table 32: Octaplane cabling

Cable	Description and function	Length	Length (metric)
X330SC short	A light-colored cable used to connect adjacent switches	12-inches	30 cm
X330LC long	A light-colored cable used to connect switches from two different physical stacks	6 feet	2 m
X330RC redundant	A black cable used to connect the top and the bottom switches of a stack.	6 feet	2 m
X330L-LC extra long	A light-colored cable used to connect switches from two different physical stacks	24 feet	8 m
X330L-RC long redundant	A black cable used to connect the top and the bottom switches of a stack	24 feet	8 m

Power supply

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

Motherboard

The motherboard resides in the G700 Media Gateway and controls the following elements:

- The VoIP Engine, which supports up to 64 channels. If more than 64 channels are needed, a VoIP media module is required. The VoIP Engine performs the following functions:
  - IP/UDP/RTP processing
  - Echo cancellation

- G.711 A-/μ-Law
- G.729, G.726, and G723.1 encode/decode
- T.38 and Avaya Proprietary FAX relay
- FAX pass-through
- Modem pass-through
- Modem relay
- Clear channel
- Teletypewriter device (TTY) tone relay
- Silence suppression
- Jitter buffer management
- Packet loss concealment
- Avaya Encryption Algorithm (AEA) and Advanced Encryption Standard (AES) encryption of VoIP audio
- Packet reorder

- The gateway processor complex controls all the resources that are 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 Avaya P330 data-switch architecture. This complex provides an 8-port Layer-2 switch function and manages the Expansion and Cascade modules.
- The electrical connectivity and the physical connectivity for the four media module slots.

Note:

The motherboard cannot be replaced in the field.

For more information about the VoIP Media Module, see MM760 VoIP Media Module on page 228.

Fans

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

LEDs

The G700 Media Gateway uses two types of LEDs:

- Media module

- System level

Although some media modules have additional LEDs, a standard 3-LED pattern on each of their faceplates indicates the following conditions:

- Red – Fault condition

  This LED also lights when the media module is physically inserted and turns off when the board initializes.

- Green – Test condition
- Yellow – In-use condition

See the following figure for the LEDs on the media module.

Figure 35: Media modules LEDs

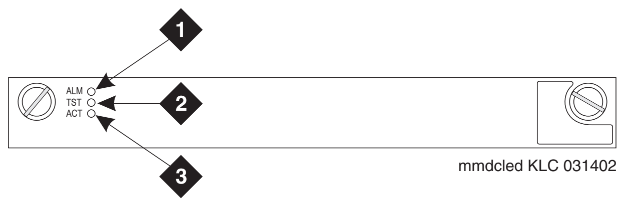

Figure notes:

1. **ALM – Alarm LED**
2. **TST – Test LED**
3. **ACT – Active LED**

Media module LEDs

Media module LEDs have the following characteristics:

- Each media module has at least three LEDs to indicate module and port status or maintenance and administration modes.
- The location, spacing, and labeling is fixed for all LEDs on all media modules.
- The LEDs are mounted on the media module's printed wiring board and placed so the LEDs show through an opening.

System-level LEDs

The system-level LED board:

- Provides visual indication of both system and Ethernet-port status and allow customers to change between these status-indication modes.

- Resides in the upper-left front of the G700 Media Gateway. The LEDs themselves reside in the board's oblong fascia panel.

You must insert or remove the LED board when you insert or remove the S8300D Server.

The LED panel is not the same size as a standard media module. You cannot insert a media module into the LED board slot or vice versa.

Specifications

The following table lists environmental considerations for the G700 Media Gateway.

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Consideration	Description
Heat dissipation	The G700 Media Gateway uses global AC, 100 VAC to 240 VAC, 50/60 Hz, 1.5 to 4.9 A, which translates to 360 to 400 Watts. However, some heat is passed out the front, by -48 VDC (up to 32 ports at 1.5 watts each for a total of 48 watts).
Altitude	Functions at altitudes of minus 197 feet (60 meters) to 10,000 feet (3,048 meters).
Air pressure	Air pressure is not specified.
Temperature and humidity	Long term operation at 41 °F (+5 °C) to 104 °F (+40 °C) at 5% to 85% humidity. Short term operation at 23 °F (-5 °C) to 122 °F (50 °C), at 5% to 90% humidity, noncondensing.
Air purity	Requires an indoor environment that is suitable for continuous human occupancy.
Lightning	The user is protected under the UL codes against overvoltage in the system. However, the system itself is susceptible to overvoltage, such as lightning, depending on the configuration. The loss of service because of an overvoltage condition can result in the loss of one or more of the following elements: • Terminal loss • Port loss • Media Module loss • Power supply within the G700
Acoustic noise generated	50 dBA maximum
Electromagnetic compatibility standards	Conforms to the electromagnetic compatibility standards for the countries in which it operates.
European Union standards	Approved to Safety Standard EN60950.

Consideration	Description
Air flow with a single fan failure	In front of the backplane, airflow is 264 linear feet per minute average. If a fan fails in front of the backplane, airflow becomes 174 lfpm average, with a range from 42 to 340 lfpm.
Air flow with the power supply fan failure	Minimal air flow at power supply if power supply fan fails.

Power 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). You can use the power supply in single or multiple G700 Media Gateways. 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 that are sized and scalable for the load. The Avaya Ethernet switches have a power unit that meets the 802.3 AF standard and provides remote power for the telephone. The power supply meets all applicable global standards for safety, immunity, and emissions and is verified by in-country testing.

Thermal protection

Thermal protection shuts down the power supply if the internal temperature exceeds the maximum rated safe operating temperature. The minimum thermal shutdown point is at an ambient temperature of 122 °F (50° C) at 10,000 feet (3,048 meters) altitude or 140 °F (60° C) at sea level. These temperature minimums are constant under all input and load conditions. You must consider the effects of component tolerances when you define the shutdown point. This consideration ensures that the supply does not shut down at ambient temperatures that are less than those previously specified. This ambient temperature is measured with a forced air flow from input to output at a nominal rate of 46 cubic feet (1.3 cubic meters) per minute (CFM) or 300 linear feet (91.4 linear meters) per minute (LFM).

Manual reset

The power supply requires a manual reset after the power supply shuts down because of overvoltage or overheating. To reset the power supply, recycle the AC input power.

AC and load center circuit breakers

For AC power, each of the G700 Media Gateways has a detachable AC power cord. This cord plugs into a wall socket or into a power strip on the rack. A circuit breaker for the panel that serves the outlet protects this circuit.

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 that protect the power feeds to the G700 Media Gateways as required by electrical codes.

AC power distribution

AC power distribution is plugged into an outlet or a power strip and can be backed up by an optional uninterruptible power supply (UPS).

AC grounding

The G700 Media Gateway contains a grounding screw on the back of the chassis. You must maintain ground connection whether you connect 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.

Related Hardware and adjuncts

Expansion modules

The G700 Media Gateway is architecturally based on the Avaya P330 and C360 switches. Therefore, customers can use selected P330 expansion modules with the G700 Media Gateway. The P330 local-area network (LAN) and wide-area network (WAN) expansion modules connect directly to the G700 Media Gateway without requiring additional hardware. Two types of expansion modules are available from Avaya:

- X330 WAN Access routing modules
- P330 LAN expansion modules

X330 WAN Access routing module

Customers with multiple branch offices need network solutions that are simple, flexible, and scalable. The Avaya X330 WAN Access routing module allows customers to deploy a unified, high-performance LAN/WAN infrastructure in one data stack.

Highlights of the Avaya X330 WAN Access Router

- Provides integrated WAN access that can be used with external firewalls or VPN Gateways
- Works with the following WAN and routing protocols
  - Point-to-Point (PPP) over channeled E1/T1
  - Frame Relay

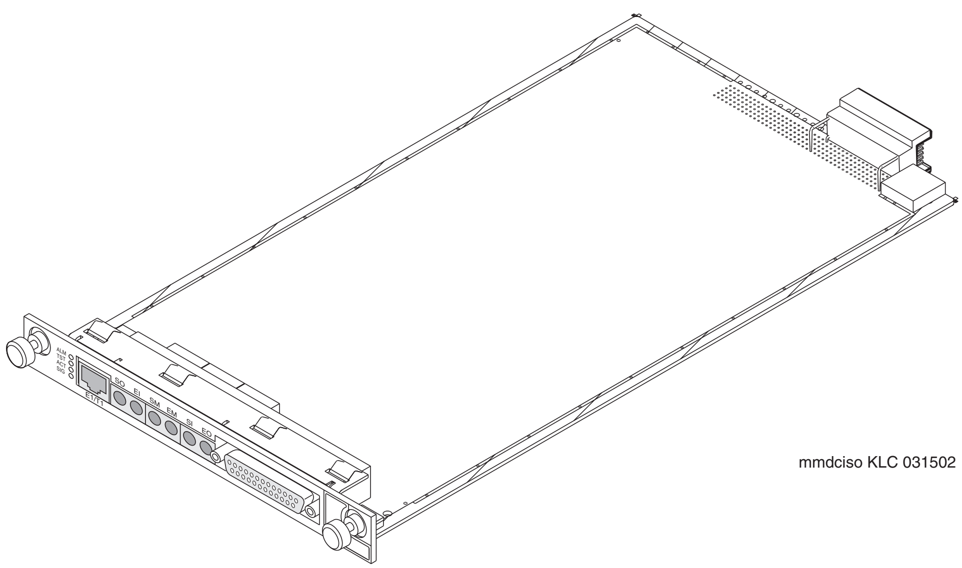
- Routing Information Protocol (RIP) v1/v2
- Single-Area Open Shortest Path First (OSPF)
- VRRP Redundancy
- Throughput: wire-speed WAN routing

Media modules

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

The media modules reside in the G700 Media Gateway and interact with the motherboard and backplane. The following figure shows a top view of a media module.

Figure 36: Top view of media module



There are nine media modules supported by the G700 Media Gateway:

- MM710 T1/E1 ISDN PRI – For information, see MM710 T1/E1 Media Module on page 212.
- MM711 Analog – For information, see MM711 Analog Media Module on page 215.
- MM712 DCP – For information, see MM712 DCP Media Module on page 218.
- MM714 Analog – For information, see MM714 Analog Media Module on page 219.
- MM716 Analog – For information, see MM716 Analog Media Module on page 222
- MM717 DCP – For information, see MM717 DCP Media Module on page 223.
- MM720 BRI – For information, see MM720 BRI Media Module on page 224.

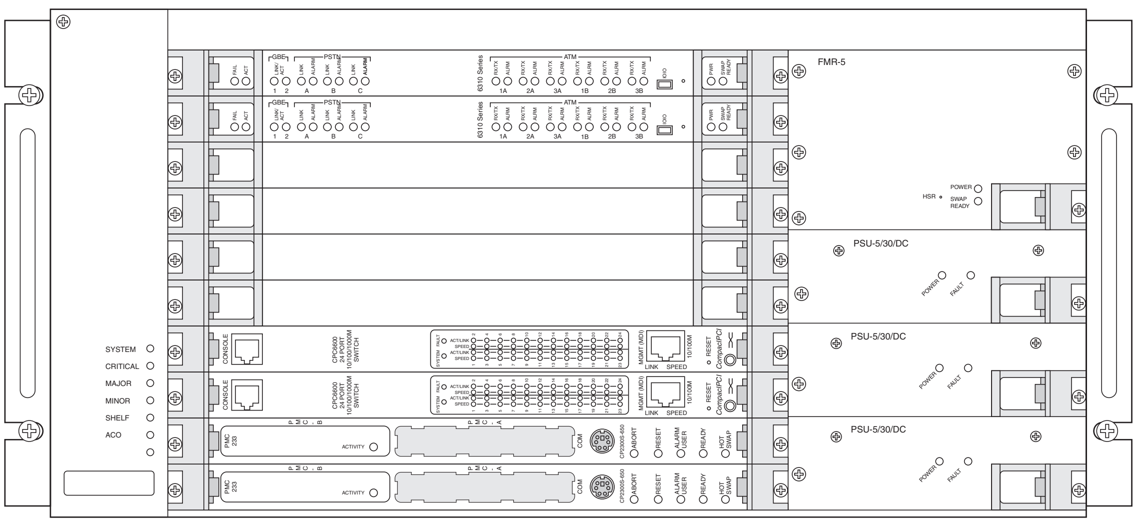
- MM722 BRI – For information, see MM722 BRI Media Module on page 225.
- MM760 VoIP – For information, see MM760 VoIP Media Module on page 228.

G860 Media Gateway

The Avaya G860 Media Gateway is a high channel density, standard compliant, VoIP Media Gateway. It provides a robust, scalable, and modular solution designed for a large campus or call center with high availability and reliability. For maximum reliability, the G860 Media Gateway features protection switching and full redundancy of all common equipment.

The G860 Media Gateway works with the duplex servers, and is supported by Communication Manager Release 4.0 and later.

Figure 37: G860 Media Gateway



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Configurations

The G860 Media Gateway chassis is only available in a redundant configuration, providing full duplication. The Trunk Processing Module can be used either in a single server configuration or in a N+1 redundant configuration.

Channels can be configured for one of the following:

- Protected: backup capability for the Media Gateway boards in which voice and signaling trunks are guaranteed constant service.
- Non-protected: no backup capability provided.

Configurations may vary according to the precise needs of the customer.

Components

Table 33: G860 Media Gateway Components

Component	Redundant configuration
Chassis	1
System Controller (SC)	2
Synchronization and Alarm Rear Transition Module (SA/RTM)	2
ES/6600 (Ethernet Switch Board - 24 Gigabit Ethernet)	2
ES/6600/RTM (Ethernet Switch 7 I/O Rear Transition Module	2
Trunk Processing Modules (TP-6310)	Up to 4
6310/RTM (TP-6310 I/O Rear Transition Module)	Up to 3
6310/RTM/HA/Redundant (TP-6310 I/O Rear Transition Module - Redundant)	1
PS/DC/5K (DC Power Supply Modules)	3
PEM/DC/5K (DC Power Entry Modules)	2
FM/5K (Fan Tray Module)	1
AF/5K (Air Filter)	1
FMR/5K (Auxiliary Fan Tray Module)	1
FPM/5K (DC Fan Tray Power Supply Module)	2
Blank panels (full configuration): Blank panel - panel only Blank panel - baffled filler panel	 1 1
Fiber cables (provided by customer) that connect to back of Trunk Processing Module	

Each G860 Media Gateway is accompanied by an accessories kit, which includes:

- RS-232 straight cable for System Controller Console Terminal (not crossed-over)
- RS-232 straight cable for Ethernet Switch Console Terminal
- CD containing system software and documentation

- CD containing optional Element Management System software

For more information, see *Installing and Operating the G860 Media Gateway*, 03-601918.

G860 Trunk Media Processing Module (TP-6310)

The G860 Trunk Processing Module (TP-6310) is a high-density, hot-swappable, compactPCI resource board with a capacity of 672 DS0 channels, supporting all necessary functions for voice, data, and fax streaming over IP networks.

Note:

The Trunk Processing Module is hot-swappable for redundant systems. However, the board must be locked in order to be replaced, which takes the board out of service.

The Trunk Processing Module provides STM-1/OC-3 (future) and T3 interfaces through its Rear Transition Module (RTM). The 6310/RTM panel contains Tx and Rx transceivers for:

- 1+1 (total 2) PSTN STM-1/OC-3 interfaces (future)
- 3 (1 active) T3 (3) PSTN interfaces (6 connectors - 3 RX and 3 TX)

The T3 PSTN interface port is an SMB connector with Tx and Rx transceivers.

The 6310/RTM is designed for protection capabilities. The 6310/RTM/HA/Redundant itself does not provide any PSTN ports. The same redundant RTM should be used for both STM-1 (future) and T3 versions.

Slots 7 to 10 are used for up to 4 Trunk Processing Modules (including the redundant TP-6310) according to customer requirements. The corresponding RTMs are located in the rear cage of the G860 in the corresponding slot. The appropriate rear RTMs are located in the rear cage of the G860 in the corresponding slot.

For redundant N:1 protection, the 6310/RTM/HA/Redundant Standby board is provided. It contains no port connections and occupies slot 10.

System controller

The system controller (SC) board controls and monitors the G860 Media Gateway operation. The SC board incorporates a 650 Mhz UltraSparc processor with 512 MB memory and uses the robust Solaris operating system environment enhanced for advanced high-availability features.

The G860 Media Gateway contains two SC boards, which are installed into their dedicated slots. Each controller contains an on-board hard disk, which stores the system controller software and configuration and performance database.

The SC board is designed according to PICMG CompactPCI standards for high-availability systems. It supports hot-swap operation, system management, and environmental monitoring. The SC board has two PCI Mezzanine Connectors (PMCs). One is occupied by the SC board with on-board hard disk and the second is reserved for future expansion of board functionality.

The two 10/100 Base-TX redundant Ethernet ports connect the SC board with the two Ethernet Switch boards through cPSB dedicated links in the midplane. The front panel PS2 COM serial port provides RS-232 console connection. The RS-232 console connection can be made through the SC front panel PS2 Com serial port or through the RS-232 serial port on the SA/RTM.

The SC board is accompanied by a Synchronization and Alarm (SA) and Rear Transition Module (RTM) board. The SA board is inserted into the midplane directly behind the main SC board and contains an RS-232 port for connecting to a console terminal.

Cooling system

The G860 Media Gateway components are cooled by a fan tray unit (FM/5K), located at the left of the card cage. An auxiliary fan tray unit (FMR/5K) is located in the top right-hand corner of the chassis, above the power supply units.

LEDs

The FM/5K fan tray unit panel contains the system's alarm indicators (LEDs) Alarm Cutoff and Reset Buttons.

The alarm indicators are connected to the fault detection and alarm system provided with the G860. As needed, LEDs indicate critical, major, or minor system faults, as well as system and shelf alarms.

Specifications

Dimensions

Table 34: G860 + TP-6310 Chassis Dimensions

Dimension	Value
Width	48.3 cm (19-inches)
Height	22.2 cm (8.75-inches)
Depth with projections	36.5 cm (13.7-inches)
Depth without projections	30 cm (11.8-inches)
Weight (fully loaded)	20.45 kg (45.1 lbs)

Power requirements

For Avaya G860 Media Gateway with Trunk Processing Module, the average power consumption for a full complement of boards is approximately 696 watts (14.5 A at 48 VDC).

Two Power Entry Modules (PEM) are provided for DC connections on the rear of the chassis. Power is required to be between -40.5 and -60 VDC. Each PEM unit contains one input terminal. Each of the DC input terminals is reverse current protected. The input terminals on each of the PEM units provide redundancy protection for the power entry circuitry.

The following are recommendations for DC power input:

- When using DC power as the primary input, ensure that the power supply complies with the safety requirements of Call Agent CAN/CSA-C22.2 No. 60950-00 and UL 60950, and EN 60950.
- For high availability, connect two separate DC power sources to avoid total power failure if one of the DC power sources fails.

Electromagnetic compatibility

The chassis is designed to comply with known EMC/RFI standards, including FCC Part 15, Class B; ICES-003, Class A; EN 55022, Class B; EN 300 386.

Compliance measures include:

- Venting holes for intake and exhaust, sized to provide for blockage of frequencies within the specified range
- Blank panels with contact fingers used for covering empty slots when a configuration requires this
- RFI filters built-in to the DC power inputs, assuring that conductive interference does not reach the power supply modules, or that switching signals generated by the power supply modules do not propagate over the main feed
- Air filters integrate a honeycomb EMI shield in its assembly. The honeycomb structure consists of "cells" that are engineered to trap and absorb EMI noise while maintaining 95% to 99% aperture for minimal airflow impedance. A gasket installed around the frame makes sure there is conductivity of the frame to the enclosure.

Environmental requirements

Table 35: Environmental requirements

Physical protection requirements	Test level
Humidity	5 to 90%
Altitude	-60 to 3048m (10,000 ft)
Drop test, packaged	Drop height: 600 mm
Drop test, unpackaged	Drop height: 75 mm
Earthquake	Zone 4
Office vibration	5-100-5 Hz/0.1g, 0.1 oct/min; 3 axes
Transportation vibration	5-100 Hz, 0.1 oct/min; 100-500 Hz, 0.25 oct/min
Thermal shock	-40 to +25 degrees C/ -40 to 77 degrees within 5 mins +70 to +25 degrees C/ -158 to 77 degrees F within 5 mins

The following summarizes environmental conditions for the G860 Media Gateway:

- Temperature
  - Extended short-term range for operation: -50C to +55 degrees C; -58 to +131 degrees F
  - Recommended ambient temperature: +5 to +40 degrees C; +40 to +104 degrees F
- Humidity
  - Relative humidity range for operation: 5 to 90%
  - Nominal relative humidity: 70% (wet bulb)

- Lightning protection

  In addition to correct earthing, sufficient lightning protection must be included at the site in order to prevent damage to the equipment. Damage can result either from a direct strike of lightning or from propagated high voltage surges.

  In order to avoid damage caused by lightning surges, installation of equipment should be compatible with Class 3 classification as defined by EN61000-4-5 Annex B, where the surge level may not exceed 2kV.
- Altitude: up to 3048m (10,000 ft)
- Earthquake: zone 4
- Rack requirements
  - Telco rack: 48.3 cm (19-inch)
  - Space: as per GR-63-CORE; maintenance access 762 mm (2 ft 6 in); wiring access 610 mm (2 ft)

Electrical aspects

The main midplane routes all signals and power to and from the plug-in boards residing in the slots, in both the front and rear sections of the chassis. Each slot is equipped with a key on the midplane to match the appropriate board type in order to prevent inserting a wrong board type into the slot.

Related hardware and adjuncts

Ethernet switch

All of the VoIP traffic (media and signaling) is routed between the gateway and the IP network through the Ethernet switch. The Media Gateway board communicates with the Ethernet switch through two redundant 100/1000 mbps cPSB links.

The SC boards communicate with the Ethernet switch through two redundant 100 mbps cPSB links. This configuration ensures redundant operation protection upon failure of any of the communication elements.

Both Ethernet switch boards are interconnected according to the PICMG 2.16 cPSB standard in a dual-star configuration, with one ES board in active mode and the other in standby mode. This configuration ensures full redundant Ethernet routes to all boards in the chassis. Failure of the active ES board automatically switches the second ES board from standby to active mode. Each of the ES boards has two fiber optic or copper Gigabit uplink interfaces for connection to the IP backbone network.

The ES/6600 board provides 24GbE ports, of which five are 1000 Base-T ports for connection to external equipment.

Power supply and power entry module

The power supply has the following features:

- DC input
- Wide range: -40.4 to -72 VDC input
- Active current load sharing on positive outputs (V1, V2 & V3)
- DC input, reverse-polarity protected
- Integral LED status indicators
- Hot-pluggable connector, with staged pin lengths
- Hot swappable
- Optimized thermal management
- No minimum load, any output
- Control and monitoring features

PS/DC/5K PEM technical specifications

- Output:
  - Output power 250 watts maximum, continuous
  - Outputs (V1-V5) +3.3 V at 40 A; +5 V at 40A, +12 V at 5.5 A; -12 V at 1.5 A
  - Temperature coefficient +/- 0.02% / degrees C
  - Controls and signaling TTL
- General characteristics:
  - Efficiency 75% at full load
  - Safety standards EN 60950, UL 1950, CSA 22.2 No. 950
- DC input:
  - PEM/DC Power Entry Module for DC
  - Input -40.5 to -60 VDC

APM/5K and FPM/5K - Advanced Fan Power Module

The Advanced Fan Power Module is the power supply for the fan tray unit. It is provided in a DC version. Two FPM/5K units are provided for redundant protection. The APM/5K and FPM/5K are not hot-swappable.

Element Management System

The Element Management System (EMS) is an advanced solution for standard-based management of Media Gateways within VoIP networks, covering all areas vital for the efficient operation, administration, management, and provisioning of the G860 Media Gateway. The EMS features a client/server architecture, enabling customers to access the EMS from multiple, remotely located work centers and workstations.

The EMS server runs on Sun Microsystems Solaris.

High-level capacities

Table 36: G860 maximum capacities for single server and redundant configurations

Capacity	Single server	Redundant
T3 links non-protected T3 links protected	12	12 12

Circuit packs, channel service units, and power supplies

120A channel service unit

The 120A channel service unit (CSU), when combined with one circuit pack, provides an integrated CSU that:

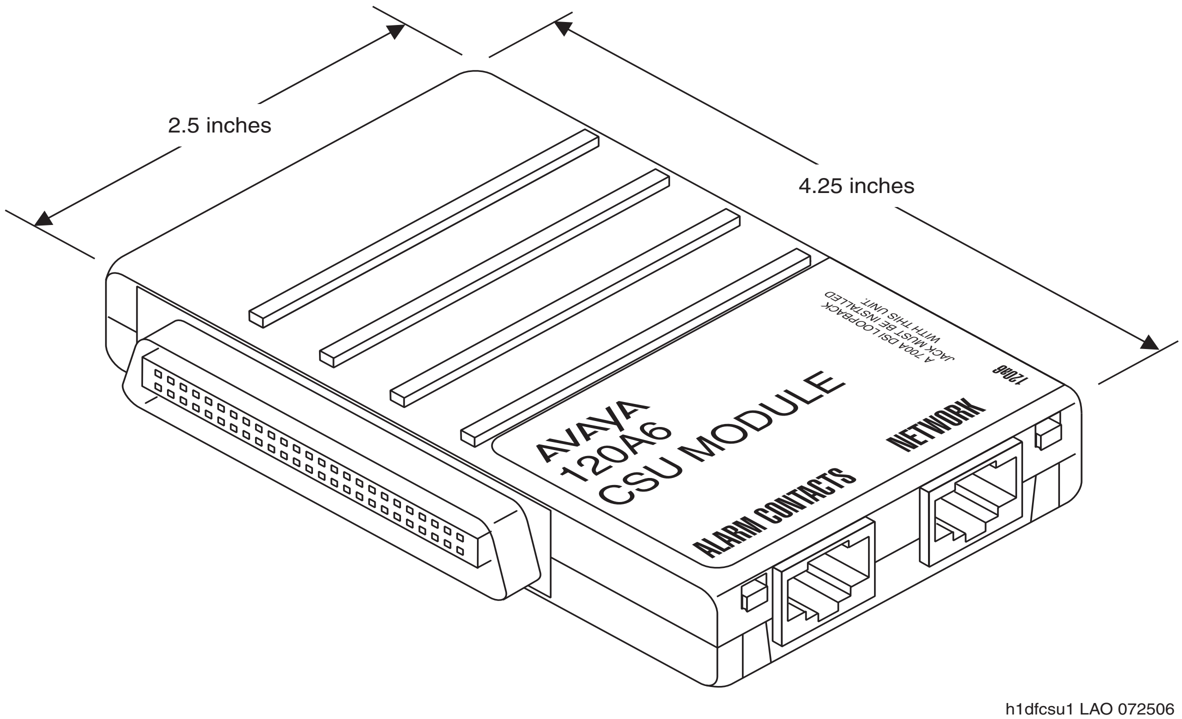
- Converts digital frames for communications between a local area network (LAN) and wide area network (WAN)
- Provides a barrier for electrical interference from either side of the unit
- Echoes loopback signals for testing the network

The 120A CSU performs similar functions to an external CSU but with the following advantages:

- highly reliable
- uses less equipment and space
- powered by the system
- easy to install and operate

The 120A CSU connects to a DS-1 circuit pack through the I/O connector panel on the back of the cabinet. A modular cable plugs into the CSU module at one end and into a 700A loopback jack, smart jack, or other service-provider interface on the other end.

Figure 38: 120A CSU



The following circuit packs support 120A CSU:

- TN464E to TN464HP
- TN2464CP and earlier
- TN767D or TN767E
- TN2313 or TN2313AP

The 120A CSU is supported on DEFINITY, Multivantage, and Communication Manager servers that support TN circuit packs.

650A AC power unit

This global power-factor-corrected supply accepts 47-Hz to 63-Hz AC input, while auto-ranging between 85 VAC and 264 VAC input. The 650A power unit provides 330 watts of total output and multiple DC outputs as follows:

- +5.1 VDC at 28 A
- –5.1 VDC at 1.0 A
- –48 VDC at 4.5 A

- +8- VDC to +14 VDC at 1.6 A (fan-speed control)

  This output (+12 VDC nominal) controls the fan speed. The voltage varies with the ambient air temperature at the inlet below the power supply. If this voltage reaches +14 VDC, the system activates a FANALM signal.
- –115 VDC to –150 VDC at 200 mA (neon bus)

The 650A power unit has three switch-selectable outputs for ringing:

- 20-Hz AC output at 85 V RMS and 80 mA, centered about –48 VDC at 180 mA
- 25-Hz AC output at 72 V RMS and 8 to 80 mA, centered about –48 VDC at 180 mA
- Two 50-Hz AC outputs at 28 V RMS, effectively 56 V, and 220 mA, biased about –48 and 0 VDC at 70 mA balanced

655A power supply

The G650 can use one or two 655A power supplies that can have both AC and DC input power present. Either power supply can provide all the power needed by the G650. When there are two power supplies, they share the power load. One power supply can operate on AC power and the other on DC power. But, if AC power is available, the system always uses AC power. The 655A power supply is:

- The only power supply supported in the G650
- Not backward compatible to other carrier types

If you use only one 655A power supply, place it in slot 0. If you are using two power supplies, place them in slots 0 and 15.

Note:

You can insert or remove a redundant power supply and not affect the G650 if the other 655A power supply is operating.

Detailed description

Input power

The 655A power supply can operate on either AC or DC input power. But, if AC power is available, the system always uses AC power. One power supply can operate on AC power, and the other on DC power. The power supplies use AC power first and change to DC power if AC power fails or is not present.

AC power

Commercial AC is the primary input power source. Both slot 0 and slot 15 have dedicated AC input. The 655A power supply can operate on AC input that ranges from 90 to 264 VAC at 47 to 63 Hz. The nominal ranges for AC power are:

- 100 to 120 VAC at 50 or 60 Hz
- 200 to 240 VAC at 50 or 60 Hz

DC power

-48VDC power can be supplied simultaneously as backup power. One -48VDC power input point is provided on G650 backplane and is distributed through the backplane to each power supply.

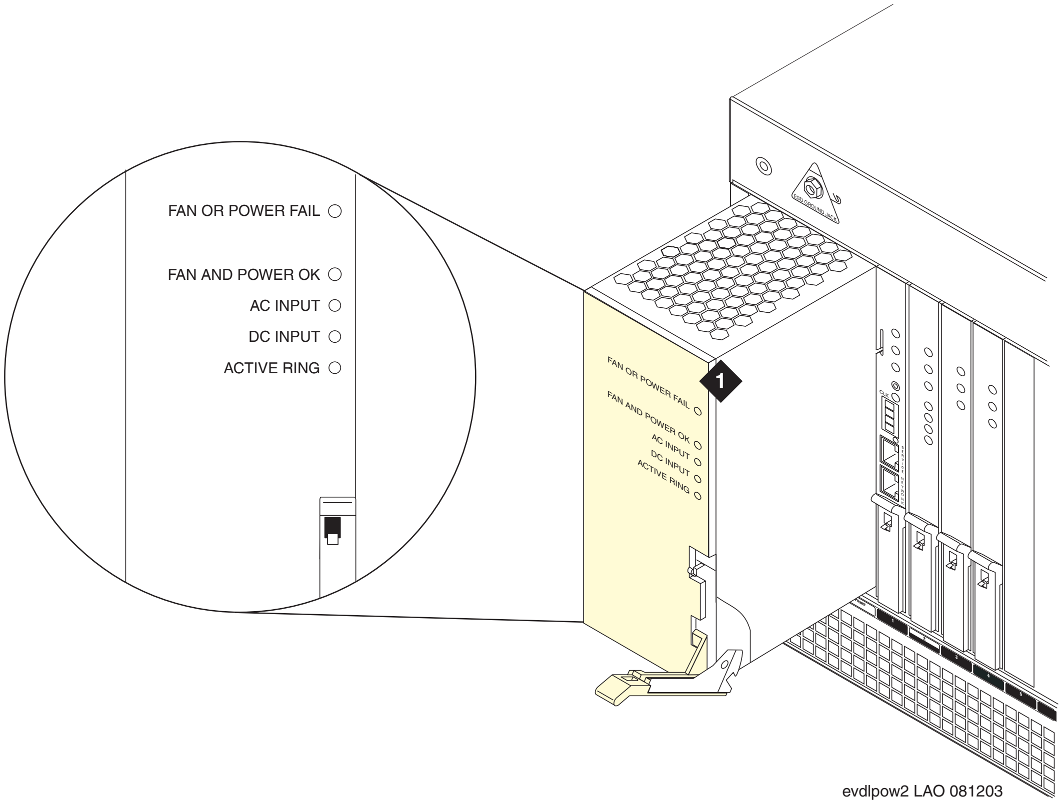
LEDs

The five LEDs on the faceplate of the 655A power supply are in a vertical line with the red LED on top. These five LEDs provide the following status:

- Red
    - Lights when there is a failure in either the power supply or the fans. For a G650 with redundant power supplies, a failure in the fan assembly lights this LED on both power supplies
    - Flashes off once per second when the software shuts down the ring voltage output of a power supply
- Yellow
    - Lights when the status of the power supply and fans is uninterrupted
    - Flashes once per second when the software shuts down a single power supply in a carrier with operational redundant power supplies
- Green
    - Lights when there is AC or DC power supply
    - Lights when the power supply causes ringing to G650

For an example of 655A faceplate LEDs, see Figure 39: 655A faceplate LEDS.

Figure 39: 655A faceplate LEDS



655A ring generation

The 655A provides either North American ringing (20Hz) or European/International (25Hz) ringing. The 655A also has a setting to provide no ringing. This setting is applicable when the customer supplies a ring generator that is external to the power supply. An example of an external ring generator is the TN2202 French ringing circuit pack.

The 655A power supply provides a physical slide switch to select the frequency of the ring generator. The options are:

- 20Hz: North American
- 25Hz: European and international
- Other: No ringing output. Applicable when an external ring generator is used such as the TN2202 French ringing circuit pack.

You must remove the power supply from G650 when you change the ringing frequency selection. The ringing frequency selection switch is on the back of the power supply.

Only one 655A supplies ringing to G650. The power supply in slot 0 in G650 with an A carrier address is the default for ringing. The system uses this default 655A unless the 655A has failed or the software forces the system to shut down. When a G650 carrier has redundant power supplies, one supply automatically supplies ringing if the other power supply fails.

A 655A provides ringing to only one G650 carrier. For example, the 655A power supplies in carrier A supply ringing to carrier A only. Meanwhile, the power supplies in carrier D supply ringing to carrier D only. If the ring generation in both of a carrier's power supplies fail, no other power supply provides ringing for the carrier.

655A replaceable DC-input fuse

The 655A provides a replaceable 25 ADC-input fuse that protects the DC input from reverse voltage on the -48VDC input. If reverse voltage is applied to G650 and 655A power supply, the 655A fuse will blow open protecting the 655A from damage.

If G650 does not operate on DC input, you need to inspect the fuse by removing the 655A power supply from G650.

The fuse is located on the rear surface of the 655A power supply. A spare fuse is also located on the rear surface.

TN429D incoming call line identification (ICLID)

The TN429 incoming call line identification (ICLID) circuit pack provides eight ports for direct inward/outward dialing (DIOD) trunks. Each port provides a 2-wire interface to the central office (CO) public exchange for incoming calls and outgoing calls. The CO provides caller names and numbers to the circuit pack. The CO displays the names and numbers on digital telephones, DCP and BRI, that are equipped with a 32-character or a 40-character alphanumeric display. In United State, the ICLID supports name and number. In Japan and other countries that comply with ICLID requirements, the ICLID displays the number only.

This ICLID is required for the Japan ANI feature where the calling number passes through to the switch. An in-band detector/converter might be required. For more information, contact your Avaya representative.

The ICLID provides the required CO disconnect functions and the interface to CAMA/E911.

TN433 speech synthesizer

The TN433 speech synthesizer for Italian provides four ports. These ports retrieve fixed messages for leave word calling, automatic wake up, and attendant console features for the visually impaired. These fixed messages include good morning, time-of-day, and extension number. Each of the ports has touchtone detection. The TN433 speech synthesizer has administrable A-Law and Mu-Law companding capabilities.

TN436B direct inward dialing trunk (8 ports)

The TN436B direct inward dialing (DID) trunk for Australia provides eight ports for DID. These ports are 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 system. The TN436B DID for Australia has administrable timers.

TN438B central office trunk (8 ports)

The TN438B CO trunk for Australia provides eight ports for loop-start CO trunks. Each of the eight ports has tip and ring signal lead. 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 (4 ports)

The TN439 tie trunk circuit pack for Australia and Japan provides four ports for 2-wire tie trunks with loop disconnect signaling. The TN439 has administrable A-Law and Mu-Law companding and administrable timers.

TN457 speech synthesizer

The TN457 speech synthesizer for British English provides four ports. These ports retrieve fixed messages for leave word calling, automatic wake up, and attendant console features for the visually impaired. These fixed messages include good morning, time-of-day, and extension number. Each of the ports has touchtone detection. The TN457 speech synthesizer has administrable A-Law and Mu-Law companding capabilities.

TN459B direct inward dialing trunk (8 ports)

The TN459B DID circuit pack for the United Kingdom provides eight ports for immediate-start or wink-start 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 system. The TN459B DID circuit pack has administrable timers and a backward busy circuit that complies with signaling requirements.

TN464HP DS-1 interface, T1 (24 channels) or E1 (32 channels)

The TN464HP circuit pack provides:

- Circuit pack-level, administrable A-Law or Mu-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 for unpolarized, balanced pairs
- Support for CO, TIE, DID, and off-premises station (OPS) port types that use any of the following protocols:
  - 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-1equipment in wideband ISDN-PRI applications
- Test-jack access to the DS-1 or E1 line and support of the 120A integrated channel-service unit (ICSU) module
- Support for the enhanced maintenance capabilities of the ICSU. These circuit packs can communicate with Avaya Interactive Response System.
- Downloadable firmware
- Support for echo cancellation

  You can select the echo cancellation capability of the TN464HP on a per-channel basis. The TN464HP DS-1 interface automatically turns off echo cancellation when the interface detects a 2100-Hz phase-reversed tone generated by high-speed modems (56-kbps). But the interface does not turn off echo cancellation when the interface detects a 2100-Hz straight tone generated by low-speed modems (9.6 kbps). Echo cancellation improves a low-speed data call.

The TN464HP DS-1 interface is intended for customers who are likely to encounter echo. This echo can be over circuits that are 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 certain local service providers. These local service providers 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 circuits and 4-wire digital circuits take place. The TN464HP DS-1 interface cancels echo with delays of up to 96 milliseconds.

TN465C central office trunk (8 ports)

The TN465C CO trunk circuit pack supports multiple countries.

This circuit pack contains, eight analog CO trunk ports, loop-start trunk signaling, 12- and 16-kHz periodic pulse metering (PPM) detection and counting, administrable timers, battery-reversed signaling, and multicountry selectable signaling.

For more information about TN465C, contact your Avaya representative.

TN479 analog line (16 ports)

The TN479 analog line circuit pack has 16 ports and supports three ringer loads and three 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.

Note:

This circuit pack is no longer sold.

The following table lists the telephones that TN479 supports and their wiring sizes and ranges.

Telephone	Wire size (metric area/diameter)	Maximum range (feet)
500-type	24 AWG (0.2 mm^2/0.5 mm)	3,000 (914 m)
2500-type	24 AWG (0.2 mm^2/0.5 mm)	3,000 (914 m)
7100-series	24 AWG (0.2 mm^2/0.5 mm)	3,000 (914 m)
7101A	not supported	not supported

Telephone	Wire size (metric area/diameter)	Maximum range (feet)
7103A	not supported	not supported
8100-series	24 AWG (0.2 mm^2/0.5 mm)	2,500 (762 m)
9100-series	24 AWG (0.2 mm^2/0.5 mm)	2,500 (762 m)

TN497 tie trunk (4 ports)

The TN497 tie trunk circuit pack for Italy has four ports for 2-wire tie trunks with loop disconnect signaling.

Note:

This circuit pack is no longer sold.

Each port can be administered for:

- A-Law or Mu-Law companding timers
- Translatore Giunzione Unscente (TGU) (outgoing tie)
- Translatore Giunzione Entrante (TGE) (incoming tie)
- Translatore Giunzione Interno (TGI) (internal tie)

TN556D ISDN-BRI 4-wire S/T-NT interface (12 ports)

The TN556D ISDN-BRI circuit pack has 12 ports that connect to ISDN-BRI terminals. Each port on a TN556 ISDN-BRI circuit pack has:

- TXT
- TXR
- PXT
- PXR signal leads

Up to eight ports can be used for Adjunct Switch Application Interface (ASAI) links. Each port operates at 192 kbps and has two B-channels and one D-channel.

The TN556D ISDN-BRI circuit pack has a maximum range of up to 1900 feet (579 meters) from the system to the telephone when the circuit pack is connected with a 24-AWG (0.20 mm^2/0.51 mm) wire. The TN556D uses standard ANSI T1.605 protocol. Up to 24 terminals can be connected, where each terminal uses one B-channel and shares the D-channel. The TN556 also has multipoint support. The capacity for the multipoint support depends on the protocol. In countries that do not support Service Profile Identifier (SPID), there is a limitation of one BRI telephone per port.

The TN556D ISDN-BRI circuit pack supports A-Law or Mu-Law companding. The TN556D ISDN-BRI circuit pack also functions as a trunk when connecting to a TE interface, such as a TN2185B in another switch. It can be used for lines and trunks simultaneously. The TN556D ISDN-BRI circuit pack 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).

TN574 DS-1 Converter — T1, 24 Channel

The TN574 is supported. TN1654 has replaced TN574.

TN725B speech synthesizer

The TN725B speech synthesizer supports English and is used in the United State.

The TN725B speech synthesizer circuit pack has four ports that send voice message information to telephones. These messages activate leave word calling, automatic wake up, voice message retrieval, and Do Not Disturb features. The ports can detect tones.

TN726B Data Line (8 ports)

The TN726B data line circuit pack has eight serial asynchronous EIA port. These ports have modem interfaces that are connected through asynchronous data units (ADUs) to EIA ports, such as RS-232, on DTE. The TN726B circuit pack uses Mode 2 or Mode 3 data transfer protocol.

Note:

This circuit pack is no longer sold.

The DTE can be adjuncts and peripheral equipment such as:

- data terminals
- printers
- host computers

- personal computers (PCs)
- graphics and fax systems
- and call detail acquisition and processing systems (CDAPSs)

With software-administered system access ports, a TN726B circuit pack connects through a cross-connect field to a TN553 packet data line circuit pack. The TN553 circuit pack then converts mode 2 protocol to mode 3 protocol. Mode 3 protocol transfers the TN726B circuit pack from the packet bus to the TDM bus for EIA connections.

Each port on a TN726B circuit pack 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 MET line circuit pack has four ports that connect to multibutton electronic telephone (MET) sets. Each port has tip and ring signals (analog voice), and digital signals to control terminals such as BT, BR, LT and LR.

Note:

This circuit pack is no longer sold.

TN744E call classifier and tone detector (8 ports)

The TN744 call classifier and tone detector circuit pack has eight ports of tone detection on the TDM bus. The TN744 circuit pack 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. The tone detectors are also used for call classifier options for various countries. The TN744 circuit pack detects special intercept tones that are used in network intercept tone detection in OCM. The TN744 circuit pack also detects tones when a central office (CO) answers a call.

The TN744 circuit pack provides tone generation and detection for R2-MFC direct inward dialing (DID) signaling. DID signaling is used in installations outside the United States. The TN744 circuit pack supports A-Law and Mu-Law companding. TN744 also allows gain or loss to be applied to pulse code modulation (PCM) signals that are received from the bus. The TN744 circuit pack detects 2025-, 2100-, or 2225-Hz modem answerback tones and provides normal broadband and wide broadband dial-tone detection.

The TN744 circuit pack 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-multifrequency code (R2-MFC), Spain MF, and Russia MF. Gain or loss and conferencing can be applied to PCM signals that are 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 circuit pack can serve as an incoming register for Russia multifrequency shuttle register signaling (MFR). Use the TN744 with the TN429C analog line CO trunk for CAMA/E911.

TN746B analog line (16 ports)

The TN746B analog line circuit pack has 16 ports. Each port supports one telephone. Supported auxiliary equipment includes:

- fax machines
- answering machines
- modems
- amplifier handsets

Note:

This circuit pack is no longer sold.

The TN746B circuit pack supports on-premises building wiring with either touchtone or rotary dialing, and with or without the LED and neon message waiting indicators. The TN746B circuit pack supports off-premises wiring with either DTMF dialing or rotary dialing. Off-premise wiring occurs out-of-building only with certified protection equipment. LED or neon message waiting indicators are not supported off-premises. The TN746B circuit pack provides -48 VDC current in the off-hook state. The ringing voltage is -90 VDC.

The TN746B, along with a TN755B neon power unit per carrier or per single-carrier cabinet, supports on-premises telephones. These telephones are equipped with neon message waiting indicators. The TN746B circuit pack supports three ringer loads. Only one telephone can have an LED or neon message waiting indicator.

TN746B supports A-Law and Mu-Law companding and administrable timers. The TN746B supports:

- Queue warning-level lights that are associated with the direct department calling (DDC) features and the uniform call distribution (UCD) features
- Recorded announcements that are associated with the Intercept Treatment feature

- PagePac paging system for the Loudspeaker Paging feature

Additional support is provided for external alerting devices. These devices are associated with the Trunk Access from Any Station (TAAS) feature, neon message waiting indicators, and modems. Secondary lightning protection is provided on the TN746B circuit pack. The TN746B circuit pack supports up to eight ports ringing simultaneously. The system can achieve the maximum of eight ports ringing simultaneously. To do so, the system uses four ports from the set of ports numbered one through eight and four ports from the set of ports numbered nine through 16.

Combined conversion of Modem Pooling requires a port for each combined resource that is to be supported. One port must be on a TN754 and another port on a TN742, TN746B or TN769 Analog circuit pack.

The following table lists the TN746B-supported telephones and their wiring sizes and ranges:

Telephone	Wire size (AWG)	Maximum range (feet)
2500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
7100 series	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
7101A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
7103A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
8100 series	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)
9100 series	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)

TN747B central office trunk (8 ports)

The TN747B CO trunk circuit pack has eight 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 circuit packs also provide battery-reversed signaling.

TN750C recorded announcement (16 channels)

Note:

The TN2501AP circuit pack replaces the TN750 circuit pack. However, the TN750 circuit pack is still supported.

The TN750 recorded announcement circuit pack records and stores announcements to be played back on demand as part of a calling feature. The TN750 circuit pack has sampling rates of 16, 32, or 64 kilobits per second (kbps). The TN750 circuit pack records announcement messages from on-premises telephones or off-premises telephones. The circuit pack can store up to 128 recorded announcements to a maximum of 8 minutes total. The TN750 circuit pack has 16 channels, and each channel can play any announcement. Up to 25 call connections can listen to each channel.

A total of 10 TN750C circuit packs in a system provides an announcement capacity of 42.6 minutes (at 32 kbps) and 160 ports. In other words, 160 announcements can play simultaneously. The compression rate, which is adequate for VDN of origin announcements, provides a total capacity of 85.3 minutes. Use of multiple TN750C circuit packs allows a more efficient method of providing many types of announcements and provides improved management of integrated announcements.

TN753B direct inward dialing trunk (8 ports)

The TN753B DID trunk circuit pack has eight ports that are used for immediate-start or wink-start direct inward dialing (DID) trunks. Each port has tip and ring signal leads. For the Slovak Republic, vintage 17 (or greater) is required. The TN753B circuit pack supports A-Law and μ-Law companding with vintage 17 or greater.

The Brazil Block Collect Call requires the TN753B circuit pack.

TN754C DCP digital line (4-wire, 8 ports)

The TN754C DCP digital line circuit pack has eight 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 circuit pack has administrable A-Law and Mu-Law companding.

The following table lists the TN754-supported equipment and shows each of their wiring sizes and ranges.

Supported equipment	Wire sizes (AWG)	Maximum range (feet)
7400 data modules	24 (0.2 mm^2/0.5 mm)	5000 (1524 m)
7400 data modules	26	4000 (1219 m)
7400 series telephones	24 (0.2 mm^2/0.5 mm)	3500 (1067 m)
7400 series telephones	26	2200 (670 m)
8400 series data modules	24 (0.2 mm^2/0.5 mm)	3500 (1067 m)
8400 series telephones	24 (0.2 mm^2/0.5 mm)	3500 (1067 m)

The TN754 circuit pack provides greater call-handling capacity for high-traffic applications and supports the group paging feature.

Combined conversion of Modem Pooling requires two ports for each combined resource that is supported. One port is on a TN754 circuit pack and another port is on a TN746B circuit pack or a TN769 analog circuit pack.

TN755B neon power unit

The TN755B circuit pack produces 150 VDC to operate neon message waiting lights on terminals that are connected to TN746B analog line circuit packs.

A TN755B circuit pack is required in G650 cabinets that support analog sets with neon message waiting.

This circuit pack and the neon message waiting function are not available on systems that use the TN2202 ring generator circuit pack for France balanced-ringing.

TN760E tie trunk (4-wire, 4 ports)

The TN760 tie trunk circuit pack has four ports. These ports are used for Type 1 or Type 5 4-wire E & M lead signaling tie trunks. Trunk types include automatic, immediate-start, wink-start, and delay-dial. Each port on a TN760 circuit pack has the following signaling leads:

- T

- R
- T1
- R1
- E
- M

The TN760 circuit pack provides release link trunks that are required for the Centralized Attendant Service (CAS) feature and has administrable A-Law and Mu-Law companding. The TN760 circuit pack supports outgoing, Multilevel Precedence and Preemption (MLPP).

Option switches on each TN760 circuit pack port can select the following connections:

- Type 1 E & M standard unprotected format
- Type 1 E & M compatible unprotected format
- Type 1 E & M compatible protected format
- Type 5 single server format

For Belgium, the Slovak Republic, the Commonwealth of Independent States, and the Netherlands, vintage 11 or greater is required.

TN762B hybrid line (8 ports)

The TN762B hybrid line circuit pack has eight ports that connect to multiappearance hybrid analog and digital telephones. The TN762B can connect to 7300-series telephones, MDC-9000 cordless telephones, and MDW-9000 cordless telephone with separate base station and charging stations.

Each port on a TN762B circuit pack has VT and VR (analog voice), CT, CR, P-, and P+ signal leads. P+ signal leads are digital signals that control terminals.

Note:
This circuit pack is not used in a G650 Media Gateway.

TN763D auxiliary trunk (4 ports)

The TN763 auxiliary trunk has four ports. Each port has the following signal leads:

- T
- R

- SZ
- SZ1
- S
- S

The TN763D circuit pack is used to access on-premises applications such as music on hold, loudspeaker paging, code calling, and recorded telephone dictation. The TN763 circuit pack supports external recorded announcement equipment and is administrable to select A-Law or µ-Law companding.

TN767E DS-1 interface, T1 (24 channels)

Note:

This circuit pack is not used in a G650 Media Gateway.

The TN767 DS-1 interface circuit pack provides a DSX1-level physical interface to the DS-1 facility. The TN767 circuit pack has unpolarized line out (LO) and line in (LI) signal lead pairs.

The TN767 circuit pack supports DS-1 rate digital facility connectivity. The circuit pack supports CO, Tie, DID, and off-premises station (OPS) port types. These port types use the robbed-bit signaling protocol. On DEFINITY CSI and SI Servers, this circuit pack supports ISDN-PRI connectivity. For these applications, the signaling D-channel can connect from the TN767 circuit pack to the processor interface by a permanent switched call over the TDM bus.

On S8510 and S8800 Servers, this circuit pack does not directly support D-channel signaling and thus does not directly support ISDN-PRI connectivity. However, the TN767 circuit can indirectly support D-channel signaling provided the central office supports nonfacility associated signaling (NFAS). In this case, you use NFAS administration on the server. This administration associates the D-channel of another T1/E1 circuit pack, usually a TN464, with the TN767 circuit pack.

The TN767 circuit pack communicates with Avaya IVR. The TN767 also provides the enhanced maintenance capabilities of the 120A channel-service unit (CSU) and the enhanced integrated channel-service unit (ICSU).

DS-1 tests include:

- loopback tests at the DS-1 circuit pack edge or the 120A (if used)
- bit error rate (BER) loopback tests at the far-end CSU
- BER 1-way DS-1 facility tests

Other tests include loopback testing specifically designed to locate DS-1 facility faults.

TN769 analog line (8 ports)

The TN769 analog line circuit pack has eight ports, each with tip and ring signal leads.

Note:
This circuit pack is no longer sold.

The TN769 circuit pack supports:

- On-premises or off-premises wiring with either touchtone or rotary dialing and with or without LED or neon message waiting indicators
- Three ringer loads, such as three telephones with one ringer load each
- Up to four simultaneous ports ringing
- Queue warning-level lights that are associated with the direct department calling (DDC) feature and uniform call distribution (UCD) feature
- Recorded announcements for intercept treatment
- Dictating machine for the Recorded Telephone Dictation Access feature
- PagePac paging system for the loudspeaker paging feature
- External alerting devices for the Trunk Access from Any Station (TAAS) feature
- Modems

The TN769 circuit pack does not support off-premises message waiting indicators.

The TN769 circuit pack provides secondary lightning protection and supports μ-Law companding.

Each carrier with neon message indicators requires the TN769 circuit pack, along with a TN755B neon power circuit pack to support neon message waiting indicators. Only one telephone can have an LED or neon message waiting indicator.

Combined conversion of Modem Pooling requires both

- a port on a TN754B circuit pack and
- a port on a TN746B circuit pack or a TN769 analog circuit pack

for each combined resource that is to be supported.

The following table lists the TN769-supported telephones and shows each of their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
2500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)

Telephone	Wire size (AWG)	Maximum range (feet)
7102 series	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
7101A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
7103A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
8100 series	24 (0.2 mm^2/0.5 mm)	10,000 (3,048 m)
9100 series	24 (0.2 mm^2/0.5 mm)	10,000 (3,048 m)

TN771DP maintenance and test

The TN771DP maintenance test circuit pack performs maintenance functions. These functions include packet bus reconfiguration. This reconfiguration 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 two devices and error and flow control on multiple logical links. LAPD swaps spare leads with the malfunctioning leads to recover packet bus failures that involve up to three malfunctioning leads. Malfunctioning leads, in this case, are one or two data or parity leads and one control lead.

Other maintenance functions include ISDN-PRI testing that originates and terminates loopback tests on ISDN facilities. The testing provides bit and block error rate information that indicates ISDN facility quality.

The TN771DP circuit pack can be updated using the firmware download feature, which requires use of the TN799 C-LAN circuit pack interface.

A TN771DP circuit pack is required for:

- Any CSI system that uses a TN2198 BRI circuit pack. Otherwise, a TN771DP circuit pack is not required. This applies to S8100 in CMC.
- In critical reliability-systems, duplicated server and duplicated port network connectivity (PNC), requires a TN771DP circuit pack in each port network. In standard or high-reliability systems, a TN771DP circuit pack is optional.
- All R system PPNs. For duplex Server, a critical reliability R system requires a TN771DP circuit pack in each PN. An R system with ATM network duplication requires a TN771DP circuit pack in each PN.
- All CSI models that use a TN2198 BRI circuit pack

A maximum of one TN771DP circuit pack is allowed in any port network.

A TN771DP circuit pack is never used with the S8100 Server.

TN775C maintenance

The TN775C circuit pack is used in maintenance to monitor power failure signals in an expansion port network (EPN) cabinet. The TN775C circuit pack also monitors the clock, monitors and controls the power supplies and battery charger, and monitors air flow and high-temperature sensors. The TN775C circuit pack provides two serial links to communicate with Expansion Interface (EI) circuit packs. The TN775C also provides an RS-232 interface for connection to an administration terminal. Each circuit pack contains a 3-position switch to control emergency power transfer.

Note:

This circuit pack is not used in a G650 Media Gateway.

The TN775C contains a DC-to-DC power converter. The TN775C is used in maintenance to monitor the processor in an EPN. A Survivable Remote Processor (SRP) supports this EPN.

TN787K multimedia interface

The TN787 multimedia interface circuit pack is used in conjunction with the TN788 multimedia voice conditioner circuit pack. The TN787 provides service circuit functionality for the Multimedia Call Handling (MMCH) feature. This feature provides both voice and multimedia data service between multimedia complex endpoints. Up to six endpoints can conference to a single multimedia call occurrence.

Note:

This circuit pack is no longer sold.

The TN787 circuit pack provides a TDM-bus interface and a DS-1 adjunct cable interface. The TN787 circuit pack routes the H.221 multimedia information to the DS-1 interface to free more TDM-bus timeslots. Freeing more timeslots allows the system to carry more audio, video, and data bit streams between multimedia complex endpoints. The TN787 circuit pack provides support for multiple port networks (PNs).

TN788C multimedia voice conditioner

The TN788C multimedia voice conditioner circuit pack is used in conjunction with the TN787F/G multimedia interface circuit pack. Together, they provide service circuit functionality for the

MMCH feature. This feature provides both voice service and multimedia data service between multimedia complex endpoints.

Note:

This circuit pack is no longer sold.

Note:

A TN788C V1 circuit pack only supports μ-Law companding. A TN788C V2 or later supports A-Law and μ-Law.

The TN788C circuit pack is the audio processor for the Px64 multimedia conference bridge. The TN788C circuit pack contains eight digital signal processors. The processors include four for encoding and four for decoding. Each encoder/decoder pair is assigned to a Px64 endpoint to process its audio channel. Connection to and from the audio of the endpoint is by way of a TN787 multimedia interface port. This connection is through the TDM-bus timeslots.

Each of the eight digital signal processors communicate with the main processor on the circuit pack through eight 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

Note:

This circuit pack is no longer sold.

The TN789B radio controller circuit pack is an interface between a switch and two Wireless Fixed Base (WFB) radio units. This interface is used for the DEFINITY Wireless Business System. The TN789B circuit pack contains a main processor to handle data line circuit (DLC) and upper medium access (MAC) layers of firmware. The TN789B circuit pack also contains two lower MAC processors, one processor for each radio interface that is referred to as 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 consists of three 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 TN789B circuit pack includes a standard TDM-bus interface from a system, two radio interfaces to two separate radio units, and two synchronization ports. In addition, two RS-232 interfaces provide for a debug terminal and for setting up the wireless terminal.

TN791 analog guest line (16 ports)

The TN791 is a 16-port analog guest line circuit pack. The TN791 is used for international offers and for offer category B in the United State and Canada. Each of the 16 ports supports one telephone, such as 500 (rotary dial) and 2500 terminals (DTMF dial). The ports also support LED and neon message waiting indicators. A separate power supply is required for neon message indicators.

The TN791 circuit pack supports on-premises wiring with either touchtone or rotary dialing and with or without the LED and neon message waiting indicators.

The TN791 circuit pack supports three ringer loads. Only one telephone can have an LED or neon message waiting indicator. The TN791 supports up to eight ports ringing simultaneously. To achieve this maximum, the system uses four ports from the set of ports numbered one through eight and four ports from the set of ports numbered 9 through 16.

The TN791 circuit pack supports A-Law and µ-law companding and administrable timers. Secondary lightning protection is provided.

The following table lists the TN791-supported telephones and shows each of their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
2500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
6200 type	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)
7100 series	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
7101A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
7103A	24 (0.2 mm^2/0.5 mm)	15,200 (4,633 m)
8100 series	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)
9100 series	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)

TN792 duplication interface

In a high reliability or critical reliability DEFINITY SI system, a duplication interface copies the contents of the memory from the primary Server to a standby Server. Therefore, the standby Server can take over immediately when the former fails. The TN792 duplication interface (DUPINT) uses the Enhanced M-Bus of the DEFINITY SI TN2404 processor for this memory shadowing function. The Enhanced M-bus supports a 32-bit addressing and data access (versus 16-bit for the M-bus). In this case, the Enhanced M-bus transfers data faster and shadows a larger area of memory than the M-bus. The M-bus is still supported.

Note:

This circuit pack is no longer sold.

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 circuit pack cannot communicate with a TN792 circuit pack.

A duplicated server optical cable connects the TN792 circuit packs. This cable eliminates the additional electromagnetic emissions that 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 circuit pack is compatible with the existing duplication cables.

TN793CP analog line with Caller ID for multiple countries (24 ports)

The TN793CP is an analog line, 24-port circuit pack that supports caller ID telephones and caller ID devices that conform to Bellcore Standard GR-30-CORE, Issue 2, and Bellcore-compliant signaling using V.23 Frequency Signal Keying (FSK). This means that the TN793CP supports caller ID devices in the United States and most other countries. Each port can support one of the following:

- Analog telephone, such as a 2500 telephone (DTMF dial)
- Answering machine
- FAX
- Loop-start CO port (used for Communication Manager Messaging)

The TN793CP provides:

- Touchtone or rotary dialing
- Rotary digit 1 recall
- Ground-key recall
- Programmable flash timing
- Selectable ringing patterns
- On-premises LED and neon message waiting
- Caller ID with Call Waiting
- Secondary lightning protection

WARNING:

The TN793CP does *not* support the telephones (used primarily in France) that use 50 Hz balanced ringing.

The TN793CP supports on-premises (in-building) wiring. The TN793CP circuit pack supports off-premises wiring with either DTMF or rotary dialing, but LED or neon message waiting indicators are not supported off-premises.

The TN793CP circuit pack, along with a TN755B neon power circuit pack supports on-premise telephones that are equipped with neon message waiting indicators. The TN793CP supports three ringer loads. Only one telephone can have an LED or neon message waiting indicator. A maximum of 12 ports can be rung simultaneously. To achieve this maximum, the system uses four ports from the set of ports numbered one through eight, four ports from the set of ports numbered 9 through 16, and four ports from the set of ports numbered 17 through 24.

The TN793CP circuit pack supports A-Law and μ-law companding and administrable timers. The TN793 circuit pack supports queue warning level lights. These lights are associated with the direct department calling (DDC) and the uniform call distribution (UCD) features, recorded announcements that are associated with the Intercept Treatment feature, and PagePac paging system for the Loudspeaker Paging feature. Additional support is provided for external alerting devices. These devices are associated with the Trunk Access from Any Station (TAAS) feature, neon message waiting indicators, and modems. The TN793CP provides -48 VDC current in the off-hook state. Ringing voltage is -90 VDC.

The TN793CP supports DTMF sending levels that are appropriate for Avaya Interactive Response.

The multinational support of the TN793CP circuit pack is identical to that of the TN2215 circuit pack. Therefore, the TN793CP allows country-specific transmission selection. The TN793CP is also impedance and gain selectable for multiple countries. For more information, contact your Avaya representative.

The following table lists the TN793CP-supported telephones and shows each of their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
2500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
6200 type	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)
7100 series (no longer sold)	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
8100 series (no longer sold)	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)
9100 series (no longer sold)	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)

TN797 analog trunk or line circuit pack (8 ports)

The TN797 circuit pack provides a combination 8-port analog trunk and line circuit pack for the United State, Canada, and other countries that have the same analog standards. The TN797 circuit pack provides you with the capability to administer any of the eight ports as any of the following trunks or lines:

- loop start or ground start CO trunk
- CAMA E911 trunk
- wink-start or immediate-start DID trunk
- on-premises or an off-premises analog line with or without LED Message Waiting Indication

The TN797 does not support incoming caller ID (ICLID) on the analog trunk to the CO. The TN797 does not support caller ID on the line side to the telephone.

TN799DP control LAN (C-LAN) interface

The TN799DP control LAN (C-LAN) interface provides TCP/IP connectivity over Ethernet or Point to Point Protocol (PPP) to adjuncts the following:

- Avaya Call Management System (CMS)
- INTUITY AUDIX
- Distributed Communication System (DCS)
- printers
- call detail recording (CDR)
- property management systems (PMS)

The C-LAN operates at 10 or 100 mbps and full duplicated server or half duplicated server, both of which are administrable. The C-LAN provides connectionless UDP sockets for IP solutions support. The C-LAN also supports 500 remote sockets, with support for 4-KB UDP sockets. The C-LAN supports variable length ping and the traceroute and netstat network testing commands.

The C-LAN circuit pack provides call control for all IP endpoints that are connected to the S8800 Server using the G600 Media Gateway or G650 Media Gateway. You can use a maximum number of 64 C-LAN circuit packs for each configuration. The number of required C-LAN circuit packs depends on the number of devices that are connected. The C-LAN number also depends on which options that the endpoints use. It might be advantageous to segregate IP voice control traffic from device control traffic as a safety measure.

A CLAN socket is a software object that can connect a C-LAN to the IP Network. A simple calculation determines the default value for C-LAN socket usage of H.323 tie trunks. Divide the total number of H.323 tie trunks that use sharing by 31. Each IP endpoint requires the use of some number of C-LAN sockets. A C-LAN circuit pack supports a maximum of 500 sockets.

The C-LAN differs from an IP Media Processor. The difference is that the C-LAN controls the call, while TN2302AP provides the codecs that are used for the audio on the call.

To keep the firmware on the CLAN circuit pack up-to-date, you can download C-LAN firmware updates from the Web. To take advantage of this downloadable firmware capability, you must already have at least one C-LAN circuit pack in your system. You must also have access to the public Internet. The C-LAN can serve as an FTP or SFTP server for file transfers — primarily firmware downloads. The C-LAN cannot serve as an SFTP client.

With Communication Manager Release 3.1 and later, the C-LAN can also receive firmware downloads from a central firmware depository on an SCP-enabled file server.

For more information on firmware downloads, and instructions for downloading, see http://www.avaya.com/support/.

TN801B MAPD (LAN gateway interface)

The TN801 LAN gateway interface is part of the Multiapplication Platform DEFINITY (MAPD). With TN801, you can perform direct integration of PC-based applications into the switch. The TN801 circuit pack works as the interface for solutions such as Computer Telephony Integration (CTI) and Adjunct-Switch Application Interface (ASAI). The TN801 circuit pack provides:

- packet bus and TDM-bus interfacing
- physical mounting for a CPU
- external interfaces
- mapping of circuit-switched connections between the TDM bus and the expansion circuit pack

TN802B MAPD (IP interface assembly)

The TN802 IP interface circuit pack supports voice calls and fax calls from the switch across a corporate intranet or the Internet. This circuit pack is still supported, but is now replaced with the TN2302AP IP media processor. The IP trunking software runs on an embedded PC that runs Windows NT. The TN802 circuit pack supports IP Solutions, including IP trunking and MedPro (H.323) with IP softphones.

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 the TN802 in MedPro mode, you activate it through administration to use the H.323 trunking feature. MedPro mode is necessary to support IP softphones.

TN1654 DS-1 converter, T1 (24 channels) and E1 (32 channels)

The TN1654 converter installs in place of the conventional fiber. The TN1654 converter supports from one to four T1 or E1 facilities. The TN1654 also provides a total of 92 T1 channels or 120 E1 channels. These channels run in each direction between the processor port network (PPN) and an expansion port network (EPN). This capacity is enough for the EPN to easily support several hundred stations.

The switch architecture provides for EPNs that are remotely located from the PPN. An EPN that is within 5 miles (8 kilometers) of the PPN can be coupled using a multimode fiber-optic cable. EPNs that are within 22 miles (35.4 kilometers) of the PPN can be coupled using single-mode fiber-optic cable. You must use a DS-1 converter complex to connect an EPN when the distance between the PPN and the EPN exceeds a certain distance or private right-of-way is unavailable. The maximum distance is 5 miles (8 kilometers) for a multimode cable or 22 miles (35.4 km) for a single-mode cable. One DS-1 circuit pack is placed on each end of the DS-1 converter complex.

The TN1654 DS-1 converter requires a set of Y-cables to connect to a TN570B Expansion Interface circuit pack.

TN2138 central office trunk (8 ports)

The TN2138 central office (CO) trunk circuit pack provides eight analog loop start **CO** trunk ports for Italy. Each port has a tip and ring signal lead. The TN2138 has 50-Hz, 12-kHz, and 16-kHz periodic pulse metering (PPM).

Note:

This circuit pack is no longer sold.

TN2139 direct inward dialing trunk(8 ports)

Note:

This circuit pack is no longer sold.

The TN2139 direct inward dialing (DID) trunk for Italy provides eight analog DID trunk ports for analog DID signaling. Each of the eight ports has a tip and ring signal lead.

TN2140B tie trunk (4-wire, 4 ports)

The TN2140B tie trunk is used in Hungary and Italy. The TN2140B provides four ports for 4-wire E&M lead signaling tie trunks. The TN2140 provides continuous E&M signaling and discontinuous E&M signaling. The TN2140 also provides administrable A-Law and μ-Law companding and standard Type 1 and Type 5 signaling. The TN2140B is required for Hungary.

TN2146 direct inward dialing trunk (8 ports)

The TN2146 provides eight analog DID trunk ports for Belgium and the Netherlands. Each of the eight ports has tip and ring signal lead. The TN2146 uses four Dual Subscriber Line Audio processing Circuits (DSLACs). One DSLAC is used for each pair of ports. The circuits are administered to meet trunk transmission characteristics. You can set the DSLACs 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 digital TDM bus on the system. The TN2146 circuit pack provides either A-Law or Mu-Law companding.

Note:

This circuit pack is not used in a G650 Media Gateway.

TN2147C central office trunk (8 ports)

The TN2147 has eight analog central office (CO) trunk ports. Each port has tip and ring signal leads. The TN2147 uses four (one for each pair of ports) DSLACs. These DSLACs are administered to meet a given transmission and impedance requirement. The DSLACs convert analog signals to digital signals and digital signals to analog signals. These conversions interface the analog CO trunks to the system's digital TDM bus of the system.

The TN2147C provides multicountry signaling based on a trunk type of loop-start, ground-start, or battery reverse loop-start.

TN2181 DCP digital line (2-wire, 16 ports)

The TN2181 circuit pack has 16 DCP ports. These ports can connect to 2-wire terminals such as the 6400-series, 8400-series, 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 meters).

The TN2181 circuit pack supports either A-Law or μ-Law companding. The TN2181 also supports the 8400-series data modules.

Note:
This circuit pack is no longer sold.

TN2182C tone clock, tone detector, and call classifier (8 ports)

The TN2182 tone clock integrates the following functions onto one circuit pack for all system reliability configurations:

- tone generator
- tone detection-call classifier
- system clock
- synchronization

Note:
This circuit pack is not used in a G650 Media Gateway.

The TN2182 supports eight ports for tone detection and allows gain or loss applied to PCM signals received from the bus. The TN2182 supports:

- stratum-4 enhanced clock accuracy
- MFC signaling, such as Russia MF
- Russia multifrequency shuttle register signaling (MFR)

- A-Law and Mu-Law companding

The TN2182CP performs the following functions:

- 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
- provides normal and wide broadband dial-tone detection

In most configurations, the 2-circuit or 3-circuit pack combination can include either the tone generator, tone detector, and call classifier. This combination can be replaced with this one circuit pack to free one or two port slots.

Use the TN2182CP circuit pack with the TN429D analog line central office trunk for CAMA/E911 and incoming caller ID (ICLID). A TN2182 is required for main processor 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 Offer B only for US and Canada) on page 188.

TN2184 DIOD trunk (4 ports)

TN2184 is a Direct Inward/Outward Dialing (DIOD) trunk circuit pack used for Germany. The TN2184 circuit pack contains four port circuits. Each circuit interfaces with a 2-wire analog CO trunk with the TDM switching network of the system. Each port allows incoming calls and outgoing calls to include addressing information. The ports receive this information from the CO for incoming calls and send it to the CO for outgoing calls. The TN2184 detects periodic pulse metering (PPM) signals for call-charge accounting on outgoing calls.

The TN2184 combines the features of a CO trunk and a DID trunk. The TN2184 provides both outgoing calls and incoming calls with addressing information in both directions.

Note:

This circuit pack is not used in a G650 Media Gateway.

Note:

This circuit pack is no longer sold.

TN2185B ISDN-BRI S/T-TE interface (4-wire, 8 ports)

The TN2185B supports eight 4-wire ISDN-BRI line S interfaces. Each interface operates at 192 kbps, with two B channels (64 kbps) and one D-channel (16 kbps). The TN2185B interfaces with the LAN bus and the TDM bus to provide the TE side of the BRI interface. The TN2185B is similar to the TN2198 except that the TN2185B is a 4-wire S-interfaces instead of a 2-wire U-interface.

For each port, information communicates over two 64-kbps bearer channels called B1 and B2. Information also communicates 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 can be packet-switched, but not both at once. The D-channel is always packet-switched. For voice operation, the circuit pack has a Mu-Law or A-Law option that applies uniformly to all circuit-switched connections on the circuit pack. The circuit-switched connections 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-kbps wideband channel.

The TN2185B has a maximum range up to 18,000 feet (5486 meters) from the system to the NT1 device. In an environment with multiple telephones, the B-channels are shared only on a per-call basis. For example, if Channel B2 is for data, then the use of this channel by one telephone excludes the others from having access to Channel B2. When a device communicates over the D-channel to access B1 or B2, that channel is owned until the call is taken down. The D-channel is always shared among the terminals. The TN2185B circuit pack can be used as an alternative to the TN464 circuit pack or the TN2464 circuit pack.

The TN2185B supports the ability to outpulse in-band DTMF signals or end-to-end signaling.

TN2185B supports QSIG Call Completion but not QSIG Supplementary Services. You can use ISDN-BRI trunks as inter-PBX tie lines that use the QSIG peer protocol.

TN2198 ISDN-BRI U interface (2-wire, 12 ports)

The TN2198 circuit pack is used to connect 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 can connect to one or two telephones. Unlike the TN2185 circuit pack, the TN2198 does not provide a trunk-side interface.

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. Information also communicates 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 one telephone, such as the 500 rotary dial analog telephone and 2500 DTMF dial telephones.

The D-channel supports the LAPD protocol and is consistent with the CCITT Q.920 recommendations for D-channel signaling.

In an environment with multiple telephones, 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 one telephone excludes the other telephones from having access to the B2 channel. When a device communicates over the D-channel to access B1 or B2, that channel is owned until the call is taken. The D-channel is always shared among the telephones. TN2198 interfaces with the TDM bus and the packet bus in the switch backplane and terminates with 12 ISDN basic access ports.

The TN2198 has a maximum range of up to 18,000 feet (5486 meters) from the system to the NT1 device and uses standard protocol ANSI T1.601. The TN2198 has a 160-kbps line rate that consists of:

- Two bearer channels at 64 kbps each
- A D-channel at 16 kbps
- Framing at 12 kbps
- Maintenance at 4 kbps

The TN2198 supports a maximum of 24 telephones or data modules.

The TN2198 is not offered as a BRI Tie Trunk.

TN2199 central office trunk (3-wire, 4 ports)

The TN2199 central office (CO) trunk circuit pack is designed for use in Russia.

The TN2199 is a 4-port, 3-wire, loop-start trunk circuit pack that can be used as a:

- DID trunk
- Two-way or one-way incoming or one-way outgoing CO trunk

The TN2199 combines the functionality of a DID trunk and a one-way outgoing CO trunk (DIOD trunk). To accomplish MF shuttle signaling, the TN2199 circuit pack must be combined with a TN744D Call Classifier circuit pack.

The TN2199 circuit pack supports incoming automatic number identification (ANI).

TN2202 ring generator

The TN2202 ring generator circuit pack is designed for use in France.

The TN2202 ring generator circuit pack supplies 50-Hz ringing power. The TN2202 supplies balanced ringing to telephones that connect to the TN2183/TN2215 multicountry analog line circuit pack. A modified backplane allows this balanced ringing. The telephones must be administered for France analog transmission.

The TN2202 plugs into the power unit slot and is required for each carrier that contains analog lines requiring 50-Hz ringing. A carrier backplane that uses TN2202 requires a one-lead modification. This modification is required for all products that are made for France. TN2202 can:

- produce two symmetric voltages (usually 28 V RMS) with respect to ground
- take –48 VDC, –5 VDC, and ground from the backplane
- generate 2 × 28 V RMS with added –48 VDC

TN2207 DS-1 interface, T1 (24 channels) and E1 (32 channels)

The TN2207 circuit pack supports digital signal level 1 (DS-1) 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 that use the following protocols:

- Robbed-bit signaling
- Proprietary bit-oriented signaling (BOS) 24th-channel signaling
- DMI-BOS 24th-channel signaling

The circuit packs also support ISDN-PRI connectivity T1 or E1.

Note:

This circuit pack is not used in a G650 Media Gateway.

In a 24-channel DS-1 mode, a DS-1 interface is provided to the DS-1 facility. The TN2207 circuit packs provide circuit pack-level administrable A-Law and Mu-Law companding, CRC-4 generation and checking for E1 only, and stratum-3 clock capability.

TN2207 provides test jack access to the DS-1 or E1 line and supports 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.

TN2207 has additional hardware to support direct cables to a TN787 MMI circuit pack.

TN2209 tie trunk (4-wire, 4 ports)

The TN2209 tie trunk is designed for use in Russia.

The TN2209 tie trunk has four ports used for Type 1 or Type 5 4-wire E&M lead signaling tie trunks. The tie trunks can be one of four types: 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 TN760D each port has modified E&M signal leads for universal hardware compatibility. The TN2209 provides release link trunks that are required for the Centralized Attendant Service (CAS) feature and has administrable A-Law and Mu-Law companding.

TN2224CP DCP digital line (2-wire, 24 ports)

The TN2224CP is designed for use in the United State, Canada, and international countries for offer B only.

The TN2224 has 24 DCP ports that can connect to 2-wire digital telephones. Such telephones include 2400-series and 6400-series telephones, the 302C and the 302D attendant console, and the Callmaster IV, V, and VI.

The TN2224 supports either A-Law or Mu-Law companding.

The following table lists the TN2224CP-supported telephones and their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
302C/D console	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
Callmaster-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
2400-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
6400-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)

TN2215/TN2183 analog line for multiple countries (16 ports) (international offers or Offer B only for US and Canada)

The TN2215 and the TN2183 analog line circuit packs are designed for international offers or for offer B in the United State and Canada.

TN2215 and 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:

- analog telephone
- answering machine
- FAX
- loop-start CO port

TN2215 and TN2183 provides rotary digit 1 recall, ground-key recall, and programmable flash timing. TN2215 and TN2183 provide additional support for selectable ringing patterns, LED message waiting, and secondary lightning protection.

TN2215 and TN2183 supports on-premises wiring with either touchtone or rotary dialing, and with or without the LED message waiting indicators. TN2215 and TN2183 supports off-premises wiring with either DTMF or rotary dialing. LED message waiting indicators are not supported off-premises. Neon message waiting indicators are not supported.

A maximum of six to eight simultaneous ringing ports is allowed depending on the ringing cadence selected. The TN2215 and the TN2183 supports A-Law and Mu-Law companding and administrable timers.

TN2215 and TN2183 also support balanced ringing. When balanced ringing is configured for France, use the TN2202 ring generator circuit pack.

TN2215 and TN2183 support DTMF sending levels that are appropriate for Avaya IVR.

TN2215 and TN2183 are impedance and gain selectable for multiple countries. For more information, contact your Avaya representative.

The following table lists the TN2215- and TN2183-supported telephones and their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
2500 type	24 (0.2 mm^2/0.5 mm)	20,000 (6,096 m)
6200 type	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)

Telephone	Wire size (AWG)	Maximum range (feet)
7102A series	24 (0.2 mm^2/0.5 mm)	3,100 (945 m)
8100 series	24 (0.2 mm^2/0.5 mm)	12,000 (3,657 m)

TN2224CP DCP digital line (2-wire, 24 ports)

The TN2224CP has 24 DCP ports that can connect to 2-wire digital telephones. Such telephones include the 6400-series, 8400-series, or 9400-series telephones and the 302C or 302D attendant console.

TN2224 circuit pack supports either A-Law or Mu-Law companding.

The following table lists the TN2224-supported telephones and their wiring sizes and ranges.

Telephone	Wire size (AWG)	Maximum range (feet)
302C/D console	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
Callmaster-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
2400-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)
6400-series	24 (0.2 mm^2/0.5 mm) or 26	3,500 (1,067 m)

TN2242 digital trunk

The TN2242 digital trunk circuit pack supports versions of channel-associated signaling and ISDN-PRI signaling. These signaling versions are peculiar to the TTC private networking environment that is used in Japan. The TN2242 supports the special line-coding and framing that are used on 2.048-mbps Japanese trunks. The TN2242 connects the switch with other vendor equipment and with other DEFINITY switches through the TDM device. The TDM device is commonly used throughout Japan for this purpose.

TN2301 logic switch

The TN2301 provides service to the customer when one of the following is true:

- the link to the main processor fails
- the link to the main processor is severed
- the processor or Center Stage Switch (CSS) fails

The TN2301 Survivable Remote Switch (SRS) logic circuit pack connects the expansion port network (EPN) links to the appropriate processor port network (PPN) for call processing. The EPN links can be fiber or T1/E1. This connection is under the control of the TN775C Maintenance circuit pack which monitors the condition of the expansion interface TN570B.

The TN2301 is not used in an ATM-PNC.

TN2302AP IP media processor

The TN2302AP IP Media Processor is the H.323 audio platform and includes a 10/100 BaseT Ethernet interface. TN2302AP provides voice over internet protocol (VoIP) audio access to the switch for local stations and outside trunks. TN2302AP provides audio processing for between 32 and 64 voice channels, depending on the CODECs in use. TN2302AP is compatible with and can share load balancing with the TN2602AP Media Resource 320 circuit pack. See Comparison of TN2302AP Media Processor and TN2602AP IP Media Resource 320.

TN2302AP supports hairpin connections and the shuffling of calls between TDM connections and IP-to-IP direct connections. TN2302AP can also perform the following functions:

- Echo cancellation
- Silence suppression
- Fax relay service using T.30 and T.38 standards
- Dual-tone multifrequency (DTMF) detection
- Conferencing

TN2302AP can be updated using the firmware download feature.

The TN2302AP, starting with vintage 32, supports the following conversion resources for codec regarding voice, conversion between codecs, and fax detection:

- G.711, A-law or Mu-law, 64 kbps
- G.723.1, 6.3 kbps or 5.3 kbps audio
- G.729A, 8 kbps audio
- G.729, G.729B, G.729AB

The TN2302AP also supports transport of the following devices:

- Fax, Teletypewriter device (TTY), and modem calls over a corporate IP intranet using pass-through mode
- Fax and TTY calls using proprietary relay mode

  Note:
  TN2302AP does not support encryption of faxes sent to nonAvaya endpoints.

- 64-kbps clear channel transport in support of BRI secure telephones and data appliances (includes support for H.320 video over IP-connected Port networks)
- T.38 Fax over the Internet (including endpoints connected to nonAvaya systems)
- Modem tones over a corporate IP intranet

  Note:
  The path between endpoints for modem tone transmissions must use Avaya telecommunications and networking equipment.

For more information, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

TN2308 direct inward dialing trunk (8 ports)

The TN2308 uses eight ports for immediate-start or wink-start direct inward dialing (DID) trunks for Brazil. Each port has tip and ring signal leads.

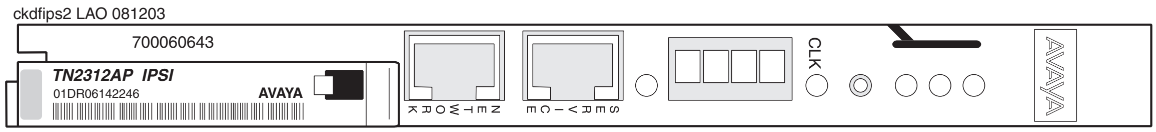
The switch requires the TN2308 to support Brazil Block Collect Call. The TN2308 transmission characteristics comply with the Brazilian telecommunication standards for PBXs.

TN2312BP IP server interface

The TN2312BP IP server interface (IPSI) provides transport of control messages. The messages are sent from the S8510 or S8800 Server to the port networks of server using the customer LAN and WAN. Through these control messages, the server controls the PNs.

An example of the TN2312BP IPSI faceplate is provided in Figure 40: IPSI faceplate.

Figure 40: IPSI faceplate



Detailed description

Dedicated and nondedicated network for control messages

You can configure the path for control messages to be over a LAN dedicated to communication between the server and IPSI. In this case, the network for carrying calls, the bearer path, is separate from the dedicated LAN for control messages. The bearer path uses nondedicated LAN of customer, a center stage switch (CSS) configuration, or an asynchronous transmission mode (ATM) network.

You can also configure the path for control messages to use the customer's nondedicated LAN. In this case, the bearer path and control path use the same network.

TN2312BP IPSI capabilities

The TN2312BP IPSI always resides in the tone clock slot on a Media Gateway and uses a 10/100 BaseT interface to connect to:

- The server
- A laptop computer connected to the server through a services port

The IPSI provides the following functions:

- PN clock generation and synchronization for Stratum 4 type II only
- PN tone generation
- PN tone detection, global call classification, and international protocols
- Processing of product serial numbers for license file activation
- Environmental maintenance, only on a G650 Media Gateway

To access the TN2312BP IPSI remotely, use the Telnet and SSH protocols. The TN2312BP IPSI can serve as an SSH client as well, for remote access from the TN2312BP IPSI to Communication Manager server. The C-LAN can also serve as an FTP or SFTP server for file transfers and primarily firmware downloads.

Note:

The IPSI cannot serve as an SFTP client. Additionally, the SSH/SFTP capability is only for the control network interface, not the Services interface.

The IPSI supports the following functions and devices:

- Eight global call classification ports
- Network diagnostics
- Download of SIPI firmware updates using Communication Manager Web pages, the **`loadipsi`** command from the server's Linux command line, or the Software Update Manager.

The TN2312BP IPSI is compatible with G650 Media Gateway and provides environmental maintenance only when it is used in a G650 Media Gateway.

IPSI support for system maintenance

TN2312BP IPSI can only be placed in G650 with a carrier address set to A or B. When set to A, TN2312BP IPSI acts as the serial bus master. The TN2312BP IPSI also provides environmental maintenance for G650. This includes:

- Power supply, cabinet, and ring generator maintenance
- External device alarm detection
- Emergency transfer control
- Customer-provided alarm device control

The TN2312BP IPSI and the 655A power supply provide the following information to G650:

- Environment maintenance
  - Inlet temperature of G650
  - Exhaust temperature of G650
  - Hot Spot temperature status
  - Voltage, +5, -5, or -48
  - Fan speed
  - Fan alarm
  - Ring status
  - Ring control
  - Ringer Setting
  - Ring Detection
  - Input Power, AC or DC

- **External device alarm detection**

  The external device alarm detection uses two external leads. External devices such as an uninterruptible power supply (UPS) or voice messaging system can use these leads to generate alarms. The external device uses Communication Manager alarm reporting capability. Ground potential on either of these leads results in an alarm being generated. You can administer the alarm level, product ID, alternate name, and alarm description for each lead. The alarm levels are major, minor, and warning.

- **Emergency transfer control**

  Emergency transfer control provides -48 VDC to operate an external emergency transfer panel. Communication Manager controls the state of the emergency transfer and generates an alarm when the emergency transfer is set to other than auto.

- **Customer-provided alarm device control**

  Customer-provided alarm device (CPAD) provides a contact closure across a pair of external leads. These leads can control a customer-provided alarm device or an alarm indicator. The level of alarm can be administered system wide to cause a contact closure. The alarm levels are major, minor, warning, or none. When the alarm level matches the alarm level that was administered, the TN2312BP IPSI closes this contact for all G650s. This closure occurs by a carrier address set to A. When TN2312BP IPSI is in emergency transfer, it closes this contact to activate the CPAD.

I/O adapters

The TN2312BP IPSI requires an adapter that provides for the alarm input, CPAD, and emergency transfer leads. This adapter also allows the IPSI Ethernet connection to be made to the back of the IPSI slot.

Compatibility

The TN2312BP IPSI can replace the TN2312AP IPSI in the G650 Media Gateways.

However, the IPSI does not provide environmental maintenance for these Media Gateways.

Environmental maintenance requires monitoring of the AuxSig backplane lead cabinet when the TN2312BP is installed in a G650 Media Gateway with Communication Manager 2.0. If this lead detects a failure in the power supply or fan assembly, it sends an alarm.

See the following table for IPSI and Media Gateway compatibility.

Media Gateway	Communication Manager 1.x	Communication Manager 2.0	DEFINITY R10	Environmental maintenance provided by:
G650		Yes		TN2312BP IPSI

Number of IPSI circuit packs per configuration

For configurations where voice bearer is over CSS or ATM, each IPSI typically controls five port networks. Each IPSI achieves control by tunneling control messages over the bearer network to PNs that do not have IPSIs. An IPSI cannot be placed in:

- A PN that has a Stratum-3 clock interface
- A remote PN that is using a DS-1 converter
- A Survivable Remote Expansion Port Network (SREPN)

A simple formula determines the number of IPSI-connected PNs that should support an S8510 or S8800 configuration. Divide the total number of PNs in the configuration by five and add one. The additional IPSI provides fault tolerance. For example, if you have 20 PNs, divide 20 by 5 to get 4, then add 1. You need a minimum of five IPSIs to support the 20 PNs.

For configurations where voice bearer is over IP, there must be one IPSI in each PN.

A direct connect configuration only supports one IPSI-connected PN.

TN2313AP DS-1 interface (24 channels)

The TN2313AP DS-1 port circuit pack interfaces a DS-1 trunk to the switch backplane by port slots that are standard for DEFINITY products. The TN2313AP is compatible with the following:

- previous 24-channel DS-1 circuit packs, including the TN464F, vintage 19, and earlier
- TN2464, vintage 19 and earlier
- TN767E DS-1

Except, the TN2313AP does not provide for packet adjunct capabilities. The TN2313AP supports a variety of applications, including networking of the following:

- DEFINITY switches
- international trunk types
- video teleconferencing
- wideband data transmission

On S8510 and S8800 Servers, this circuit pack does not directly support D-channel signaling and thus does not directly support ISDN-PRI connectivity. However, the TN767 circuit can indirectly support D-channel signaling provided that the central office supports nonfacility associated signaling (NFAS). In this case, use NFAS administration on the server to associate the D-channel of another T1/E1 circuit pack, usually a TN464, with the TN767 circuit pack.

The TN2313AP DS-1 interface can be configured as 24 channels at 1.544 mbps. The TN2313 can supply two 8-kHz reference signals to the switch backplane. These signals can be used by the tone-clock circuit pack to synchronize the system clock and the received line clock.

The TN2313AP is downloadable firmware.

TN2464CP DS-1 interface with echo cancellation, T1/E1

The TN2464CP DS-1 circuit pack is designed for international use in both category A and category B. The TN2464CP has echo cancellation circuitry and firmware download capability. The TN2464CP supports T1 (24-channel) and E1 (32-channel) digital facilities. The TN2464CP has the same functionality as the TN464HP, which is offered in the United States and Canada.

The TN2464CP circuit pack provides:

- Test jack access to the T1/E1 line
- Circuit-pack-level administrable A-law and Mu-law companding
- CRC-4 generation and checking (E1 only)
- Support for the 120A 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 Avaya Interactive Response
- Channel-associated signaling protocols for many countries. For details, contact your Avaya representative

To update TN2464CP with the firmware download feature, use the TN799 C-LAN interface.

TN2501AP voice announcements over LAN (VAL)

The TN2501AP is an integrated announcement circuit pack that:

- Offers up to one hour of announcement storage capacity
- Provides shorter backup and restore time
- Is firmware that can be downloaded
- Plays announcements over the TDM bus, similar to the TN750C circuit pack

- Has 33 ports, including
  - One dedicated telephone access port for recording and playing back announcements using port number 1
  - One Ethernet port using port number 33
  - 31 playback ports using port numbers 2 to 32
- Uses a 10-mbps/100-mbps ethernet interface to allow portability of announcements and firmware files over a LAN
- Uses announcement files that are in ".wav" format (CCITT A-law and µ-law, 8 kHz, 8-bit mono)

The VAL can serve as an FTP or SFTP server for file transfers — primarily firmware downloads. The VAL cannot serve as an SFTP client.

With Communication Manager Release 3.1 and later, the VAL can also receive firmware downloads from a central firmware depository on an SCP-enabled file server.

For more information on firmware downloads and instructions for downloading, see http://www.avaya.com/support/

Configuration

Figure 41 shows the configuration options for the TN2501AP (VAL) circuit pack within a system.

Figure 41: VAL configuration options

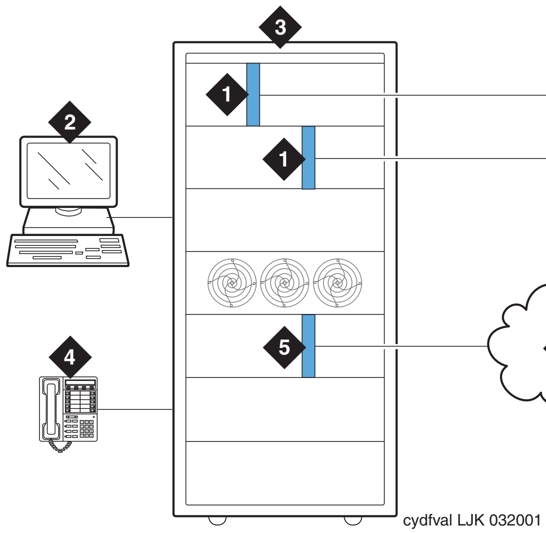


Figure notes:

1. **TN2501AP VAL announcement circuit pack**
2. **System access terminal (SAT)**
3. **Switch**
4. **Phone for recording announcements**
5. **TN799DP (C-LAN) is required when using IP SAT or VAL Manager.**
6. **Your LAN (See LAN cable)**
7. **Computer or remote recording studio for:**
   - recording and storing announcements
   - FTP client application
8. **VAL Manager application (PC only)**
9. **Microphone**

Hardware specifications

The following table contains a list of the required VAL hardware.

Part	Number
TN2501AP	1
Backplane Adapter (Label reads IP Media Processor)	1

To establish LAN connections, the TN2501AP circuit pack requires a:

- Backplane Adapter that attaches to the Amphenol connector on the back of the cabinet, corresponding to the TN2501AP integrated announcement circuit pack slot.
- LAN cable that attaches to the Backplane Adapter.

Backplane Adapter

The following figure shows the Backplane Adapter (label reads IP Media Processor).

Figure 42: Backplane Adapter

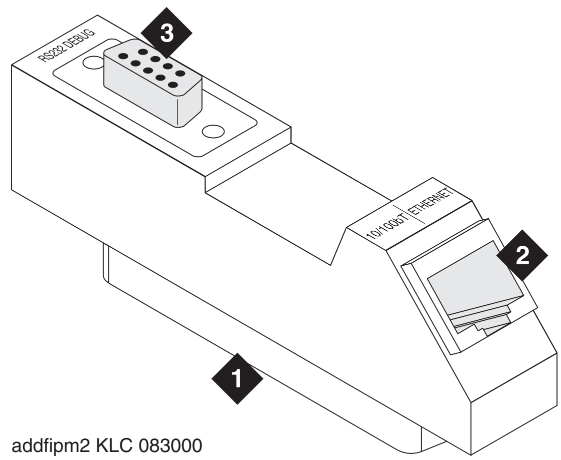


Figure notes:

1. **Amphenol connector attaches to the back of the switch cabinet, corresponding to the TN2501AP circuit pack's slot.**
2. **RJ-45 LAN cable connection**
   - 10 mbps uses Category 3 cable
   - 100 mbps uses Category 5 cable
3. **This connector is not used for VAL.**

LAN cable

The TN2501AP circuit pack does not include cables to connect the circuit pack to your LAN. The following table lists the cable category and connection port.

Ethernet connection speed	Cable	Connection description
10 mbps	Category 3	Connects through the RJ45 jack (See Figure 42),
100 mbps	Category 5	Connects through the RJ45 jack (See Figure 42),

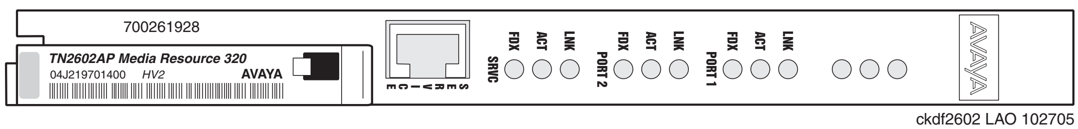
TN2602AP IP Media Resource 320

The TN2602AP IP Media Resource 320 provides high-capacity voice over Internet protocol (VoIP) audio access to the switch for local stations and outside trunks. The TN2602AP provides audio processing for the following types of calls:

- TDM-to-IP and IP-to-TDM — for example, a call from a 4602 IP telephone to a 6402 DCP telephone
- IP-to-IP — for example, a non-shuffled conference call

For an example of the TN2602AP faceplate, see Figure 43: IP Media Resource 320 faceplate.

Figure 43: IP Media Resource 320 faceplate



The TN2602AP IP Media Resource 320 circuit pack has two capacity options, both of which are determined by the license file installed on Communication Manager:

- 320 voice channels, considered the standard IP Media Resource 320
- 80 voice channels, considered the low-density IP Media Resource 320

Only two TN2602AP circuit packs are allowed per port network.

Detailed description

Load balancing

Up to two TN2602AP circuit packs may be installed in a single port network for load balancing. The TN2602AP circuit pack is also compatible with and can share load balancing with the TN2302 and TN802B IP Media Processor circuit packs. Actual capacity may be affected by a variety of factors, including the codec used for call and fax support.

Note:

The maximum number of time slots available for a port network is 484. Therefore, when a port network uses two TN2602AP circuit packs for load balancing, each with 320 voice channels, the total number of voice channels available is 484.

Bearer duplication

Two TN2602AP circuit packs may be installed in a single port network (PN) for bearer duplication. In this configuration, one TN2602AP is an active IP media processor and one is a standby IP media processor. If the active media processor fails or connections to it fail active connections failover to the standby media processor and remain active. This duplication prevents active calls in progress from being dropped in case of failure. The interchange between duplicated circuit packs affects only the PN in which the circuit packs reside.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls may be dropped.

Virtual IP and MAC addresses to enable bearer duplication

Duplicated TN2602AP circuit packs in a PN share a virtual IP and virtual MAC address. These virtual addresses are owned by the currently active TN2602. In addition to the virtual IP address, each TN2602 has a "real" IP address. All bearer packets sent to a PN that contains duplicated TN2602AP circuit packs, regardless of whether the packets originate from TN2602s in other PNs or from IP phones or gateways, are sent to the virtual IP address of the TN2602 pair in that PN. The TN2602AP circuit pack that is active receives those packets.

When failover to the standby TN2602 occurs, a negotiation between TN2602s to determine which TN2602 is active and which is standby takes place. State-of-health, call state, and encryption information is shared between TN2602s during this negotiation. The newly-active TN2602AP circuit pack sends a gratuitous address resolution protocol (ARP) request to ensure that the LAN infrastructure is updated appropriately with the location of the active TN2602. Other devices within the LAN update their old mapping in ARP cache with this new mapping.

Requirements for bearer duplication

The Communication Manager license file must have entries for each circuit pack, with the entries having identical voice channels enabled. In addition, both circuit packs must have the latest firmware that supports bearer duplication.

Duplicated TN2602AP circuit packs must be in the same subnet. In addition, the Ethernet switch or switches that the circuit packs connect to must also be in the same subnet. With the shared subnet, the Ethernet switches can use signals from the TN2602AP firmware to identify the MAC address of the active circuit pack. This identification process provides a consistent virtual interface for calls.

Combining duplication and load balancing

A single port network can have up to two TN2602AP circuit packs only. As result, the port network can have either two duplicated TN2602AP circuit packs or two load balancing TN2602AP circuit packs. However, in a Communication Manager configuration, some port networks can have a duplicated pair of TN2602AP circuit packs and other port networks can have a load-balancing pair of TN2602AP circuit packs. Some port networks can also have single or no TN2602AP circuit packs.

Note:

If a pair of TN2602AP circuit packs previously used for load balancing are re-administered to be used for bearer duplication, only the voice channels of the circuit pack that is active can be used. For example, if you have two TN2602 AP circuit packs in a load balancing configuration, each with 80 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 80 instead of 160 channels available. If you have two TN2602 AP circuit packs in a load balancing configuration, each with 320 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 320 instead of 484 channels available.

Features

The IP Media Resource 320 supports hairpin connections and the shuffling of calls between TDM connections and IP-to-IP direct connections. The IP Media Resource 320 can also perform the following functions:

- Echo cancellation
- Silence suppression
- Adaptive jitter buffer (320 ms)
- Dual-tone multifrequency (DTMF) detection
- AEA Version 2 and AES media encryption
- Conferencing
- QOS tagging mechanisms in layer 2 and 3 switching (Diff Serv Code Point [DSCP] and 802.1pQ layer 2 QoS)
- RSVP protocol

The TN2602AP IP Media Resource 320 circuit pack supports the following codecs for voice, conversion between codecs, and fax detection:

- G.711, A-law or Mu-law, 64 kbps
- G.726A-32 kbps
- G.729 A/AB, 8 kbps audio

The TN2602AP also supports transport of the following devices:

- Fax, Teletypewriter device (TTY), and modem calls using pass-through mode
- Fax, V.32 modem, and TTY calls using proprietary relay mode

  Note:
  V.32 modem relay is needed primarily for secure SCIP telephones (formerly known as Future Narrowband Digital Terminal (FNBDT) telephones) and STE BRI telephones.

- T.38 fax over the Internet, including endpoints connected to nonAvaya systems
- 64-kbps clear channel transport in support of firmware downloads, BRI secure telephones, and data appliances

The TN2602AP supports SRTP media encryption.

Firmware download

The IP Media Resource 320 can serve as an FTP or SFTP server for firmware downloads to itself. However, this capability is activated by and available for authorized services personnel only.

I/O adapter

The TN2602AP IP Media Resource 320 circuit pack has a services Ethernet port in the faceplate. The TN2602AP circuit pack also requires an input/output adapter that provides for one RS-232 serial port and two 10/100 Mbs Ethernet ports for LAN connections (though only the first Ethernet port is used). This Ethernet connection is made at the back of the IP Media Resource 320 slot. See Figure 44: IP Media Resource 320 I/O adapter.

Note:
The TN2302AP can also use this I/O adapter.

Figure 44: IP Media Resource 320 I/O adapter

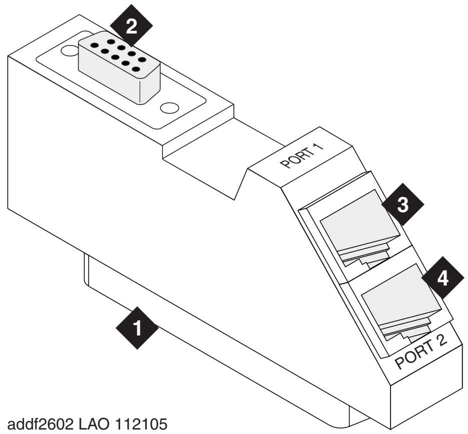


Figure notes:

1. **Amphenol connector to backplane connector corresponding to TN2602AP slot**
2. **RS-232 connector for services**
3. **Port 1: RJ45 LAN cable connection for 100 mbps CAT5 cable**
4. **Port 2: RJ45 LAN connection for future use (do not use)**

Comparison of the TN2602AP and TN2302AP circuit packs

The following table compares key features of the TN2602AP IP Media Resource 320 circuit pack and the TN2302AP Media Processor circuit pack.

Table 37: Comparison of TN2302AP Media Processor and TN2602AP IP Media Resource 320

Supported Features	TN2302AP Media Processor (V10 and later)	TN2602AP IP Media Resource 320 (standard and low density)
VoIP Media Processing Channels	64 (G.711)	320 (standard) or 80 (low density), based on license
License control	no	yes
T.38 Fax Interoperability	yes	yes
Fax Pass Through	yes	yes
Fax Relay – Proprietary	yes	yes
		1 of 3

Table 37: Comparison of TN2302AP Media Processor and TN2602AP IP Media Resource 320 *(continued)*

Supported Features	TN2302AP Media Processor (V10 and later)	TN2602AP IP Media Resource 320 (standard and low density)
Modem Pass Through	yes	yes
Modem Relay – Proprietary	yes	yes
TTY Pass Through	yes	yes
TTY Relay	yes	yes
Clear channel	yes	yes
Echo Cancellation	yes (32 ms full tail)	yes (128 ms tail, 24 ms window)
DTMF Detection/Generation	yes	yes
Communication Manager can load balance between multiple boards	yes	yes
Bearer duplication	no	yes
AEA.2, AES media encryption	yes (use of AES reduces channel availability by 25%)	yes (use of AES does not reduce channel availability)
Resiliency to DOS attacks	yes	yes
Firmware download	yes (requires C-LAN)	yes (self-downloadable)
Reporting and recovery from bad/corrupt embedded SW	yes	yes
Built-in test support • Sanity confirmation at boot • Loop back tests • Shallow IP and TDM loop back mode • Embedded firmware self test routines upon board initialization	yes	yes
Ping test support	yes	yes
		2 of 3

Table 37: Comparison of TN2302AP Media Processor and TN2602AP IP Media Resource 320 *(continued)*

Supported Features	TN2302AP Media Processor (V10 and later)	TN2602AP IP Media Resource 320 (standard and low density)
VoIP engine monitoring	yes	yes
VoIP engine resets	yes	yes
Trace route support	yes	yes.[1]
RS232 port user interface	yes	yes
Enable/disable FTP & Telnet services	Enable/disable Telnet only in V58 and later.	yes
Enable/disable SFTP and SSH services	no	yes
Service access	RS232 port out the back – no password required	Faceplate services Ethernet port or RS232 port in the back. VxWorks shell access. Password protected
Ethernet ports	A single 10/100mbps Ethernet port out the back. Uses an adapter.	Two 10/100mbps Ethernet ports. Only one used. Uses an adapter to access both ports.
Codecs	• G.711 (64 channels maximum, unencrypted; 48 channels maximum, encrypted) • G.729B and G.723.1 (32 channels maximum, unencrypted; 24 channels maximum, encrypted)	• G.711 (320 channels maximum, unencrypted or encrypted) • G.729A, G.729AB, (320 channels maximum, unencrypted or encrypted) • G.726A (320 channels maximum)
		3 of 3

1. For additional information on trace route, including limitation with the TN2602AP circuit pack, see the Maintenance documentation.

Hardware requirements

The TN2602AP IP Media Resource 320 feature requires the following hardware:

- TN2602AP circuit pack with one 10/100BaseT Ethernet port for services access
- Media Resource 320 adapter with one RS-232 serial port and two 10/100BaseT Ethernet ports
- Slot in the Media Gateway that is CAT5 compliant.
- A CAT5 or equivalent cable, supplied by the customer

The TN2602AP works in the G650 Media Gateways (cabinets/carriers) supported by Release 3.1 of Avaya Communications Manager. G650 is the preferred media gateway for TN2602AP IP Media Resource 320.

TNCCSC-1 PRI to DASS converter

The TNCCSC-1 circuit pack converts ISDN-PRI to a Direct Access Secondary Storage (DASS) interface. DASS is a 2-mbps interface that uses a 75-Ohm coaxial transmission facility. One TNCCSC-1 circuit pack can support two TN464 DS-1 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 Digital Private Network Signaling System (DPNSS) interface. DPNSS is a 2-mbps interface that uses a 75-Ohm coaxial transmission facility. One TNCCSC-2 circuit pack can support two TN464 DS-1 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 circuit pack, except that the TNCSSC-3 has a 120-Ohm twisted pair interface.

TN-C7 PRI to SS7 converter

The TN-C7 converter provides a gateway interface between the TN464 circuit pack and the public signaling network. The TN-C7 integrates DASS, DPNSS, and SS7 into a single circuit pack type. The TN-C7 supports international service provider call center customers. The TN-C7 converter is not designed for operation in the United State or Canada.

TN-CIN voice, fax, and data multiplexer

The TN-CIN provides QSIG and private networking transparency on demand across a switched network. The TN-CIN integrates circuits over a single separate digital link. The circuits include up to three G.728 LD-CELP voice or fax circuits, six CAFT voice or fax circuits, and two data circuits. The three or six voice or fax circuits are presented as a G.703 E1 data stream that uses either QSIG peer-to-peer or channel-associated signaling.

All voice or fax circuits support low bit rate voice compression at 8 to 16 kbps when the circuits use CAFT. When circuits use LD-CELP, all voice or fax circuits support the same voice compression at 16 kbps. 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 and 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 is available for data applications. This port uses V.24 or V.11 or V.35 at up to 115.2 kbps synchronous or V.24 at up to 115.2 kbps asynchronous. The port also incorporates dynamic bandwidth allocation, also known as variable data clocking. A low-speed V.24 data port of up to 96 kbps synchronous or 57.6 kbps asynchronous is available for data applications.

Media modules

MM312 DCP Media Module

Avaya MM312 Media Module provides 24 Digital Communications Protocol (DCP) ports with RJ-45 jacks. The MM312 supports simultaneous operation of all 24 ports. Each port can be connected to a 2-wire DCP telephone. The MM312 does not support 4-wire DCP telephones.

The MM312 is supported only in the G350 Media Gateway.

The MM312 supports the following loop length:

- 5500 feet (1676 meters) over 0.65 mm (.025 in.) wire (22 AWG)
- 3500 feet (1067 meters) over 0.5 mm (.02 in.) wire (24 AWG)
- 2200 (671 meters) over 0.4 mm (.016 in.) wire (26 AWG)

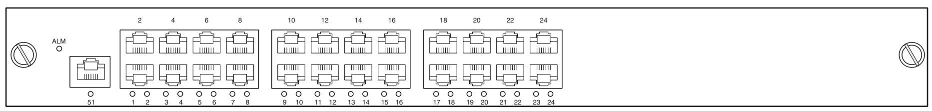
DANGER:
The ports on the MM312 are intended for in-building use only. Phone lines connected to those ports are not to be routed out-of-building. Failure to comply with this restriction could cause harm to personnel or equipment.

MM314 LAN Media Module

The Avaya MM314 Media Module provides:

- 24 Ethernet 10/100 Base-T Ethernet access ports with inline Power over Ethernet (PoE).
- One Gigabit Ethernet Small Form-Factor Pluggables (SFP) GigaBit Interface Converter (GBIC) slot which supports any of the following SFP GBICs: 1000-SX, 1000-LX, 1000-ELX or 1000-TX.

The MM314 is supported only in the G350 Media Gateway.



h2cmm314 LAO 111705

The MM314 supports 48VDC inline power provided over standard category 5 UTP cables, up to 100 meter range, on each PoE port.

The MM314 supports the following features:

- Priority power budgeting with configurable priorities
- Automatic load detection on ports
- Automatic device discovery
- Enable/disable port powering option
- Port monitoring
- Automatic recovery from overload shutdown

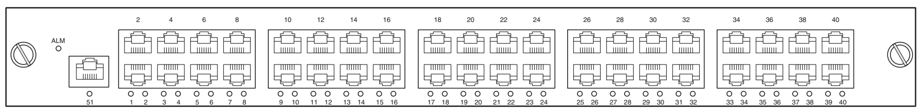
Automatic recovery from no-load shutdown

MM316 LAN Media Module

The MM316 LAN media module provides:

- 40 Ethernet 10/100 Base-T Ethernet access ports with inline Power over Ethernet (PoE).
- One Gigabit Ethernet copper port for server connection or uplink to another switch or router.

The MM316 is supported only in the G350 Media Gateway.



h2cmm316 LAO 111605

The MM316 supports 48VDC inline power provided over standard category 5 UTP cables (up to 100m range) on each PoE port.

The MM316 supports the following features:

- Priority power budgeting with configurable priorities
- Automatic load detection on ports
- Automatic device discovery
- Enable/disable port powering option
- Port monitoring
- Automatic recovery from overload shutdown
- Automatic recovery from no-load shutdown

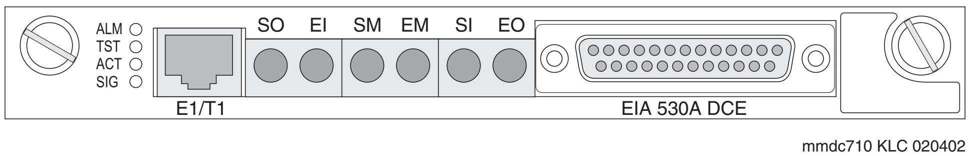
The MM316 is compatible with ACM version 2.0 and later and G350 Media Gateway firmware version 25.0.0 and later.

MM710 T1/E1 Media Module

The Avaya MM710 T1/E1 Media Module terminates a T1 or an E1 connection to either private enterprise network trunks or to public network trunks. The MM710 T1/E1 has a built-in Channel Service Unit (CSU). An external CSU is not necessary.

The MM710 T1/E1 is supported in the G700, G450, G430, and G350 Media Gateways.

Figure 45: Avaya MM710 T1/E1 Media Module



Detailed description

Highlights of the MM710 include:

- Software-selectable T1 or E1 operation
- Integrated CSU
- Both A-law (E1) and μ-law (T1) gain control and echo cancellation ability
- D4, ESF, or CEPT framing
- ISDN PRI capability: 23 B-channel + 1 D-channel or 30 B-channel + 1 D-channel
- AMI, ZCS, B8ZS (T1) or HDB3 (E1) line coding
- Trunk signaling to support US and international central office (CO) or tie trunks
- Echo cancellation in either direction
- Fractional T1 support
- OIC DB 25-pin interface
- Bantam loopback jack tests T1 circuits or E1 circuits

The MM710 supports the universal DS-1 that conforms to the ANSI T1.403 1.544 mbps T1 standard and to the ITU-T G.703 2.048 mbps E1 standard.

The MM710 does not support Code Mark Inversion line coding used in Japan.

Echo cancellation

The MM710 Media Module can cancel echoes in either direction for any DS0. The MM710 can cancel echoes with tail-end delays up to 96 milliseconds. The MM710 is compatible with either A-law or Mu-law code.

CSU function

The CSU functionality that is built into the MM710 Media Module has the following capabilities:

- Long-haul or short-haul transmission
- Reception of signals as low as -36 dB
- Compensate for distances up to 655 feet (200 meters) in short-haul operation
- Attenuation up to -22.5 dB can be programmed when driving repeaters for long-haul transmission

Loopback and BERT functions

The loopback and bit error rate testing (BERT) functionality in the MM710 Media Module has the following characteristics:

- Provides a passive loopback for the far-end in an unpowered state
- Can be set up for line or payload loopbacks
- Supports incoming and outgoing ESF FDL requests
- Can generate and respond to in-band loop up and loop down codes per ANSI-T1.403
- Supports the generation and detection of test patterns and injection of bit errors for Bit Error Rate Testing

E1 impedance

By itself, the MM710 Media Module 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 of the MM710 Media Module provide access to the incoming and outgoing T1 signals or E1 signals:

- SM allows passive monitoring of the incoming line
- EM allows passive monitoring of the outgoing line
- SO allows intrusive monitoring of the incoming signal from the network. When used, the SO jack breaks the connection of that signal to the framer

- EI allows injection of a signal towards the framer. When used, the EI jack isolates the network Rx signal.
- SI allows injection of a signal towards the network. When used, the SI jack isolates the framer Tx signal from going out to the network.
- EO allows intrusive monitoring of the signal from the framer. When used, the EO jack breaks the connection of that signal to the network jack RJ48C.

LEDs

The MM710 faceplate supports four LEDs. These LEDs include the three standard Media Module LEDs and the SIG LED that indicates that the MM710 Media Module is receiving a valid signal.

DB 25 DCE connector

MM710 includes a DB DCE connector is included and can connect a data service unit (DSU) in a future release.

Loopback jack

When you order an MM710 T1/E1 Media Module, Avaya recommends that you include the optional 700A loopback jack. With the loopback jack installed, you can 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.

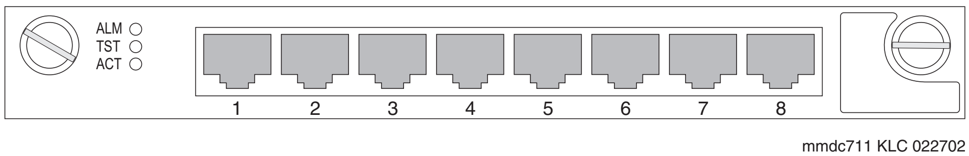
The jack is usually used for CO trunk installations. The jack is inserted as close to the network or service provider T1 facility as possible. When the jack is activated from the Media Gateway, the jack sets up loopbacks in both directions. The Media Gateway can then send and receive a test pattern. The test pattern verifies the function of the MM710 and T1 cable up to the network T1 facility. In normal operation, the jack passes the T1 signals through undisturbed in both directions.

MM711 Analog Media Module

The Avaya MM711 Analog Media Module provides features and functionality for analog trunks and telephones.

The MM711 is supported in the G700, G450, G430, and G350 Media Gateways.

Figure 46: Avaya MM711 Analog Media Module



Detailed description

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

- A loop start or a ground start central office trunk with a loop current of 18 to 120 mA.
- A wink-start or an immediate-start Analog Direct Inward Dialing (DID) trunk
- A two-wire analog Outgoing CAMA E911 trunk, for connectivity to the public switched telephone network (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 Analog Media Module also supports:

- Three ringer loads, which is the ringer equivalency number, for all eight ports, for the following loop lengths:
  - 20,000 feet (6096 meters) over 0.65 mm (.025 in.) wire (22 AWG)
  - 16,000 feet (4877 meters) over 0.5 mm (.02 in.) wire (24 AWG)
  - 10,000 feet (3048 meters) over 0.4 mm (.016 in.) wire (26 AWG)

  At .1 or less REN ringer loads, the supported loop length is 20,000 feet (6096 meters) at 22, 24, and 26 AWG.
- Up to eight ports ringing simultaneously

  Note:

  The Media Gateway achieves this number of ports by staggering the ringing and pausing between two sets of up to four ports.

If it has more than four ports, the MM711 also supports:

- Type 1 caller ID 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 on the CO trunk side

The following requirements apply to the external interfaces on the CO trunk side:

- The tip and ring default input impedance is 600 Ohms. The default impedance can be configured to accommodate other tip and ring impedances. One such impedance is the 900 Ohms that is used in Brazil. Another is the complex impedance that is 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 that are set up as direct inward dialing (DID).
- The acceptable loop range for the CO trunk is 18 to 60 mA.
- The MM711 supports direct inward and outward dialing (DIOD) for Japan.

MM711 supports the following trunk types:

- Loop-start and ground-start CO trunks
- DID
- CAMA

Caller ID

The MM711 Analog Media Module supports incoming caller ID (ICLID) on analog CO loop-start trunks for all supported countries that require this feature. The MM711 supports Type 1 caller ID (CID) devices, and firmware signaling requirements are implemented on a per-port basis. The firmware supports these 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.

A call can still be terminated on a trunk that is administered for ICLID. The call is terminated even if there is no ICLID information or error in transmission of ICLID information. Japan is an exception.

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 to 300 Ohms, Rp: 750 to 1000 Ohms, Cp: 115 to 220 pF
- A ringing frequency range of 20 Hz, 25 Hz, or 50 Hz
- A DC current range of 20 to 60 mA
- A hook flash range of 90 to 1000 ms

Companding

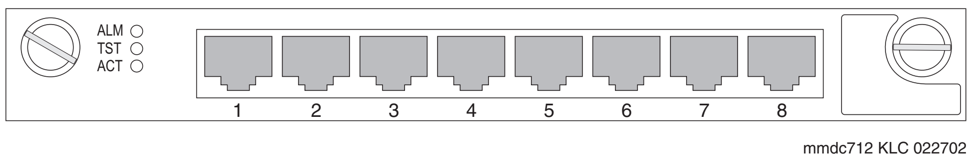
MM711 allows for A-law or Mu-law selection at installation. This is a software-selectable capability that applies to all ports on the MM711.

MM712 DCP Media Module

Use the MM712 DCP Media Module to connect up to eight two-wire Digital Communications Protocol (DCP) voice terminals.

The MM712 is supported in the G700, G450, G430, and G350 Media Gateways.

Figure 47: Avaya MM712 DCP Media Module



Hardware interface

Signal timing specifications for the MM712 support TDM bus timing in receive and transmit modes. The Media Gateway supplies only +5 VDC and –48 VDC to the MM712 Media Module. Any other required voltages must be derived on the module.

MM712 provides loop range secondary protection. The MM712 is also self-protecting from an over-current condition on a tip and ring interface. The MM712 supports the following loop length:

- 5500 feet (1676 meters) over 0.65 mm (.025 in.) wire (22 AWG)
- 3500 feet (1067 meters) over 0.5 mm (.02 in.) wire (24 AWG)
- 2200 (671 meters) over 0.4 mm (.016 in.) wire (26 AWG)

DANGER:
The ports on the MM712 are intended for in-building use only. Phone lines connected to those ports are not to be routed out-of-building. Failure to comply with this restriction could cause harm to personnel or equipment.

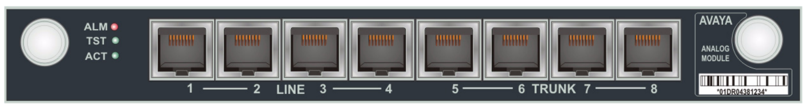
MM714 Analog Media Module

The Avaya MM714 Media Module provides four analog telephone ports and four analog trunk ports.

Note:

You must use the four analog line ports and not the four analog trunk ports for analog DID trunks.

The MM714 is supported in the G700, G450, G430, and G350 Media Gateways.



Detailed description

MM714 provides the capability to configure any of the four trunk ports as:

- A loop start or a ground start central office trunk with a loop current of 18 to 120 mA.
- A two-wire analog Outgoing CAMA E911 trunk, for connectivity to the public switched telephone network (PSTN). MF signaling is supported for CAMA ports.

MM714 provides the capability to configure any of the four line ports as:

- A wink-start or an immediate-start Analog Direct Inward Dialing (DID) trunk
- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication.

The MM714 Analog Media Module also supports:

- Three ringer loads, which is the ringer equivalency number, for all four line ports, for the following loop lengths:
  - 20,000 feet (6096 meters) over 0.65 mm (.025 in.) wire (22 AWG)
  - 16,000 feet (4877 meters) over 0.5 mm (.02 in.) wire (24 AWG)
  - 10,000 feet (3048 meters) over 0.4 mm (.016 in.) wire (26 AWG)

  At .1 or less REN ringer loads, the supported loop length is 20,000 feet (6096 meters) at 22, 24, and 26 AWG.
- Up to four ports ringing simultaneously
- Type 1 caller ID 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 on the CO trunk side

The following requirements apply to the external interfaces on the CO trunk side:

- The tip and ring default input impedance is 600 Ohms. The default impedance can be configured to accommodate other tip and ring impedances. One such impedance is the 900 Ohms that is used in Brazil. Another is the complex impedance that is used in the European Union.
- A hard-wired ground wire is added for each IROB-to-earth ground.
- The MM714 supports DTMF, MF, and pulsing.
- The MM714 supports R2MFC address signaling.
- The acceptable loop range for the CO trunk is 18 to 60 mA.
- The MM714 supports direct inward and outward dialing (DIOD) for Japan.

Caller ID

The MM714 Analog Media Module supports up to four incoming caller ID (ICLID) on analog CO loop-start trunks for all supported countries that require this feature. The MM714 supports Type 1 caller ID (CID) devices, and firmware signaling requirements are implemented on a per-port basis. The firmware supports these formats:

- Single Data Message Format (SDMF)
- Multiple Data Message Format (MDMF)
- Caller ID generation on line ports

The MM714 accommodates on-hook transmission, which is necessary to receive caller ID signals.

A call can still be terminated on a trunk that is administered for ICLID. The call is terminated even if there is no ICLID information or error in transmission of ICLID information. Japan is an exception.

Analog line interface requirements

The MM714 provides pass through for fax signal on its analog line ports.

The MM714 supports up to four analog telephone sets with:

- An impedance range of Rs: 215 to 300 Ohms, Rp: 750 to 1000 Ohms, Cp: 115 to 220 pF
- A ringing frequency range of 20 Hz, 25 Hz, or 50 Hz
- A DC current range of 20 to 60 mA
- A hook flash range of 90 to 1000 ms

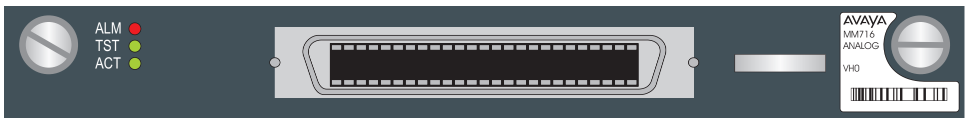
The MM714 provides -48 VDC for ports that are set up as direct inward dialing (DID).

Companding

MM714 allows for A-law or Mu-law selection at installation. This is a software-selectable capability that applies to all ports on the MM714.

MM716 Analog Media Module

The MM716 provides 24 analog ports supporting telephones, modem, and fax. These ports can also be configured as DID trunks with either wink-start or immediate-start. The 24 ports are provided through a 25 pair RJ21X amphenol connector, which can be connected by an amphenol cable to a breakout box or punch down block.



mmdc716 LAO 102505

The MM716 provides the capability to configure any of the 24 ports as:

- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication
- A wink-start or an immediate-start DID trunk

The MM716 Analog Media Module also supports:

- Three ringer loads, which is the ringer equivalency number, for all ports, for the following loop lengths:
  - 20,000 feet (6096 meters) over 0.65 mm (.025 in.) wire (22 AWG)
  - 16,000 feet (4877 meters) over 0.5 mm (.02 in.) wire (24 AWG)
  - 10,000 feet (3048 meters) over 0.4 mm (.016 in.) wire (26 AWG)

  At .1 or less REN ringer loads, the supported loop length is 20,000 feet (6096 meters) at 22, 24, and 26 AWG.
- Up to 24 ringing simultaneously ports
- Type 1 caller ID
- Ring voltage generation for a variety of international frequencies and cadences

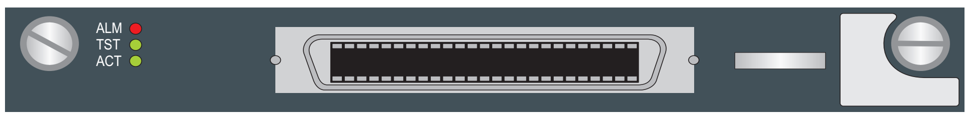
The MM716 is compatible with Avaya Communication Manager Release 3.1 and later and branch gateway firmware version 25.0.0 and later.

MM717 DCP Media Module

The Avaya MM717 Media Module provides 24 Digital Communications Protocol (DCP) ports connected through an RJ21X Amphenol connector. The MM717 supports simultaneous operation of all 24 ports. Each port can be connected to a 2-wire DCP telephone. The MM717 does not support 4-wire DCP telephones.

The MM717 is supported in the G700, G450, G430, and G350 Media Gateways.

Figure 48: Avaya MM717 DCP Media Module



mmdc717 LAO 102505

Signal timing specifications for the MM717 support TDM Bus Timing in receive and transmit modes. The G700, G450, G430, and G350 Media Gateways supply only +5 VDC and –48 VDC to the MM717 Media Module.

MM717 provides loop range secondary protection. The MM717 is also self-protecting from an over current condition on a tip and ring interface. The MM717 supports the following loop length:

- 5500 feet (1676 meters) over 0.65 mm (.025 in.) wire (22 AWG)
- 3500 feet (1067 meters) over 0.5 mm (.02 in.) wire (24 AWG)
- 2200 (671 meters) over 0.4 mm (.016 in.) wire (26 AWG)

The MM717 Media Module is connected to the wall field or breakout box with a B25A unshielded 25-pair cable.

DANGER:
The ports on the MM717 are intended for in-building use only. Phone lines connected to those ports are not to be routed out-of-building. Failure to comply with this restriction could cause harm to personnel or equipment.

MM720 BRI Media Module

The Avaya MM720 BRI Media Module contains eight ports that can be administered either as BRI trunk connections or BRI endpoint (telephone and data module) connections.

Note:

The MM720 BRI Media Module cannot be administered to support both BRI trunks and BRI endpoints at the same time. MM720 BRI Media Module supports a combination of B-channels using BONDing module Mode 1, to form a 128-kbps channel or a higher bandwidth connection.

Starting with Communication Manager Release 3.1, the MM720 BRI Media Module supports combining both B-channels, using BONDing, to form a higher bandwidth connection.

For BRI trunking, the MM720 BRI Media Module supports up to eight BRI interfaces, or up to 16 trunk ports, to the central office at the ISDN S/T reference point.

For BRI endpoints, each of the 8 ports on the MM720 BRI Media Module can support one integrated voice/data endpoint or up to 2 BRI stations and/or data modules. Supported endpoints must conform to AT&T BRI, World Class BRI, or National ISDN NI1/NI2 BRI standards. The MM720 BRI Media Module provides -40 volt phantom power to the BRI endpoints.

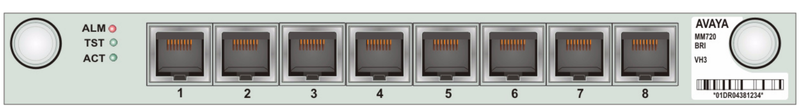
MM720 Media Module provides two 4-wire S/T ISND BRI. The maximum range of the S/T interface is 1800 feet between the NT and the TE.

Information is communicated in two ways:

- Over two 64-kbps channels, called B1 and B2, that can be circuit-switched simultaneously
- Over a 16-kbps channel, called the D-channel, that is used for signaling

The circuit-switched connections have an A-law or Mu-law option for voice operation. The circuit-switched connections operate as 64-kbps clear channels when in the data mode.

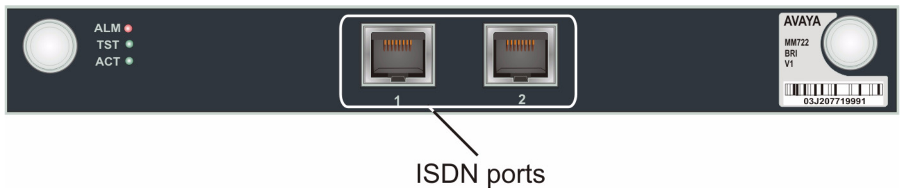
The MM720 is supported in the G350, G450, G430, and G700 Media Gateways.



MM722 BRI Media Module

The Avaya MM722 Media Module supports up to two BRI interfaces. MM722 can be configured on the TE side. Each port interfaces to the central office at the ISDN T reference point. Information is communicated in the same manner as for the MM720.

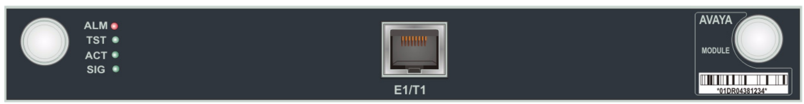
MM722 is supported in the G700, G450, G430, and G350 Media Gateways.



MM340 E1/T1 data WAN Media Module

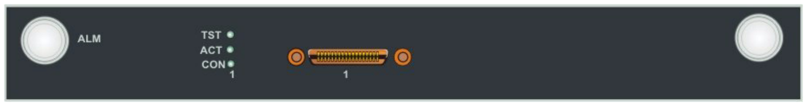
The Avaya MM340 Media Module provides one WAN access port for the connection of an E1 or T1 data WAN. The MM340 may be deployed as an interface to an IP-routed private enterprise network or as an interface to an Internet service provider.

MM340 E1/T1 data WAN Media Module is not supported in the G700 and G430 Media Gateways.



MM342 USP data WAN Media Module

The Avaya MM342 Media Module provides one USP WAN access port. The MM342 may be deployed as an interface to an IP-routed private enterprise network or as an interface to an Internet service provider.



MM342 is not supported in the G700 and G430 Media Gateways.

The MM342 supports the following WAN protocols:

- EIA530
- V.35/ RS449
- X.21

For these connections, one of the following cables is necessary:

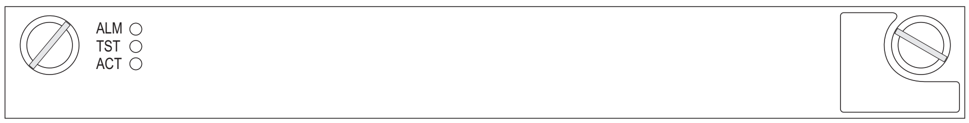
- Avaya Serial Cable DTE V.35 (USP to V.35)
- Avaya Serial Cable DTE X.21 (USP to X.21)

MM760 VoIP Media Module

The Avaya MM760 VoIP Media Module is a clone of the motherboard VoIP engine. MM760 provides additional 64 VoIP channels with G.711 compression.

MM760 VoIP Media Module is supported only in the G700 Media Gateway.

Figure 49: Avaya MM760 VoIP Media Module



Detailed description

The capacity of the MM760 is 64 G.711 TDM/IP simultaneous calls or 32 compression codec, G.729, G.726, 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:

Some customers might want an essentially nonblocking system. You must add an additional MM760 Media Module if the customer uses more than two MM710 Media Modules in a single chassis. The additional MM760 provides an additional 64 channels.

Ethernet interface

The MM760 must have its own Ethernet address. The MM760 requires a 10/100 Base T Ethernet interface to support H.323 endpoints for DEFINITY IP trunks and stations from another G700 Media Gateway.

Voice compression

The MM760 has resources for compression and decompression of voice for G.711 (A-Law and Mu-Law), G.729 and 729B, G.726, and G.723 (5.3K and 6.3K).

The VoIP engine supports the following functionality:

- RTP and RTCP interfaces
- Dynamic jitter buffers

- DTMF detection
- Hybrid echo cancellation
- Silence suppression
- Comfort noise generation
- Packet loss concealment
- SRPT media encryption

The MM760 also supports the following types of transmissions:

- Fax, Teletypewriter device (TTY), and modem calls over a corporate IP intranet using pass-through mode
- Fax and TTY calls using proprietary relay mode

  Faxes sent to nonAvaya endpoints cannot be encrypted.
- 64kbps clear channel transport in support of BRI Secure Phone and data appliances
- T.38 Fax over the Internet (including endpoints connected to nonAvaya systems)
- Modem tones over a corporate IP intranet

  Note:

  The path between endpoints for fax and modem tone transmissions must use Avaya telecommunications and networking equipment.

For more information, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

Telephony Interface Modules

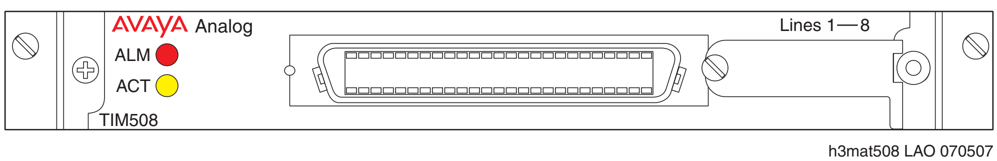
IG550 supports the following telephony interface modules:

- TIM508 Analog
- TIM514 Analog
- TIM516 Analog
- TIM518 Analog
- TIM510 E1/T1
- TIM521 BRI

TIM508 analog media module

The TIM508 Analog Telephony Interface Module provides eight analog telephone ports. You can alternatively administer some or all of the ports as analog DID trunks.

Figure 50: The TIM508 Analog Telephony Interface Module



Configuring TIM508 line ports

The TIM508 provides the capability to configure any of the eight line ports as:

- A wink-start or an immediate-start DID trunk
- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication

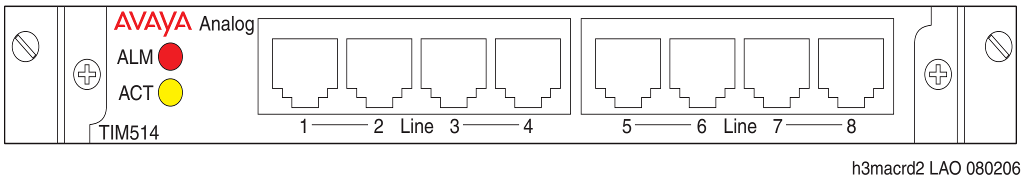
TIM508 also supports:

- Three ringer loads, which is the ringer equivalency number for up to 2,000 feet (610 meters) for the four station ports
- Up to eight ports ringing simultaneously
- Type 1 caller ID and Type 2 caller ID for line ports
- Ring voltage generation for a variety of international frequencies and cadences

TIM514 analog media module

The TIM514 Analog Telephony Interface Module provides four analog telephone ports and four analog trunk ports. You can only use the four analog line ports, ports 1 through 4, for analog DID trunks. The four analog trunk ports, ports 5 through 8, must not be used in this way.

Figure 51: The TIM514 Analog Telephony Interface Module



Configuring TIM514 trunk ports

The TIM514 provides the capability to configure ports 5 through 8 as:

- A loop start or a ground start central office trunk with a loop current of 18 to 120 mA
- A two-wire analog Outgoing CAMA E911 trunk, for connectivity to the PSTN. MF signaling is supported for CAMA ports
- Direct Inward/Outward Dialing (DIOD) for Japan only

Configuring TIM514 line ports

The TIM514 provides you with the capability to configure ports1 through 4 as:

- A wink-start or an immediate-start DID trunk
- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication

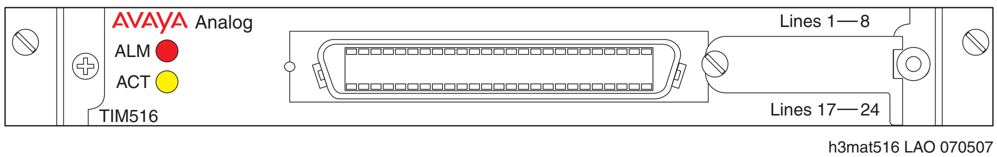
TIM514 also supports:

- Three ringer loads, which is the ringer equivalency number for up to 2,000 feet (610 meters) for all four line (station) ports
- Up to four ports ringing simultaneously
- Type 1 caller ID and Type 2 caller ID
- Ring voltage generation for a variety of international frequencies and cadences

TIM516 analog media module

The TIM516 Analog Telephony Interface Module provides 16 analog telephone ports.

Figure 52: The TIM516 Analog Telephony Interface Module



Configuring TIM516 line ports

The TIM516 provides the capability to configure any of the line ports as:

- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication

  Note:
  The TIM516 does not support Off Premise Stations (OPS) or DID/DIOD trunks.

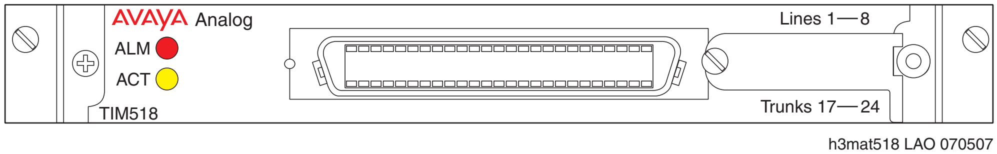
TIM516 also supports

- Three ringer loads, which is the ringer equivalency number for up to 2,000 feet (610 meters) for all sixteen ports
- Up to 16 ports ringing simultaneously
- Type 1 caller ID and Type 2 caller ID for line ports
- Ring voltage generation for a variety of international frequencies and cadences

TIM518 analog media module

The TIM518 Analog Telephony Interface Module provides eight analog telephone ports and eight analog trunk ports. Some or all of the line ports can be administered as analog DID trunks instead.

Figure 53: The TIM518 Analog Telephony Interface Module



Configuring TIM518 line ports

The TIM518 provides you with the capability to configure any of the first eight line ports as:

- A wink-start or an immediate-start DID trunk
- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication

Configuring TIM518 trunk ports

The TIM518 provides the capability to configure ports 9 through 16 as:

- A loop-start or a ground-start central office trunk with a loop current of 18 to 120 mA
- A two-wire analog Outgoing CAMA E911 trunk, for connectivity to the PSTN. MF signaling is supported for CAMA ports.

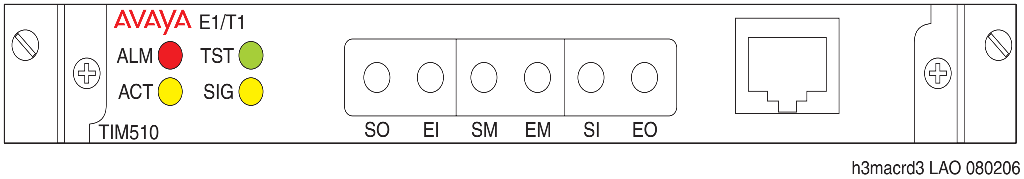
TIM518 also supports:

- Three ringer loads, which is the ringer equivalency number for up to 2,000 feet (610 meters) for all eight ports
- Up to eight ports ringing simultaneously
- Type 1 caller ID and Type 2 caller ID for line ports
- Type 1 caller ID for trunk ports
- Ring voltage generation for a variety of international frequencies and cadences

TIM510 E1/T1 Telephony Interface Module

The TIM510 T1/E1 Telephony Interface Module terminates a T1 or E1 trunk. The TIM510 has a built-in Channel Service Unit (CSU) so an external CSU is not necessary. The CSU is only used for the T1 circuit.

Figure 54: The TIM510 Telephony Interface Module



TIM510 supports the following features:

- DS-1 level support for a variety of E1 and T1 trunk types
- Trunk signaling to support United States and international CO or tie trunks
- Echo cancellation in either direction

TIM521 BRI Telephony Interface Module

The TIM521 BRI Telephony Interface Module provides four ports with RJ-45 jacks that can be administered as BRI trunk connections.

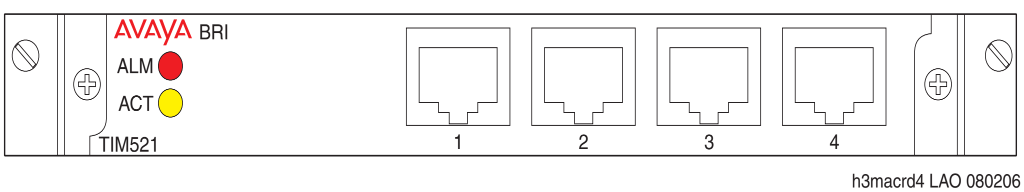
The TIM521 supports up to four BRI interfaces to the central office at the ISDN TE reference point. Information is communicated over each port in two ways:

- Over two 64-kbps channels, called B1 and B2, that can be circuit-switched simultaneously
- Over a 16-kbps channel, called the D-channel, that is used for signaling. The TIM521 occupies one time slot for D-channel use, regardless of whether one, two, three, or four D-channels are in use.

The circuit-switched connections have an A-law or Mu-law option for voice operation. The circuit-switched connections operate as 64-kbps clear channels when in the data mode.

Each port interfaces to the central office at the ISDN T reference point.

Figure 55: The TIM521 Telephony Interface Module



Note:

The TIM521 module does not support BRI stations, video endpoints, or combining both B channels together to form a 128-kbps channel.

Juniper Physical Interface Modules for serial and WAN connectivity

For more information on optional Juniper Physical Interface Modules, see *J2320, 2350, J4350 and J6350 Services Router Getting Started Guide*, Release 8.4.

Deskphones and softphones

Avaya IP Deskphones

1110 IP Deskphone

The Avaya 1110 IP deskphone is designed to meet standard communication needs for use in places such as schoolroom, main lobby or conference center environments.

1110 IP deskphone has the following characteristics:

- Integrated with an IEEE 10/100 Base-T Ethernet switch with LAN and PC ports and a fully backlit display
- Integrated listen-only speaker
- Four-way navigation cluster with Enter key
- Supports 802.3af standard-based PoE or local AC power via a global power supply
- Wall mountable
- Eight fixed keys such as Hold, Goodbye, Handsfree (listen only), Volume Up, Volume Down, Expand, Services and Inbox
- Four context-sensitive soft keys for easy to use navigation
- Supports dual media stream IP Call Recording with Avaya Contact Recording and Quality Monitoring (CRQM)
- Supports Static and Full Dynamic Host Control Protocol
- Supports E.164 dialing protocol

1120E IP Deskphone

The Avaya 1120E IP deskphone is designed for knowledge workers and administrative personnel. This is a multiline and intermediate-level desktop phone.

The four-line Avaya 1120E IP Deskphone supports standard-based Session Initiation Protocol (SIP).

Avaya 1120E IP deskphone has the following characteristics:

- High-resolution grayscale display
- Four-way navigation cluster with “Enter” key
- Supports 802.3af standard-based PoE or local AC power via a global power supply

- Secured communications with standard-based signaling encryption, media encryption, and user-based authentication for network access control
- Supports converged (voice and data) applications via External Application Server APIs
- Supports both Avaya Communication Server protocol (UNIStim) and RFC 3261 compliant protocol
- Session Initiation Protocol (SIP) firmware for business telephony
- Backlit LCD display with local contrast settings for enhanced viewing
- Wall mountable
- Supports transmission of duplicate media streams with Avaya Call Recording Quality Monitoring (CRQM)
- Integrated 10/100/1000 Base-T Auto-Sensing Ethernet switch for shared PC access (one LAN port and one PC port)
- Manually configurable for 10 and 100 mbps speeds when used with Ethernet Switches which do not support auto-sensing capabilities
- Minimum Category 5e cabling required for Gigabit Ethernet deployment (Category 5e cable included as standard)

1140E IP Deskphone

The Avaya 1140E IP deskphone is designed for managers, executives, knowledge workers and senior administrative staff. This is a multiline, and professional-level deskphone.

The Avaya 1140E IP deskphone has the following characteristics:

- Delivers enhanced communication capabilities
- Large high-resolution graphical grayscale display with an IEEE 10/100/1000 switch with LAN and PC ports, an integrated Bluetooth audio gateway, and a standard USB port.
- Four-way navigation cluster with Enter key
- High-resolution, fully-backlit, graphical, eight-level grayscale, 240 x 160 pixel display with anti-glare screen
- Supports 802.3af standard-based PoE or local AC power via a global power supply
- Secured communications with standard-based signaling encryption, media encryption, and user-based authentication for network access control
- Proactive Voice Quality Management (PVQM) for enhanced administration and diagnostics
- Supports both Avaya Communication Server protocol (UNIStim) and RFC 3261 compliant protocol
- Session Initiation Protocol (SIP) firmware for business telephony

- Fourteen fixed keys such as Handsfree, Headset, Volume Up, Volume Down, Mute, Hold, Goodbye, Directory, Inbox/Message, Outbox/Shift, Quit, Copy, Services and Expand
- Four context-sensitive soft keys for easy to use navigation
- Supports transmission of duplicate media streams with Avaya Call Recording and Quality Monitoring (CRQM)
- Supports 802.1x and Extensible Authentication Protocol (EAP-MD5) for network authentication and access control
- Supports E.164 dialing and SIP Protocols

1150E IP Deskphone

The Avaya 1150E IP Deskphone is specifically optimized for call-intensive environments such as IP ACDs and contact centers. The 1150E is a multiline phone.

The 1150E IP Deskphone deskphone has the following characteristics:

- High-resolution grayscale display with an IEEE 10/100/1000 switch with LAN and PC ports, an integrated Bluetooth audio gateway for agent support, and a standard USB connection.
- Seven additional fixed keys with built in LEDs dedicated to IP Contact Center agents including: In-Calls, Not Ready, Make Set Busy, Supervisor, Supervisor Listen/Talk, Emergency, and Activity
- Four-way navigation cluster with “Enter” key
- Dual headset jacks for agent support and supervisor plug-in
- 802.3af Power over Ethernet as standard or local power option via a separately orderable AC power adapter
- Visual Alerting/Message Waiting Indication LED for incoming call and voice message pending notification
- Supports 802.1ab Link Layer Discovery Protocol
- Tight linkage with Avaya Communication and Application Servers, offering a rich suite of reliable, business-grade telephony, and IP Contact Center communication and application features
- 12 fixed business telephony keys including Hold, Goodbye, Volume Up, Volume Down, Mute, Directory, Message/Inbox, Shift/Outbox, Quit, Copy, Expand, and Services
- Four context-sensitive soft keys
- Seven dedicated Agent fixed keys with built-in LEDs
- Seven dedicated Supervisor fixed keys with built-in LEDs
- Supports transmission of duplicate media streams with Avaya Contact Recording and Quality Monitoring (CRQM)

- Supports IEEE 802.3af power standard, prestandard Avaya and Cisco proprietary powering schemes
- Supports 802.1ab Link Layer Discovery Protocol (LLDP) for network discovery and inventory management
- Static and Full Dynamic Host Control Protocol settings (Full DHCP factory default)
- Audio quality of service: G.711 a-law, G.711 µ-law, G.729a and Annex B

1165E IP Deskphone

The Avaya 1165E IP Deskphone is multiline, professional-level deskset. It offers a high-resolution, fully-backlit QVGA color LCD display, an integrated Bluetooth audio gateway, and an integrated phone switch with Gigabit Ethernet LAN and PC ports.

1100 Series Expansion Module

The Avaya 1100 Series Expansion Module adds 18 self-programmable line or feature keys to the 1120E, 1140E, and 1150E IP Deskphones.

The 1100 series expansion module has the following characteristics:

- Line/programmable feature expansion for 1120E IP Deskphone, 1140E IP Deskphone, 1150E IP Deskphone, and 1165E IP Deskphone
- 18 additional line/programmable feature keys per module, extending desktop investment in 1100 Series IP Deskphones with added scalability
- Up to three Expansion Modules 1100 Series IP Deskphone
- High-resolution, pixel-based, graphical grayscale display for presentation of enhanced graphical content
- 802.3af Power over Ethernet or local AC power options to deliver maximum customer flexibility in deployment
- Wall mountable using standard jack pins or mounting screws
- Tight integration with Avaya Communication Server 1000 feature for rich and reliable service
- Firmware download support via Trivial File Transfer Protocol (TFTP) or UNIStim File Transfer Protocol (UFTP)

1210 IP Deskphone

The Avaya 1210 IP Deskphone is standard-level desktop phone designed for places such as lobbies, lunch rooms, and other common areas.

The 1210 IP deskphone has the following characteristics:

- Single-line IP Phone supports up to eight fixed telephony keys, four context-sensitive soft keys, and two shortcut/feature keys
- 3-lines x 24 character and 143 x 32 pixel display with anti-glare screen combined with a flexible two-position adjustable footstand to optimize viewing under varied lighting conditions
- Four-way navigation cluster with Enter key to maximize user choice and flexibility in navigation
- Superior audio quality to leverage some of the latest technologies and to ensure crystal-clear conversations
- Integrated headset and speakerphone supports a high-quality, two-way speakerphone (handsfree) and other optional headsets for executives or multi-tasking workers
- Integrated 10/100 Base-T Ethernet switch with LAN and PC ports
- Supports 802.3af standard-based PoE or local AC power via power adapter
- Ten fixed keys such as Handsfree, Headset, Volume Up, Volume Down, Mute, Hold, Goodbye, Applications, Services, and Conference
- Four context-sensitive soft keys for easy-to-use navigation

1220 IP Deskphone

The Avaya 1220 IP deskphone is designed for office workers with moderate call activity. It offers four softkeys and six shortcut keys.

The 1220 IP deskphone has the following characteristics:

- Multiline IP Deskphone supports up to 4-line or programmable feature keys, eight fixed telephony keys, four context-sensitive soft keys, and six shortcut/feature keys
- 5-lines x 25 characters display with anti-glare screen combined with a flexible two-position adjustable footstand optimizes viewing under varied lighting conditions
- Four-way navigation cluster with Enter key
- Integrated 10/100 Base-T Ethernet switch with LAN and PC ports
- Supports 802.3af standard-based PoE or local AC power via power adapter

1230 IP Deskphone

The Avaya 1230 IP deskphone is a premium, intermediate deskphone designed for managers, knowledge workers, and administrative assistants with advanced communication needs. It has 10 programmable soft keys, preprogrammed with key features.

1200 Series Expansion Module

The Avaya 1200 series expansion module is available as an 18-key LED KEM with paper labels or as a 12-key LCD KEM that is self-labeling. Compatible with the 1220 and 1230 models, it delivers increased scalability for investment protection and expanded feature access for user flexibility.

1603 IP Deskphone

The Avaya 1603 IP telephone is designed for the walk-up user and the everyday user. It is ideal for common areas in offices, stockrooms, lobbies, or drop-in desks. Visitors, including customers, are examples of walk-up users who need a simple, familiar interface. 1603 IP telephone is also perfect for the everyday phone user for whom a phone is not the one critical piece of their communication needs.

The Avaya 1603 IP telephone has the following characteristics:

Hardware

- Each button includes dual LEDs (red, green) providing explicit status for the user
- Includes fixed feature keys for common telephone tasks including conference, transfer, drop, hold, mute
- 2-way speakerphone
- 2-lines by 16 character display, backlit for easier viewing in all lighting conditions
- 3-line appearance/feature key buttons, with dual LEDS (red, green)
- Full-duplex speakerphone
- Message waiting indicator
- Dual position flip stand
- Four-way navigation cluster button
- Three contextual softkey buttons
- Volume button separate volume levels in the handset, headset, speaker, and ringer
- Quick-access Voicemail Message button
- Telephony application button to return to main telephone screen
- Avaya menu button (options and settings access)
- Buttons for Contacts, Call log, Redial, Speaker, Mute, Headset, Hold, Conference, Transfer, Drop
- Ethernet (10/100) line interface with a secondary 10/100 port for colocated laptop or PC
- PoE 802.3af class 2 device, also supports a local power supply
- Headset interface

- Wall mount kit available

Software

- Contacts application that supports up to 100 entries
- Call log that contains last 100 calls
- H.323 protocol support
- Standard-based codec support: G.711, G.726, G.729A/B
- Supports languages such as English, French, Spanish, German, Italian, Dutch, Portuguese, Russian

Requirements

- Communication Manager Release 3.0 or later releases
- Local or centralized electrical power, through an 802.3af switch or local power supply
- HTTP file server

1603SW-I SIP Telephone

The 1603SW-I telephone is a multiline SIP telephone for use by enterprises with Avaya Aura® Communication Manager and Avaya Aura® Session Manager call processing systems or enterprises with Avaya Aura® Midsize Business Template call processing system.

1603SW IP Deskphone

The Avaya 1603SW IP deskphone is designed for users with basic communication needs. This model includes all features of the 1603, but has an extra 10/100 Ethernet port for a colocated laptop or system.

1608 IP Telephone

The Avaya 1608 IP Telephone is designed for the everyday user who typically relies on several forms of communication, including voice and email, and expects a quality telephone but receives no more than five or six calls per day.

1608 IP Telephone supports 8-line appearances / feature keys. Each button includes dual LEDs (red, green) providing explicit status for the user. For a familiar look and feel, 1608 IP Telephone includes several fixed feature keys for common telephone tasks including conference, transfer, drop, hold, and mute. In addition, 1608 IP Telephone includes a high quality, full-duplex speakerphone, and supports a broad portfolio of Avaya wired and wireless headsets through its integrated headset jack.

1608 IP Telephone features a context-sensitive user interface, along with three softkeys and a four-way navigation cluster that is ideal for scrolling through the local contacts list or call logs.

The display on the 1608 IP Telephone measures 3-lines by 24 characters and is backlit for easier viewing in all lighting conditions.

The Avaya 1608 IP telephone has the following characteristics:

Hardware

- Backlit display — 3.5’ diagonal, 3 rows by 24 characters
- Ergonomic hearing aid compatible handset — supporting TTD acoustic coupler
- Full-duplex speakerphone
- 8-line appearance/feature key buttons, with dual LEDS (red, green)
- Full-duplex speakerphone
- Message waiting indicator
- Dual position flip stand
- Four-way navigation cluster button
- Three contextual softkey buttons
- Volume button separate volume levels in the handset, headset, speaker, and ringer
- Quick-access Voicemail Message button
- Telephony application button to return to main telephone screen
- Avaya menu button options and settings access
- Buttons for Contacts, Call log, Redial, Speaker, Mute, Headset, Hold, Conference, Transfer, Drop
- Ethernet (10/100) line interface with a secondary 10/100 port for colocated laptop or PC
- PoE 802.3af class 2 device, also supports a local power supply
- Headset interface
- Wall mount kit available

Software

- Contacts application that supports up to 100 entries
- Call log that contains last 100 calls
- H.323 protocol support
- Standard-based codec support: G.711, G.726, G.729A/B
- Supports languages such as English, French, Spanish, German, Italian, Dutch, Portuguese, Russian

Requirements

- Communication Manager Release 3.0 or later releases

- Local or centralized electrical power, through an 802.3af switch or local power supply
- HTTP file server

1616 IP Telephone

The Avaya 1616 supports 16-line appearances/feature keys on the phone itself and a 32 button expansion model provides access to a total of 48 feature keys or speed dial buttons. Each button features a dual LED (red, green) providing explicit status for the user. For a familiar look and feel, the 1616 includes several fixed feature keys for common telephone tasks including conference, transfer, drop, hold, and mute. In addition, 1616 includes a high quality, full-duplex speakerphone, and supports a broad portfolio of Avaya wired and wireless headsets through its integrated headset jack.

The 1616 features a context-sensitive user interface, along with three softkeys and a four-way navigation cluster that is ideal for scrolling through the local contacts list or call logs. The viewing angle of the display on the 1616 is adjustable and measures 4-lines by 24 characters. 1616 IP Telephone displays additional caller-related information with active appearances for easier call handling. The display is backlit for easier viewing in all lighting conditions.

The Avaya 1616 IP telephone has the following characteristics:

Hardware

- Backlit display — 3.5' diagonal, 4 rows by 24 characters with adjustable display angle
- Ergonomic hearing aid compatible handset supporting TTD acoustic coupler
- Full-duplex speakerphone
- 16-line appearance/feature key buttons with dual LEDS (red, green)
- Full-duplex speakerphone
- Message waiting indicator
- Dual position flip stand
- Four-way navigation cluster button
- Three contextual softkey buttons
- Volume button separate volume levels in the handset, headset, speaker, and ringer
- Message button
- Telephony application button to return to main telephone screen
- Avaya menu button options and settings access
- Buttons for Contacts, Call log, Redial, Speaker, Mute, Headset, Hold, Conference, Transfer, Drop
- Ethernet (10/100) line interface with a secondary 10/100 port for colocated laptop or PC
- Module interface for 32 button expansion module

- PoE 802.3af class 3 device, also supports a local power supply
- Headset interface
- Wall mountable
- Optional Gigabit Adapter for Gigabit connectivity to a PC

Software

- Contacts application that supports up to 100 entries
- Call log that contains last 100 calls
- H.323 protocol support
- Standard-based codec support: G.711, G.726, G.729A/B
- Supports languages such as English, French, Spanish, German, Italian, Dutch, Portuguese, Russian

Requirements

- Communication Manager Release 3.0 or later releases
- Local or centralized electrical power, through an 802.3af switch or local power supply
- HTTP file server

2007 IP Deskphone

The Avaya 2007 IP telephones deliver telephony features and applications for the business requirement. The Avaya 2007 IP Deskphone is designed for managers, executives, knowledge workers, and for the user who uses vertical applications that leverage converged voice and data applications.

This advanced multiline IP phone has a large color touch screen display and integrated 10/100 switch.

The phones are supported on the Avaya Business Communications Manager 400, Communication Server 1000, and Communication Server 2100 platforms.

Avaya one-X® Deskphone family of IP Telephones

9620/9620L/9620C IP Telephone

The 9620 IP telephone is specifically designed for the everyday telephone users who rely on multiple communications tools, such as email and instant messaging, and require a high-quality, intuitive telephone for voice communications. The high fidelity audio of 9620 IP telephone provides crystal clear sound, eliminating background noise. The backlit display and intuitive interface simplifies access to critical telephone features, such as one-touch dialing from the contact list and accessing recent call information from call logs.

Through its integrated web browser and application interface, 9620 supports productivity-enhancing phone applications such as corporate LDAP directories and the receipt of broadcast reminders and alerts.

9620 IP telephone has the following characteristics:

Hardware

- Backlit display 3.45" diagonal 1/4 VGA quality gray-scale pixel based with adjustable display angle
- Full-duplex speaker phone
- Ergonomic wideband hearing aid compatible handset supporting TTD acoustic coupler
- Two message waiting indicators
- Flip-stand/dual position
- Wall mountable
- Four-way navigation cluster button
- Four contextual softkey buttons
- Volume button separate volume levels in the handset, speaker, and ringer
- Avaya menu button browser, options and settings access
- Message button (LED)
- Telephony application (hard button)
- Mute button (LED)
- Speaker button (LED)
- Headset button (LED)
- Contacts button
- Call log button (LED)
- 3-line appearance LEDs
- Ethernet (10/100) line interface with secondary Ethernet interface
- Module interface for future modules such as speakerphone module
- POE 802.3af compliant class 2 device with auxiliary power
- Adapter interface
- USB interface
- Wideband headset interface

Software

- Supports 12 call appearances or administrable feature keys
- Contacts application (250 entry) with hard button
- 100-entry call log with hard button or LED
- H.323 protocol with future support for SIP
- Standard-based wideband codec G.722 and the following narrow band codecs: G.711, G.726, G.729A/B
- Supports the Avaya Push API application interface for third-party telephone applications
- Supports languages such as English, Canadian French, Parisian French, Latin American Spanish, Castilian Spanish, German, Italian, Dutch, Brazilian Portuguese, Japanese (Kanji, Hiragana, Katakana), Simplified Chinese, Korean, Russian Cyrillic, and Hebrew

9630G IP Telephone

Avaya 9630G IP telephone delivers advanced communications capabilities in a solution designed for those who are absolutely dependent on voice communications for their business operations.

9630G IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The backlit display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants.

Through its integrated web browser and application interface, 9630 supports productivity enhancing phone applications such as LDAP corporate directories and integration with Microsoft Outlook calendars.

Avaya 9630G IP telephone has the following characteristics:

Hardware

- Backlit display —- 3.8" diagonal 1/4 VGA quality gray-scale pixel based with adjustable display angle
- 6-line appearance buttons with LEDs
- Full-duplex wideband speaker phone
- Ergonomic wideband Hearing aid compatible handset supporting TTD acoustic coupler
- Two message waiting indicators
- Innovative dual position flip-stand
- Wall mountable
- Four-way navigation cluster button

- Four contextual softkey buttons
- Forward/mobility button (LED)
- Volume button separate volume levels in the handset, speaker, and ringer
- Avaya menu button browser, options, and settings access
- Message button (LED)
- Telephony application (hard button)
- Mute button (LED)
- Speaker button (LED)
- Headset button (LED)
- Contacts button
- Call log button (LED)
- Ethernet (10/100) line interface with secondary Ethernet interface
- Module interface to support add-ons
- Supports one 24-button expansion module up to three with Communication Manager 4.0 and later releases, when available
- POE 802.3af compliant class 2 device with auxiliary power
- Two adapter interfaces
- USB interface
- Wideband headset interface

Software

- Supports 24 call appearances or administrable feature keys
- 250-entry contacts application (hard button)
- 100-entry call log with hard button or LED
- H.323 protocol with future support for SIP
- Standard-based wideband codec G.722 and the following narrow band codecs: G.711, G.726, G.729A/B
- Supports the Avaya Push API application interface for third-party telephone applications
- Supports languages such as English, Canadian French, Parisian French, Latin American Spanish, Castilian Spanish, German, Italian, Dutch, Brazilian Portuguese, Japanese (Kanji, Hiragana, Katakana), Simplified Chinese, Korean, Russian Cyrillic, and Hebrew

9640/9640G IP Telephone

Avaya 9640 IP telephone delivers advanced communications capabilities in a solution designed for those who are absolutely dependent on voice communications for their business operations.

The 9640 IP telephone supports a high-resolution color display with integrated web browser and application interface. It is the ideal telephone to support productivity enhancing phone applications such as LDAP corporate directories, integration with Microsoft Outlook calendars, and surveillance cameras/web cams (refreshed still images).

The 9640 IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The color display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants.

Avaya 9640 IP telephone has the following characteristics:

Hardware

- Color display 3.8" diagonal 1/4 VGA quality gray-scale pixel based with adjustable display angle
- 6-line appearance buttons with LEDs
- Full-duplex wideband speaker phone
- Ergonomic wideband hearing aid compatible handset supporting TTD acoustic coupler
- Two message waiting indicators
- Innovative dual position flip-stand
- Wall mountable
- Four-way navigation cluster button
- Four contextual softkey buttons
- Forward/mobility button (LED)
- Volume button with separate volume levels in the handset, speaker, and ringer
- Avaya menu button browser, options and settings access
- Message button (LED)
- Telephony application (hard button)
- Mute button (LED)
- Speaker button (LED)
- Headset button (LED)
- Contacts button
- Call log button (LED)

- Ethernet (10/100) line interface with secondary Ethernet interface
- Module interface to support add-ons
- Supports one 24-button expansion module up to three with Communication Manager 4.0 and later releases, when available
- POE 802.3af compliant class 2 device with auxiliary power
- Two adapter interfaces
- USB interface
- Wideband headset interface

Software

- Supports 24 call appearances or administrable feature keys
- 250-entry contacts application (hard button)
- 100-entry call log with hard button or LED
- H.323 protocol with future support for SIP
- Standard-based wideband codec G.722 and the following narrow band codecs: G.711, G.726, G.729A/B
- Supports the Avaya Push API application interface for third-party telephone applications
- Supports the following languages: English, Canadian French, Parisian French, Latin American Spanish, Castilian Spanish, German, Italian, Dutch, Brazilian Portuguese, Japanese (Kanji, Hiragana, Katakana), Simplified Chinese, Korean, Russian Cyrillic, and Hebrew

9650 IP Telephone

Avaya 9650 IP telephone delivers advanced communications capabilities in a solution designed specifically for those who are employed to speak on the telephone for the majority of the day positions such as, building receptionists and executive administrative assistants.

9650 IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The backlit display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants. 9650 supports built-in button module functionality (16 buttons) with the user interface enhanced to provide simple one-touch access to bridged appearances, speed dials, and feature keys.

Through its integrated web browser and application interface, the 9650 supports productivity enhancing phone applications such as LDAP corporate directories and integration with Microsoft Outlook calendars.

Avaya 9650 IP telephone has the following characteristics:

Hardware

- Color display 3.8" diagonal 1/4 VGA quality gray-scale pixel based with adjustable display angle
- 3-line appearance buttons with LEDs
- Eight additional auxiliary buttons used as line appearances or feature keys
- One aux shift button
- Full-duplex wideband speaker phone
- Ergonomic wideband hearing aid compatible handset supporting TTD acoustic coupler
- Two message waiting indicators
- Innovative dual position flip-stand
- Wall mountable
- Four-way navigation cluster button
- Four contextual softkey buttons
- Forward/mobility button (LED)
- Volume button separate volume levels in the handset, speaker, and ringer
- Avaya menu button browser, options and settings access
- Message button (LED)
- Telephony application (hard button)
- Mute button (LED)
- Speaker button (LED)
- Headset button (LED)
- Contacts button
- Call log button (LED)
- Ethernet (10/100) line interface with secondary Ethernet interface
- Module interface to support add-ons
- Supports one 24-button expansion module up to three with Communication Manager 4.0 and later releases, when available
- POE 802.3af compliant class 2 device with auxiliary power
- Two adapter interfaces
- USB interface
- Wideband headset interface

Software

- Supports 24 call appearances or administrable feature keys
- 250-entry contacts application (hard button)
- 100-entry call log with hard button or LED
- H.323 protocol with future support for SIP
- Standard-based wideband codec G.722 and the following narrow band codecs: G.711, G.726, G.729A/B
- Supports the Avaya Push API application interface for third-party telephone applications
- Supports languages such as, English, Canadian French, Parisian French, Latin American Spanish, Castilian Spanish, German, Italian, Dutch, Brazilian Portuguese, Japanese (Kanji, Hiragana, Katakana), Simplified Chinese, Korean, Russian Cyrillic, and Hebrew

9670G IP Deskphone (only for Avaya Aura®)

Avaya 9670G IP deskphone (only for Avaya Aura®) featured with a large touchscreen provides access to contacts and applications. An onscreen keyboard feature increases the usability of 9670G. All these features enhance employee productivity.

Avaya 1600-Series/9600-Series Specialty Handsets

Avaya offers one Amplified Speech handset for the 1600-Series IP telephones and one for the 9600-Series IP telephones. Both handsets operate in two modes:

- Mode 1: Push-To-Amplify Receive Volume

  In this mode, while the Push-To-Amplify button is pressed, the receive volume is at the level that has been set by the volume selection button. When the Push-To-Amplify button is not pressed, the receive volume is at the minimum handset volume level.
- Mode 2: Always Amplify receive Volume (Default Mode)

  In this mode, the receive volume is always at the level that has been set by the volume selection button. Pressing the Push-To-Amplify button has no effect.

The following table indicates the model number, description, order code, and Avaya IP telephone models that work with each handset. You can find the handset model number printed on a label in the battery compartment.

Handset Model	Description	Order Code	Works with phones
S1K5	9600-Series Amplified Speech	700446370	9600-Series
S1K6	1600-Series Amplified Speech	700446388	1600-Series

Avaya 4600-Series IP Telephones

Avaya 4601 IP Telephone

The Avaya 4601 is an entry-level IP telephone with 2 call appearances.

The 4601 IP telephone has the following characteristics:

- 2 call appearances with LEDs
- Fixed button with LED for voice mail retrieval
- Five fixed feature buttons that include Hold, Transfer, Conference, Drop, Redial
- Supports power over Ethernet
- Supports Quality-of-Service features including RTCP and RSVP
- Wall or desk mountable
- 10/100Base-T Ethernet network connection with RJ-45 interface
- Supports G.711, G.729A, and G.729B audio voice codecs
- Supports H.323 V2, except for automatic unnamed registration
- IP address assignment using DHCP
- Downloadable firmware for future upgrades
- Native support that provides the user with the capability to administer and maintain the 4601 IP telephone without using an alias
- 12 button, touch-tone dial pad with raised bar on the button labeled five for the visually impaired
- Message waiting light (LED)
- Hearing aid compatible
- Adjustable volume control

- Available in dark gray

Avaya 4602 IP Telephone

Avaya 4602 is an entry-level IP telephone with 2 call appearances.

Avaya 4602 IP telephone has the following characteristics:

- 2-line × 24-character display
- Two call appearances
- One fixed button for voice mail retrieval
- One-way speaker
- Seven fixed feature buttons include Speaker, Mute, Hold, Transfer, Conference, Drop, Redial
- Supports Power over Ethernet (PoE)
- Supports Quality-of-Service (QOS) features, including RTCP and RSVP
- Wall or desktop mountable
- 10/100Base-T Ethernet network connection with an RJ-45 interface
- Supports G.711, G.729A, and G.729B audio voice codecs
- Supports H.323 V2
- Sends and receives messages using Session Initiation Protocol (SIP)

  Note:

  SIP support requires SIP firmware to be installed. The 4602 IP telephone cannot be administered for SIP and H.323 at the same time.
- Supports Web interface for phone settings (SIP-enabled only)
- IP address assignment using DHCP or statically configured
- Integrated Ethernet repeater hub for optional PC connection
- Downloadable firmware for future upgrades
- Native support that provides the user with the capability to administer and maintain the 4602 IP telephone without using an alias
- 12 button, touchtone dial pad with raised bar on button labeled five for the visually impaired
- Message waiting light (LED)
- Hearing aid compatible
- Adjustable volume control
- Available in dark gray

Avaya 4602SW IP Telephone

The 4602SW IP telephone has the same feature set as the 4602 with the addition of a built-in Ethernet switch instead of the hub.

Avaya 4610SW IP Telephone

Avaya 4610SW IP telephone provides advanced feature functionality with an intuitive and innovative user interface. Avaya 4610SW provides telephony, speed dial, call log, and Web browsing functionality.

Avaya 4610SW IP telephone has the following characteristics:

- High-end feature set
- Medium screen graphic display (168 x 80 pixel, 4 grayscale)
- Advanced user interface that supports 48 speed dialing buttons, 45 call log entries, and up to three redial buttons on display
- Avaya Call Processing label editing
- Speed Dial entry editing
- User screen options
- Call log
- WML browser capability
- Full-duplex speakerphone with echo cancellation
- 10/100Base-T Ethernet network connection with an RJ-45 interface
- Integrated Ethernet switch for an optional PC connection
- Supports G.711, G.729A, and G.729B audio voice coders
- Supports H.323 V2
- Receives and displays extensible markup language (XML) page content that is pushed from an application server
- Receives and plays streaming audio that is pushed from an application server
- IP address assignment using DHCP or statically configured
- Downloadable firmware for future upgrades
- 12 call appearance or feature buttons with downloadable labels
- Adjustable desk stand
- Global icons
- Hearing aid compatible
- 12 button, touchtone dial pad with raised bar on the number five key for the visually impaired

- Message waiting light (LED)
- Adjustable volume control
- Supports CTI applications from the Avaya Softphone and is CTI-ready for other applications
- Supports Power over Ethernet (PoE)
- Supports Quality-of-Service (QOS) features, including RTCP and RSVP
- Displays network audio quality information during calls
- Supports multibyte fonts
- Native support that gives you the ability to administer and maintain the telephone without using an alias
- Four softkeys, located under the display, that enhance the user interface
- Available in dark gray

Avaya 4620SW IP Telephone

Avaya 4620SW IP telephone provides advanced feature functionality with an intuitive and innovative user interface. Avaya 4620 telephone provides telephony, speed dial, call log, and Web browsing functionality.

Avaya 4620SW IP telephone has the following characteristics:

- Large screen graphic display (168-pixel by 132-pixel 4-grayscale)
- Supports multibyte fonts
- Advanced user interface that supports 108 speed dialing buttons, 90 call log entries, and up to 6 redial buttons on the display
- Avaya Call Processing label editing
- Speed Dial entry editing
- EU24 label-button editing
- User screen options
- Wireless Markup Language (WML) browser capability
- Full-duplex speakerphone with echo cancellation
- 10/100Base-T Ethernet network connection with an RJ-45 interface
- Supports G.711, G.729A, and G.729B audio voice coders
- Supports H.323 V2
- IP address assignment using DHCP or statically configured
- Receives and displays extensible markup language (XML) page content that is pushed from an application server

- Receives and plays streaming audio that is pushed from an application server
- Infrared (IR) port to support IR dialing and other applications
- Downloadable firmware for future upgrades
- 24 call appearance or feature buttons with downloadable labels
- Adjustable desk stand
- Function key expansion unit jack to support an optional 24-button feature expansion unit (EU24)
- Global icons
- Hearing aid compatible
- A 12 button, touch-tone dial pad with raised bar on the number five key for the visually impaired.
- A message waiting light (LED)
- Adjustable volume control
- Supports CTI applications from the Avaya Softphone and is CTI ready for other applications
- Supports Power over Ethernet (PoE)
- Supports Quality-of-Service (QOS) features, including RTCP and RSVP
- Displays network audio quality information during calls
- Native support that gives you the ability to administer and maintain the telephone without using an alias
- Four softkeys, located under the display, that enhance the user interface
- Available in dark gray

Avaya 4621SW IP Telephone

Avaya 4621SW IP telephone is based on the 4620SW IP telephone hardware. The two phones have almost similar user interface. The 4621SW telephone provides advanced feature functionality with an intuitive and innovative user interface. Avaya 4621SW telephone provides telephony, speed dial, call log, and Web browsing functionality.

The changes in the 4621SW are as follows:

- Large screen with backlit graphic display
- Adjustable backlight that you can administer to turn off when idle or may remain lit
- Does not support IR interface
- Stand with one extra height setting. This setting is the same as the highest setting for the 4610SW telephone

- Native support that gives you the ability to administer and maintain the telephone without using an alias
- Supports the EU24BL adjunct. The EU24BL is the same as the EU24 except that the former has a backlit display

Avaya 4622SW IP Telephone

Avaya 4622SW IP telephone is based on the 4620SW IP telephone hardware. 4622SW IP telephone provides the same advanced feature functionality with an intuitive and innovative user interface as the 4620SW IP telephone. 4622SW IP telephone is designed for the call center environment.

The changes in 4622SW are as follows:

- Does not have a handset or speakerphone microphone
- Has two headset jacks
- Has a large screen backlit graphic display
- Adjustable backlight that you can administer to turn off when idle or may remain lit
- Does not support IR interface
- Stand has one extra height setting. This setting is the same as the highest setting for the 4610SW telephone
- Native support that gives you the ability to administer and maintain the telephone without using an alias
- Supports the EU24BL adjunct. The EU24BL is the same as EU24 except has a backlit display

Avaya 4625SW IP Telephone

Avaya 4625SW IP telephone is similar to the Avaya 4620SW IP telephone. Avaya 4625SW provides advanced feature functionality with an intuitive and innovative user interface. Avaya 4625SW telephone provides telephony, speed dial, call log, and Web browsing functionality.

Avaya 4625SW IP telephone has all the applications and options of 4620SW IP telephone. The changes in 4625SW are as follows:

- Color 1/4-VGA backlit display
- Native support that gives the ability to administer and maintain the telephone without using an alias
- Does not support multibyte characters or multibyte User Interface languages
- Does not support an IR interface

Avaya 4630 IP Screenphone

Avaya 4630 Screenphone is a fully Internet-capable IP appliance that supports IP standards. Avaya 4630 IP Screenphone provides a user-friendly window into IP enabled applications, a full suite of Communication Manager features, Lightweight Directory Access Protocol (LDAP) directory, and voice mail features of INTUITY AUDIX. 4630 provides up to six telephony-related applications through a unique user interface that was developed for ease of use and minimal touch access.

Avaya 4630 Screenphone has the following characteristics:

- 1/4 VGA color touch-screen display with user screen options
- Five fixed feature buttons Speaker, Mute, Hold, Headset, Volume control
- Full-duplex speakerphone with echo cancellation
- 120 speed dial buttons that are organized into groups for easier access
- 100 total entries in the call log of incoming and outgoing calls
- Up to eight redial buttons can be presented on the display
- 10/100 Base-T Ethernet network connection with RJ-45 interface
- Directory access to corporate telephone directory information on an LDAP server
- Voice mail access to Web-based voice mail messaging capabilities of Avaya Web Messaging
- User-customizable stock ticker
- Access to Web-based information, including support for downloading Java applets
- G.711, G.729A, and G.729B audio voice coders
- H.323 V2
- IP address assignment using DHCP or statically configured
- Infrared (IR) port to support IR dialing and other applications
- Supports CTI applications from the Avaya Softphone and is CTI ready for other applications
- Supports Power over Ethernet (PoE)
- Supports Quality-of-Service (QOS) features, including RTCP and RSVP
- Displays network audio quality information during calls
- Downloadable firmware for future upgrades
- A built-in Ethernet switch
- Hearing aid compatible
- 12 button, touch-tone dial pad with raised bar on the number five key for the visually impaired

- Message waiting light (LED)
- Integrated modular headset jack for direct connection of headset
- Adjustable volume control for the handset, the speaker, and the ringer
- K-style handset with 9-foot modular cord
- 14-foot (4.27-meter) modular line cord
- Available in black or white

Optional available components:

- 12-foot (3.66 meter) modular handset cord
- 25-foot (7.62-meter) modular line cord
- Base stand
- Avaya headsets
- Amplifier handset
- Noisy environment handset
- Push-to-talk handset

Avaya Digital Deskphones

1408 Digital Deskphone

The Avaya 1408 Digital Deskphone is designed for cubicle workers, sales staff, and other users with relatively simple telephone needs. The 1408 delivers a straightforward, productivity enhancing interface.

The Avaya 1408 digital deskphone has the following characteristics:

- Supports eight administrable feature buttons
- Each button includes dual LEDs (red, green) that provide explicit status for the user
- Includes several fixed feature keys for common telephone tasks such as conference, transfer, drop, hold, and mute
- Includes high-quality speakerphone
- Supports a broad portfolio of wired and wireless headsets through its integrated headset jack
- Context sensitive user interface along with three softkeys and a four-way navigation cluster
- Wall-mountable

- The three-line by 24-character display is white backlit for easier viewing in all lighting conditions
- Graphical Display size is 181 x 40 pixels

1416 Digital Deskphone

The Avaya 1416 Digital Deskphone is designed for receptionists, assistants, managers, and other users who answer incoming calls, transfer customers, and monitor several lines throughout a typical day. This phone provides speed-dial buttons that improve the productivity and eliminate the need to scroll through on-screen lists.

The Avaya 1416 digital deskphone has the following characteristics:

- Supports 16 administrable feature buttons and 32-button expansion module provides access to a total of 48 feature keys or speed dial buttons
- Each button features a dual LED (red, green) providing explicit status for the user
- Includes a high-quality speakerphone and supports a broad portfolio of wired and wireless headsets through its integrated headset jack
- Context sensitive user interface along with three softkeys and a four-way navigation cluster
- Wall-mountable
- The viewing angle of the display is adjustable and measures 4-lines by 24 characters
- Graphical Display size is 181 x 56 pixels
- Displays caller related information with active appearances for easier call handling
- White backlit display for easier viewing in all lighting conditions

Avaya Digital Telephones

Avaya 2402 digital telephone

The Avaya 2402 is a low-cost, low function, 2-wire digital telephone. The 2402 can be used as an alias to 6402 telephone.

The Avaya 2402 telephone has the following characteristics:

- 2-line × 24 character LCD
- 2 call appearance buttons
- Handset and 12-button dialpad
- Wall mountable
- Display of downloaded extension number

- Highly visible message waiting indicator
- Message button for expedited access to voice mail
- Buttons for conference, transfer, drop, hold, and redial
- Built-in one-way speaker with group listen operation
- Speaker, feature, and mute buttons, each with LED indicators
- Feature button allows access, by way of the dial pad, to 12 Communication Manager features that do not require indicators
- Adjustable volume control for the handset, the speaker, and the ringer
- Electronically stored part ID and serial number for use with Automatic Customer Telephone Rearrangement
- 9-foot phone cord and 14-foot (4.27-meter) gray, modular line cord
- Stand included
- Native support that gives you the ability to administer and maintain the telephone without using an alias

Avaya 2410 digital telephone

The Avaya 2410 is a 2-wire digital telephone with the following characteristics:

- 5-line × 29-character monochrome LCD with 5-column x 8-row matrix of dots that supports 5-dot × 7-dot European or Katakana characters
- Handset and 12-button dialpad
- Adjustable viewing angle
- Wall mountable
- Six general purpose buttons to access up to 12 system call appearance or features
- Downloadable firmware for future upgrades
- Downloaded call appearance or feature button labels
- Four 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
- Buttons for conference, transfer, drop, hold, and redial
- Headset jack that is separate from the handset jack
- Built-in speakerphone with group listen operation
- Speaker, headset, mute buttons, each with LED indicators

- Volume up or volume down buttons for handset, headset, speakerphone, and ringer
- 48-Entry Call Log including total incoming answered, incoming unanswered, and outgoing calls
- Automatic Gain Control on all audio interfaces
- Electronically stored part ID and serial number for use with Automatic Customer Telephone Rearrangement
- Native support that gives you the ability to administer and maintain the telephone without using an alias

Avaya 2420 digital telephone

The Avaya 2420 is a 2-wire digital telephone with the following characteristics:

- 7-line × 29-character monochrome liquid crystal display (LCD) with 5-column x 8-row matrix of dots that supports 5-dot × 7-dot European or Katakana characters
- Handset and 12-button dial pad
- Adjustable viewing angle
- Wall mountable
- Eight general purpose buttons to access up to 24 system call appearances or features
- Downloaded call appearance or feature button labels
- Four local softkey feature buttons
- Exit, previous, and next buttons to navigate the display
- Highly visible message waiting indicator
- Message button for expedited access to voice mail
- Buttons for conference, transfer, drop, hold, and redial
- Headset jack that is separate from the handset jack
- Built-in speakerphone with group listen operation
- Speaker, headset, mute buttons, each with LED indicators
- Volume up or volume down buttons for handset, headset, speakerphone, and ringer
- 100-entry call log that records the total incoming answered, incoming unanswered, and outgoing calls
- Downloadable firmware 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 analog interface application module
- Native support that gives you the ability to administer and maintain the 2420 using the associated Feature Expansion Module

Avaya 6402 and 6402D digital telephones

Avaya 6402 and 6402D are single-line digital telephones. The difference between the two is that 6402D is equipped with a 2-line by 24-character display.

Avaya 6402 telephone has the following characteristics:

- Built-in speakerphone with group listen operation
- Six fixed buttons Speaker, Feature, Hold, Redial, Transfer, Conference
- The feature button allows access by way of the dial pad, to 12 Communication Manager features that do not require indicators or display messages
- Adjustable volume control for the handset, the speaker, and the ringer
- 2-wire connectivity through 2-wire digital line circuit packs
- Internal self test for the LEDs
- Option of eight ringing patterns
- Usable with or without the stand
- Desktop or wall mountable
- Matching 9-foot (2.7-meter) handset cord and a 7-foot (2.1-meter) modular line cord
- No Adjunct jack interface for external speakerphones or headset modules
- Headsets must be connected through the handset
- Available in dark gray and white

Avaya 6408D+ digital telephone

The 6408D+ is a digital telephone with eight buttons.

Avaya 6408D+ telephone has the following characteristics:

- 2-line x 24-character LCD display that shows the time and date when the telephone is idle
- Tiltable display with three viewing angles
- Eight call appearance and colored feature buttons with dual LEDs
- Built-in 2-way speakerphone with 1-way group listen operation
- Six fixed buttons Speaker, Feature, Hold, Redial, Transfer, Conference

  Note:

  Drop must be administered on a softkey.

- 12 system features can be administered on softkeys. The softkeys are associated with the display
- Four buttons to access softkey features such as, the menu button, the exit button, the previous button, and the next button
- Answers a call with the handset onhook when the headset feature is administered
- Adjustable volume control for the handset, the speaker, and the ringer
- Message Waiting Light (LED)
- 2-wire connectivity through 2-wire digital line circuit packs
- Accepts download from Communication Manager of country-specific voice and touchtone transmission parameters
- Internal self-test to determine if LEDs light
- Option of eight ringing patterns
- Line powered
- Can be used with or without a stand
- Desk or wall mountable
- Matching 9-foot (2.7-meter) handset cord and a 7-foot (2.1-meter) modular line cord
- Available in dark gray and white

Avaya 6416D+M digital telephone

Avaya 6416D+M telephone is a multiappearance digital telephone with 16 call appearances or feature buttons.

Avaya 6416D+M has a modular plug. This plug allows you to install a 100-A tip/ring module to the desktop stand on the telephone for increased set functionality. The tip/ring module provides a connection to adjuncts such as answering machines, fax machines, modems, analog speakerphones, and Telecommunications Device for the Deaf (TDD) machines.

You can connect an XM24 expansion module to any 6424D+M telephone to expand the number of usable buttons. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone. Avaya recommends an 1151C1 local power supply or an 1151C2 local power supply with battery holdover.

Avaya 6416D+M telephone has the following characteristics:

- 10 fixed feature buttons such as, speaker, mute, conference, transfer, hold, redial, menu, exit, previous, next
- 12 assignable soft key features that are associated with the display
- Built-in speakerphone with group listen operation
- Headset jack for direct connection of headset
- Adjustable volume control for the handset, the speaker, and the ringer

- 12 button, touchtone dial pad with raised bar on the number five key for the visually impaired
- Message waiting light (LED)
- Eight personalized ringing options
- K-style handset with 9-foot (2.7-meter) modular cord
- 14-foot (4.27-meter) modular line cord
- Pull-out card tray with feature references
- Wall or desk mountable
- International portability
- Downloadable transmission parameters
- Meets Class B requirements for use in residential locations
- Available in gray or white

Optional available components:

- 12-foot (3.66-meter) modular handset cord
- 25-foot (7.62-meter) modular line cord
- HIC-1 headset interface cord
- Headset modular base unit M12LUCM
- Avaya headset
- Amplifier handset
- Noisy environment handset

The approximate dimensions of the 6416D+M are:

- Width, 10.35-inches (26.35 centimeters)
- Depth (front to back), 8.5-inches (21.59 centimeters)
- Height (with deskstand and handset in place), 4.75-inches (12.07 centimeters)

Avaya 6424D+M digital telephone

Avaya 6424D+M telephone is a multiappearance digital telephone with 24 call appearances and feature buttons.

Avaya 6424D+M has a modular plug. This plug allows you to install a 100-A tip/ring module to the desktop stand on the telephone for increased set functionality. The tip/ring module provides a connection to adjuncts such as answering machines, fax machines, modems, analog speakerphones, and Telecommunications Device for the Deaf (TDD) machines.

You can connect an XM24 expansion module to any 6424D+M telephone to expand the number of usable buttons. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone. Avaya recommends an 1151C1 local power supply or an 1151C2 local power supply with battery holdover.

Avaya 6424D+M telephone has the following characteristics:

- 2-line × 24-character LCD display showing time and date when the telephone is in an idle status.
- A tiltable display with three viewing angles.
- Built-in 2-way speakerphone with a 1-way group listen operation
- Six fixed buttons Speaker, Feature, Hold, Redial, Transfer, Conference
- 12 system features that can be administered on the softkeys associated with the display
- Four buttons to access softkey features such as menu, exit, previous, and next
- A single next button that is used with both the softkeys and the directory function
- A ribbon connector under the telephone to connect optional modules that fit into the stand
- Headset jack under the telephone, next to the handset jack, for direct connection of a headset
    - You can answer a call with the handset onhook (when the headset feature is administered)
    - You can put handset into listen-only mode for monitoring while the headset button is turned on
- No adjunct jack interface for external S201/S203 speakerphone adjuncts or headset adjuncts
- User-customizable call appearance and feature buttons, with system administrator permission
- Adjustable volume control for the handset, the speaker, and the ringer
- Message waiting light
- 2-wire connectivity through 2-wire digital line circuit packs only
- Internal self test
- Option of eight ringing patterns
- Can be used with the stand or without the stand when the 100A Analog Interface module is not present
- Desktop mountable or wall mountable (if the 100A Analog Interface Module is not present)
- Meets Class B requirements for use in residential location.
- Available in dark gray and white.

Optional components:

- Supports optional XM24 expansion module that allows for an additional 24 call appearance and feature buttons with dual LEDs.

Avaya 6424D+M telephone is powered from the system to which the telephone is connected. Adjunct station or closet power is necessary only when connecting an XM24 expansion module or the 100 A Analog Interface Module. If both modules are connected to the 6424D+M, only one power supply is necessary. The 6424D+M continues to work if the auxiliary power is interrupted but the modules do not work.

Avaya Callmaster Telephone

Avaya Callmaster IV (603H) digital telephone

Avaya Callmaster IV telephone supports applications that use the Automatic Call Distribution (ACD) feature. The ergonomic design of the Avaya Callmaster IV allows agents to handle large volumes of calls more quickly and efficiently. VuStats, a display of agent and call center statistics on the Avaya Callmaster IV, provides agents with real-time information.

Avaya Callmaster IV works in a 2-wire environment. The older Avaya Callmaster IV (603F) has a separate jack for the older 4-wire environment and reduced wiring expenses and installation change adjustments.

Avaya Callmaster IV includes as standard a built-in Recorder Interface Module (RIM) that supports connections to agent recording equipment.

Avaya Callmaster IV can be used in home office environments with a DEFINITY® Extender.

Avaya Callmaster IV has the following characteristics:

- Six rubber-domed administrable call appearance or flexible feature buttons
- 15 rubber-domed administrable flexible feature buttons
- Eight fixed feature buttons such as, Conference, Transfer, Drop, Hold, Mute, Volume, Release, Login
- 80-character alphanumeric LCD display
- 12 button, touchtone dial pad with raised bar on the number five key for the visually impaired
- Message waiting light (LED)
- Recorder interface module
- Dual headset jacks
- Eight personalized ringing options
- Adjustable volume control for the handset and the ringer
- Stand for desktop use
- International portability
- Amplifier handset

Avaya Callmaster V (607A) digital telephone

Avaya Callmaster V telephone supports applications that use the Automatic Call Distribution (ACD) feature. The ergonomic design of the Avaya Callmaster V allows agents to handle large volumes of calls more quickly and efficiently. VuStats, a display of agent and call center statistics on the Avaya Callmaster V, provides agents with real-time information.

Avaya 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:

- Two built-in headset jacks
- A built-in Recorder Interface Module (RIM) with Warning Tone. The RIM supports the recording of both agent voice and caller voice on a voice-activated analog tape recorder. A soft beep warning tone is repeated every 13.5 seconds to notify the agent and the calling party that the call is being recorded. The user can deactivate the warning tone.

Avaya Callmaster V can be used in home office environments with a DEFINITY Extender.

Avaya Callmaster V has the following characteristics:

- 16 dual-LED call appearance or feature buttons
- An adjustable 48-character liquid crystal display (LCD)
- 10 fixed feature buttons such as, speaker, mute, conference, transfer, hold, redial, menu, exit, previous, next
- 12 assignable soft key features associated with the display
- One-way listen-only speaker for group listening, dialing while the handset in place, or handsfree listening
- Adjustable volume control for the handset, the speaker, and the ringer
- Works in a 2-wire environment

Avaya Callmaster VI (606A) digital telephone

The Callmaster VI telephone is a small digital voice telephone. The Callmaster VI is used with the application software that runs on a PC. The Callmaster VI is powered from the PBX and connects to the PC by way of a standard EIA or TIA-574 serial port interface.

Avaya Callmaster VI has the following characteristics:

- Two useable headset input jacks
- Optional headset with custom cable
- Message waiting indicator
- Five preset buttons such as, Headset on and off, Mute, Two call appearances, Release
- Three administrable feature buttons

- Voice announcement recording feature
  - Up to six announcements that are 9.6 seconds in length
  - Announcements can be played automatically for incoming calls

Avaya DECT Handsets

3701 IP DECT Handset

The 3701 IP DECT Telephone is part of the Avaya IP Digital Enhanced Cordless Telephony (DECT) solution, available only in the EMEA and APAC regions. The IP DECT solution provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. The system also supports users in different offices connected through a wide area network (WAN).

The 3701 telephone has the following characteristics:

- Listen-only handsfree speaker
- SOS Emergency key for speed dialling an emergency number
- Information key that can be used for:
  - Phone number lists and voice mail indication
  - Information and speaker key flash when active
- 50 phone book entries in every handset, independent of the system phone book
- 10 possible ring tones with temporary mute
- 4-level signal strength display
- Speaker and handset volume, 3-levels and mute capability
- Manual and automatic key lock (1 minute timer)
- Temporary ring tone muting
- Silent charging
- 12 menu languages: Czech, Danish, Dutch, English, Finnish, French, German, Italian, Norwegian, Portuguese, Spanish, and Swedish.
- Illuminated 3-line graphic display (96 x 33 pixels), variable 3-level contrast
- Stand-by time of up to 200 hours
- Talk time of up to 20 hours
- Charge time of maximum 6 hours for empty batteries
- Weight of 138 grammes including 3 AAA (NiMH) batteries
- Dimensions (height x width x depth) are 146 mm x 55 mm x 28 mm

- Optional accessories include a desktop charger, an adapter cord for use with headsets, and a heavy-duty belt clip

3711 IP DECT Telephone

The Avaya 3711 IP DECT Telephone is part of the Avaya IP Digital Enhanced Cordless Telephony (DECT) solution, available only in the EMEA and APAC regions. The IP DECT solution provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. The system also supports users in different offices connected through a wide area network (WAN).

The 3711 telephone supports the same features as the 3701 IP DECT handset, but with the following differences:

- Full hands-free speakerphone operation in half duplex mode. You can either talk or listen but not both at the same time
- Headset connection (2.5 mm jack)
- Vibrating alarm
- Personal phone book with 100 entries, independent of system phone book
- Voice mail indication
- Choice of 30 ring tones
- Speaker and handset volume, 7-levels and mute capability
- 10 menu languages: Danish, Dutch, English, Finnish, French, German, Italian, Portuguese, Spanish, and Swedish.
- Illuminated 5-line graphic display (96 x 60 pixels), variable 7-level contrast

3720 DECT Handset

The Avaya 3720 DECT Handset is designed for office and industrial environments. These handsets deliver high-security, high-quality voice communications over a dedicated, encrypted wireless network. These headset provide reliable access to PBX features like hold, conferencing, and messaging.

The Avaya 3720 Handset has the following characteristics:

- High security and excellent voice quality
- Software and configuration download over-the-air (IP DECT) or via intelligent network attached cradles
- Long range and battery life (up to 20 hours)
- IP DECT network compatibility
- Headset interface
- Black and white display

- Half duplex speaker phone
- Graphical user interface
- Four way navigation key
- 2.5 mm standard headset connector
- Supports multiple languages such as German, English, French, Spanish, Russian
- Talk time up to 16 hours under optimal conditions
- Standby time up to 180 hours under optimal conditions
- Charge time below 4 hours

3725 DECT Handset

The Avaya 3725 DECT Handset is designed for users across many industries. These handsets deliver high-security, high-quality voice communications over a dedicated, encrypted wireless network. The 3725 headset adds a backlit keypad, large color display, text messaging, and liquid and dust protection.

The Avaya 3725 DECT Handset has the following characteristics:

- High security and excellent voice quality
- Software and configuration download over-the-air (IP DECT) or via intelligent network attached cradles
- Long range and battery life (up to 20 hours)
- IP DECT network compatibility
- Headset interface
- Color display
- Half duplex speaker phone
- Graphical user interface
- Five way navigation key
- Bluetooth headset interface (Bluetooth 2.0, handsfree profile)
- Liquid and dust protected (IP 44)
- Easy exchange of battery pack
- Multi-functional button (alarm call, answer call, etc.)
- Text message support (requires AIWS server, 30 messages sent/received storable, message length 160 characters)
- 2.5 mm standard headset connector

- Supports Multiple languages such as Czech, Danish, Dutch, English, Finnish, French, German, Italian, Norwegian, Portuguese, Spanish, Swedish, Polish, Greek, Hungarian, Brazilian Portuguese, Slovakian, Turkish, Russian
- Standby time 120 hours under optimal conditions

Avaya IP Wireless Telephones

3641 and 3645 IP Wireless Handsets

The Avaya 3641 and 3645 IP Wireless Handsets are integrated with Avaya Aura®, Avaya Communication Manager Release 2.2 and later, and Avaya IP Office Release 4.1 and later.

The handsets leverage the Avaya Voice Priority Processor and the Open Application Interface server. These phones are compatible with 802.11a, b, and g networks.

6120 WLAN Handset

Avaya 6120 WLAN Handset is designed for premises-based mobile communication in the workplace. This intermediate-level IEEE 802.11a/b/g wireless device is optimized for use in a general office, and in financial and hospitality industries.

6140 WLAN Handset

Avaya 6140 WLAN Handset is a professional-level IEEE 802.11a/b/g wireless device with integrated push-to-talk functionality. It is designed for premises-based mobile communications in the workplace.

Avaya Attendant Console

Avaya 302D attendant console

The Avaya 302D attendant console is a 2-wire unit with an optional 26C expansion module. Avaya 302D cannot be used in a 4-wire environment.

The Avaya 302D attendant console has the following characteristics:

- Draws power from the desktop or the telephone closet
- Desktop mountable only
- 1-line × 40-character display that supports Katakana, Roman and Euro font Character set. Label languages are Japanese, English, French, Dutch, Spanish Latin America, Italian,

German, Canadian French, Brazilian Portuguese. Two labels are included with each 302D console

- Handset and headset connection is a single modular plug on the front
- Service observing must be done through Communication Manager by the Service Observing feature
- Available in dark gray, black and white

Optional available components:

- 26C Selector Console:
  - Has 20 Hundreds group buttons and 100 Tens group buttons. Each Hundreds group button is assigned the first digit or first two digits of each group of 100 extension (room) numbers. The Tens group buttons are automatically assigned a tens-and-ones digit. The buttons can be used for 3-digit or 4-digit extensions

    The following example is for dialing 4-digit extensions. If you have rooms numbered from 7000 to 7099, you can have a Hundreds button labeled "70" and a tens group button labeled "01." Press "70", then "01" to dial extension 7001 with only two button presses.
  - Has busy or idle status display for each button
- An H1C or M12L for the headset
- An optional Training-Y connector. This connector can be used in conjunction with the headset connection for desktop listen-only supervisor support.

Avaya Softconsole

Avaya Softconsole is a software attendant console solution. Avaya Softconsole is available for industry standard IP and Avaya Digital Communications Protocol (DCP). IP connectivity is available in both Voice over IP configuration (Road Warrior) and dual connection (Telecommuter) for toll-quality audio.

Avaya Softconsole has the following characteristics:

- Busy Lamp Fields (BLF), directory and display windows can all be on the same screen at the same time.
- Flexible screen arrangement for the attendant that is saved from session to session.
- Application window scales intelligently from a minimum useful size to a full screen. You can add the useful information to the display as the attendant increases the window size.
- On-request line status, such as on-hook and off-hook, is 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

- Maximum of 100 directories
- Ability to generate e-mail to users at the click of a tool bar button or a keyboard command
- Step-by-step wizard for both installation and initial administration, with help and warning text presented with each step.

MasterDirectory Data Manager

MasterDirectory Data Manager is included as part of Avaya Softconsole. MasterDirectory is a database application that is specifically designed for directory data management. With this information management tool, users can import and consolidate directory information from voice and data systems, and export the information to directory-enabled applications. MasterDirectory can import, export, and transfer data through standard-based protocols, including the following protocols:

- Open Data Base Connectivity (ODBC)
- Lightweight Directory Access Protocol (LDAP)
- File Transfer Protocol (FTP)
- Simple Mail Transfer Protocol (SMTP)
- Text delimited files (CSV)

Using these protocols, MasterDirectory can:

- Extract data from multiple sources
- Apply filters and business logic to consolidate data
- Populate directory services and databases for use by applications

For example, MasterDirectory can collect information from multiple Avaya servers, consolidate the data with human resource databases, and send the processed data to an LDAP directory service. This directory service provides data for telephone attendant applications, Internet white pages and yellow pages, and other applications.

Avaya one-X Attendant Console

Avaya one-X® Attendant is a software attendant console solution. Avaya one-X Attendant solution provides highly efficient telephony connections for attendants, receptionists and secretaries. It allows operators to quickly and easily provide communications and presence type information for any telephony connection request.

Avaya one-X Attendant is a PC based software application that integrates telephony with external caller data and workforce information. This application solution is easily expandable as business communication requirements for small, medium and large companies evolve over time.

Avaya one-X Software Attendant solution has the following characteristics:

- Caller ID connects with databases to provide intelligent customer information

- Targeted selection of alternative responders supports responsive call support
- More effective call handling enabled via customer and staff information linked directly from the switchboard
- Simplified data integration via central or external databases (Active Directory, Domino Server, LDAP or ODBC)
- Optimized personnel productivity via smart attendant activity operations
- Better workforce resource distribution can be gained via extensive call statistical reports
- Flexibility for deployment within a call center solution
- Configurable user interface supports specific user and organizational requirements
- Flexible multitasking support for attendant operations and / or PC workstation applications
- Integrated phone book (ITB) with more than 40 fields for internal and external entries
- E-mail messaging from one-X Attendant user interface to unavailable employees
- Memo field for specific information notations
- Absence information integrated from Outlook or Lotus Notes calendars or simple web-based application
- Connect callers and responders via “Drag & Drop” software actions
- Connect one-X Attendant clients from home offices or remote locations (IP Telecommuter or IP Road warrior mode)
- Configurable soft keys buttons and F1 to F12 button
- Highly expandable allowing one-X Attendant to grow with your organization
- Improved accessibility of staff to connect with constituents via connect anywhere capabilities
- Detailed status information (busy fields, absence, and more)
- Control usage data via the user profile permissions
- User passwords to protect sensitive data
- Supports languages such as English, German, French, Italian, Spanish, Dutch, Russian, Korean, Japanese, Simple Chinese and Portuguese
- Supports Microsoft Windows XP, Windows Server 2003 Standard and Enterprise, Windows Vista, Windows Server 2008 and Windows 2008 R2 Server, Windows 7 (32/64 Bit) operating systems

Avaya analog telephones

Avaya 2500 and 2554 analog telephones

The Avaya 2500 and the Avaya 2554-series telephones are basically the same, but are equipped with small different attributes. These telephone models include:

- Desk models:
  - 2500 MMGN
  - 2500 YMPG
- Wall models:
  - 2554 MMGN
  - 2554 YMPG

All Avaya 2500 and 2554 telephones are single appearance analog telephones with conventional touchtone dialing. The 2554 YMGP telephones are equipped with the following buttons:

- flash button
- message waiting light,
- redial button
- hold button
- mute button.

Features on all four telephones are accessed by the star (*) or the pound (#) key and the appropriate feature access codes.

Avaya 2500 and Avaya 2554 telephones have the following characteristics:

- The 2500 MMGN and 2554 MMGN telephones are manufactured without Positive Disconnect and without a flash button. The 2500 YMPG and 2554 YMPG have Positive Disconnect permanently enabled. When the flash button is pressed, access is provided to switch features. When the switchhook is depressed, the call is automatically disconnected, and a dial tone is provided for a completely new call. The bottom of older models has a Positive Disconnect switch with ON and OFF positions:
  - The ON position hangs up the telephone for approximately 2 seconds, even if the switchhook depression is less. This prevents inadvertent switchhook flashes. To start switchhook flash in this mode, press the flash button.
  - In the OFF position, the switchhook functions normally.
- K-type handset
- All 2500-series telephones are equipped with a 12 button, touchtone dial pad.

- All 2500-series telephones contain two jacks. The handset cord jack is on the left side of the telephone. The line cord jack is on the right rear of the set.
- All 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.
- A coiled 6-foot (1.82-meter) modular handset cord and a 7-foot (2.13-meter) modular line cord are supplied with all four of these 2500-series model telephones. A 12-foot (3.66-meter) handset cord and 14-foot (4.27-meter) and 25-foot (7.62-meter) line cords are available as options. A coiled 6-foot (1.82-meter) modular handset cord and a permanently-attached 4-inch (10.2 centimeter) modular mounting cord are supplied with 2554-series model telephones. A longer 12-foot (3.66 meter) handset cord is available as an option.
- All 2500-series telephones have an electronic tone ringer. There is a three-position ringer volume control on the bottom of the 2500 telephone and the side of the 2554 telephone.
- The 2500 YMPG, telephones can only be mounted on a desktop. They cannot be mounted on the wall. The 2554 YMPG telephones are wall-mounted telephones. They cannot be mounted on a desktop.
- All 2500-series telephones are available in black or cream.
- The tip and ring leads power all Avaya 2500-series and 2554-series telephones. The telephones do not require any external power supply.
- All Avaya 2500-series and 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. A 2500 set can be used as a power failure set in either a loop start or a ground start environment. Use in a Ground Start environment requires the optional Modular Ground Start button.
- The 2500 and 2554 telephones are FCC registered.

Avaya 6211 analog telephone

Avaya 6211 telephone is a single-line analog telephone.

Avaya 6211 telephone has the following characteristics:

- 7-foot modular line cord
- Adjustable volume control for the handset and the ringer
- Message waiting light
- Flash button
- Set hold button with an LED indicator
- Last number redial button
- 12 button, touchtone dial pad with raised bar on the number five key for the visually impaired

- Positive disconnect through switchhook
- Can be mounted on a desktop or wall-mounted
- RJ-11 data jack
- FCC approved for emergency power failure transfers
- Line powered
- Available in gray or white

Optional available components:

- 12-foot (3.66 meter) handset cord
- 14-foot (4.27-meter) line cord
- 25-foot (7.62-meter) line cords
- Avaya headsets

Avaya 6219 analog telephone

Avaya 6219 telephone is a single-line analog telephone.

Avaya 6219 telephone has the following characteristics:

- A 7-foot modular line cord
- Adjustable volume control for the handset and the ringer
- Message waiting light
- Flash button
- Set hold button with LED Indicator
- Last number redial button
- 12 button, touchtone dial pad with raised bar on the number five key for the visually impaired
- Positive disconnect through switchhook
- Desk and wall mounting available
- RJ-11 data jack
- FCC approval for emergency power failure transfers
- Line powered
- Ten memory dialing buttons
- Personalized ringing
- Available in gray or white

Optional available components:

- 2-foot handset cord
- 14-foot (4.27 meter) and 25-foot (7.62 meter) modular line cords
- Avaya headsets

Avaya 6221 analog telephone

Avaya 6221 telephone is a single-line analog telephone.

Avaya 6221 telephone has the following characteristics:

- Handset volume control
- Ringer volume control
- Message waiting light
- Flash button
- Set hold button with LED Indicator
- Mute button
- Last number redial button
- RJ-11 Data jack
- Available in gray or white
- Ten programmable dialing buttons
- Personalized ringing
- Built-in speakerphone, accessed with the SPEAKER button

AT&T TTY 8840 Analog Telephone

The TTY 8840 is an analog single-line telephone that is specifically designed for the communications needs of either the Hearing or Speech Impaired. It can make voice telephone calls or TTY calls. Features include:

- 2-line by 24 character LCD display
- Fastdial directory
- Handset Volume control
- visual Ring Flash
- Ringer
- Auto Answer
- Auto Greeting
- Tone or Pulse dialing
- TTY On/Off button to switch between TTY and Tone dialing

Can be installed behind a digital phone with a tip/ring module. This telephone also provides access to switch features when in the touch-tone mode. Access to switch features is obtained by the * or # keys, and the appropriate feature access codes.

AT&T 958 Analog Telephone Caller ID and Speakerphone

The 958 Caller ID Telephone is a single-line analog set that is desk/wall convertible and requires one tip and ring pair for operation. The 958 telephone features:

- Caller ID/Call Waiting Capability,
- 99 Name/ Number Caller ID History,
- Remove button,
- Message Waiting/New Call Light, and
- 3-line by 15 character Display that supports Call display in English/Spanish/French.

This telephone can be used on Avaya PBXs or Central Office lines. The 958 telephones are equipped with:

- Hands Free Speakerphone,
- 50 Name/Number Directory,
- data port,
- receiver/speaker volume control,
- Hold buttons,
- FLASH button,
- REDIAL button,
- ringer volume control,
- power failure operation,
- memory Loss Protection, and
- Hearing Aid compatibility.

This telephone model also provides access to switch features in the touch-tone mode. Access is gained through the * or # dial keys and the appropriate feature access codes.

Avaya EA401 and EA401A Explosive Atmosphere telephones

Underwriters Laboratories, Inc. (UL) lists these Explosive Atmosphere telephones for the following explosive atmosphere classifications and conditions:

- Class I explosive gas or vapors, group B, C, and D
- Class II combustible dusts, group E, F, and G

DANGER:

They are not to be installed in locations where acetylene gas may become present in the atmosphere.

The EA401 Explosive Atmosphere telephone provides safe and reliable communication in hazardous locations, up to and including Class I Division 1. Only standard wiring and fittings are required to connect the telephone to the system. No barrier is necessary. Since the heavy duty cast aluminum enclosure is basically soundproof, an external device to signal incoming calls, such as the EA20R Explosive Atmosphere Line Powered Telephone Ringer, is required. Additionally, the EA10 Explosive Atmosphere handset is required.

Note:

The EA401A Explosive Atmosphere telephone is an EA401 telephone that comes already assembled with an EA20R ringer and EA10 handset.

The EA401 Explosive Atmosphere telephone has the following characteristics:

- A 10-foot (3-meter) handset cord
- Standard 12-button configuration, with an additional row of buttons for Last Number Redial, Link/Flash to access PABX features and Line Release to duplicate hanging up the handset
- No handset volume control, in compliance with the FCC Waiver
- Designed for wall mounting
- Cast copper free aluminum with powder coat finish
- One-inch (2.54 cm) diameter buttons for gloves-on operation
- Magnetic Reed Hook Switch, with no moving parts, that activates when the handset is removed from or placed in the telephone cradle
- Circuit boards with a UV cured epoxy coating, which provides protection from corrosive agents such as H2S, SO2, and NH3, and environments with high humidity
- A fitting in the bottom of the enclosure for access to the fuse
- Uses an EA10 handset, which is compatible with inductively coupled hearing aid devices

Avaya IP Conference Phones

1692 IP Conference Phone

Avaya 1692 IP Conference Phone provides the convenience and productivity benefits of a powerful, handsfree conference phone. The 1692 IP Conference Phone delivers the full set of Avaya Aura® features directly to small, medium, and large conference rooms.

Avaya 1692 IP conference phone has the following characteristics:

- Full Duplex Speakerphone with 360 degree, 12-foot microphone pickup.
- RF Hardening technology resists interference from mobile phones and other wireless devices
- High resolution backlit graphical display
- 3 Context-Sensitive Soft Keys
- 5 Fixed Feature & Navigation Keys: On/Off Hook, Redial, Mute and Volume Up & Down
- 5 Menu and Navigation keys
- 12-key telephone keypad
- Single 10/100 Base T Ethernet connection
- Supports Simple Network Management Protocol (SNMP) version 2

2033 IP Conference Phone

The Avaya 2033 IP Conference Phone is intended for small and medium-sized conference rooms and managerial/executive offices. The 2033 is a full-duplex, handsfree conference phone that offers 360-degree room coverage.

Avaya Video Telephony Solution

Avaya Video Telephony Solution integrates premier video capability from Polycom into Avaya IP Telephony. The solution provides both point-to-point and multipoint capability giving users improved collaboration capability for real-time decision making.

Product details

Using a single IP network for voice and video applications, the solution allows businesses to reduce costs, simplify network management, and make video a significant component of enterprise communications. Built on open, standard-based protocols, the solution provides a full range of video telephony capabilities including:

- Desktop video: You can place a voice call with Avaya IP Softphone and add video that appears on users computer at the click of a button. Audio can be delivered to the PC or to the enterprise desk phone.
- Six-party conferencing: Enables ad hoc video/audio conferencing support for up to six parties.
- Conference room video: Enables you to quickly launch a group voice and video call using a Polycom VSX or HDX series system.
- Multipoint video: Provides voice and video conferencing at multiple locations by leveraging a Polycom MGC or RMX Multipoint Control Unit (MCU).

For more information, see Video Telephony Solution document set at www.avaya.com/support.

Avaya Wireless Solutions

W310 WLAN Gateway

The W310 WLAN Gateway uses Light Access Points (LAP) and provides a standard-based infrastructure and a new solution for wireless applications. W310 provides a rich feature set in the security, mobility, and management area and a lower overall cost of ownership for medium and large enterprises or a hotspot service provider. Instead of adding functionality to the Access Points, W310 serves as a WLAN Gateway that centralizes the Access Point features, while reducing the Access Points to simpler, cheaper devices responsible for only basic functions.

Note:

W310 WLAN Gateway supports AP600 access points (an AP-4, AP-5, or AP-6 that has been upgraded for Light AP support) if the access points have the most recent firmware.

Note:

W310 WLAN Gateway provides wireless mobility service totally independent of Communication Manager and the servers that support Communication Manager. The W310 WLAN Gateway has no interaction with Communication Manager-based systems. For wireless applications that use Communication Manager for call-handling, see W310 WLAN Gateway for Seamless Communications or Extension to Cellular and Off-PBX Station.

Figure 56: W310 WLAN Gateway

The chassis features:

- 16 10/100BaseT Ethernet ports (ports 1 through 16), connected with a Category 5 copper cable with RJ-45 termination for 100Base-T ports. Use all eight wires in the cable. The maximum copper cable length connected to a 10/100Base-T port is 100 m (328 ft)
- Two SFP GBIC copper or fiber ports
- A console port
- Fixed ports and buttons, including:
  - Port LEDs for each Ethernet port
  - 11 LEDs for additional system function

- Left and right LED select buttons

You require the following customer-supplied equipment:

- An SFP GBIC (Small Form Factor Pluggable Gigabit Interface Converter), using LC or MT-RJ fiber cables or RJ copper cables, depending on the GBIC type.
- APC (Advanced Power Conversion PLC) front end AC-DC power shelf
- One APC 800W PSU
- 2 Power cables (20 AWG or thicker) to connect the APC power shelf to W310 switches. Cables must have terminals suitable for M3.5 screws

Voice-Enabled Wireless Local Area Network Infrastructure

The Avaya infrastructure centralizes much of the WLAN intelligence in a gateway platform. This provides better integration into the enterprise network and solves the problems that plague wireless communication today.

- Management: Reduces deployment complexities and management
- Security: Increases security by maintaining a single entry point
- Superior infrastructure for Voice over IP: Supports subnet and Virtual Local Area Network (VLAN) roaming for better inbuilding mobility and higher voice quality Low-cost Avaya W110 Light Access Points (LAPs) enable dense deployments required for in-building mobility
- Investment Protection: New features can be centrally stored for easy W110 upgrades

Avaya W310 WLAN Gateway Features

- IP Multicast filtering
- Terminal and modem interface
- Wireless services
- LAN services
- Multiple Virtual Local Area Networks (VLANs) per port
- RADIUS protocol for security
- 802.1w Rapid Spanning Tree Protocol
- 802.1X Port Based Network Access Control (PBNAC)
- 802.3af-2003 Power over Ethernet
- Seamless roaming

- Policy management
- Stations Power Saving
- MAC Access Control List
- Multiple Service Set Identifiers (SSIDs)
- User group monitoring
- W110 Controller
- Wireless applications

For more information, see the following:

- *Avaya W310 WLAN Gateway Installation and Configuration Guide*, 21-300041
- *Avaya W310/W110 Quick Setup Guide Using the CLI,* 21-300178
- *Avaya W310/W110 Quick Setup Guide Using the W310 Device Manager,* 21-300179
- *Wireless AP-4, AP-5, and AP-6 User Guide*, 555-301-708, Issue 3

Specifications

Table 38 shows the site requirements of the W310 WLAN Gateway.

Table 38: W310 specifications

Description	Value
Ambient working temperature	0-40°C (32 - 104°F)
Humidity	5-95% relative humidity (not condensing)
DC input voltage	50 to 57 VDC
DC input current	8 A
DC isolation	1500 V RMS with respect to protective ground
AC input voltage	100 to 240 VAC, 50/60 Hz
AC input current	4 A
AC power dissipation	400 W maximum

A readily accessible listed safety-approved protective device with a 15A rating must be incorporated in series with building installation AC power.

W310 WLAN Gateway for Seamless Communications

The W310 WLAN Gateway supports the Seamless Communications offer on an S8300D, S8510, or S8800 Server. Seamless Communications offers converged cellular, Wireless Local Area Network (WLAN), Internet Protocol (IP), and Session Initiation Protocol (SIP) phone service. As a result, Seamless Communications enables users to use the Motorola CN620 Mobile Office Device to experience seamless wireless phone mobility between on-premises and off-premises use. The W310 WLAN Gateway, along with the Wireless Services Manager and W110 Lite Access Points (LAPs), combine with a Communication Manager server and a Global System for Mobile Communication (GSM) cellular network to provide Seamless Communications service.

Figure 57: W310 WLAN Gateway

An S8510 or S8800 Server can support a maximum of 64 W310 WLAN Gateways. An S8300D Server can support up to 50 W310 WLAN Gateways. Each W310 WLAN Gateway can, in turn, support up to 16 W110 LAPs. One W310 WLAN Gateway can support up to 1024 users. However, the actual number of Seamless Communications users that a server can support is limited to its SIP trunk capacities and licensing of SIP and CCS users.

W310 WLAN Gateway centralizes and performs many of the functions of the access points, such as seamless mobility, security policy enforcements, enforcement of QOS, and the supply of Power over Ethernet (PoE).

In addition, the W310 WLAN Gateway has the following characteristics:

- Dimensions (H x W x D): 1.75-inches (44 mm) x 19-inches (48.3 cm) x 17.7-inches (45 cm)
- Layer 2 switching
- Fits in a EIA-310-D standard 19-inch rack.
- 16 10/100 Ethernet ports with PoE (802.3af)
- 8 10/100 Ethernet ports without PoE (not currently available for use)
- Supports up to 16 non-LAP "heavy" access points, such as Avaya AP-4, AP-5, and AP-6 models once the device has been migrated to LAP functionality

  Note:

  W310 WLAN Gateway can support only 10 heavy access points at 15 Watts per port.
- One 2-Gb Ethernet port to support redundancy or stacking (not currently available for use)
- One RS-232 serial port for command-line access

- Supports 64 wireless endpoints per LAP
- Supports 320 simultaneous voice sessions
- Supports 20 simultaneous VoIP (802.11a) calls per LAP
- 100 meter maximum distance to access points
- Two LEDs per 10/100 port to indicate PoE status and link status
- One LED for power and one LED for the 2-Gb Ethernet port
- Supports RADIUS server and Active Directory authentication
- Supports firmware download to the W310 WLAN Gateway and from the W310 WLAN Gateway to the W110 LAP

The following additional devices are used with the support of W310 WLAN Gateway Seamless Communications:

- Wireless Services Manager
- W110 Lite Access Points

Wireless Services Manager for Seamless Communications

Wireless Services Manager (WSM) handles dispatch calling (communication between walkie talkies), a function allows Motorola CN620 handsets to communicate using the “push to talk” communications style while in the WLAN. The WSM also manages the CN620 handset administration and initialization sequences and serves as a SIP proxy and registrar for WLAN SIP signalling. The WSM consists of the WSM SIP Proxy/Registrar software, Dispatch software, and a V120 Sun server.

Figure 58: Wireless Services Manager (WSM)

The V210 Sun server has the following characteristics:

- 650 MHz ultraSPARC server
- 4-GB memory

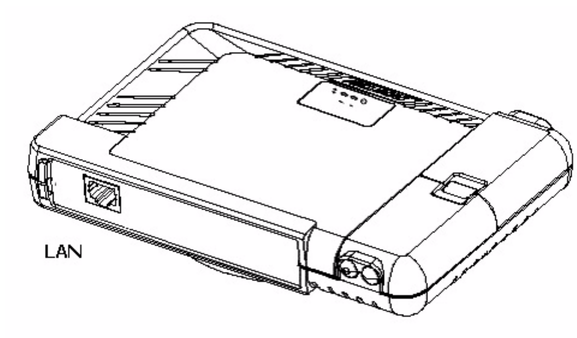
- Two 36-GB hard drives

WSM communicates with the server over SIP trunk groups. For the S8510 and S8800 Servers, the SIP trunk groups are connected over the CLAN board. For the S8300D Server, the SIP trunk groups are connected over a G700 Ethernet port.

W110 Light Access Point for Seamless Communications

The W110 Light Access Point (LAP) is an access point that operates the radio cards necessary for wireless transmission and reception. W110 supports Seamless Communications and can be used only with a W310 WLAN Gateway.

Figure 59: W110 Light Access Point



W110 LAP has the following characteristics:

- Can be mounted on a wall, ceiling, or desktop
- LEDs to indicate power status, LAN traffic, and wireless traffic
- Support 802.3af-2003 PoE
- Firmware downloadable from the W310 WLAN Gateway
- Up to 16 LAPs for one W310 WLAN Gateway
- Supports 802.11a and 802.11b/g radios

Additional documentation for Seamless Communications

For information on installing Seamless Communications, see

- *Seamless Communications Total Solution Guide*, 21-300041
- *Seamless Communications Configuration Guide*
- *Avaya W310 WLAN Gateway Installation and Configuration Guide*, 21-300041

- *Avaya W310/W110 Quick Setup Guide Using the CLI,* 21-300178
- *Avaya W310/W110 Quick Setup Guide Using the W310 Device Manager,* 21-300179
- *Wireless AP-4, AP-5, and AP-6 User Guide*, 555-301-708, Issue 3
- *Motorola NMS User Guide*
- *Motorola WSN User Guide*

Extension to Cellular and Off-PBX Station

Avaya Extension to Cellular and Off-PBX Station application types provide users with the capability to have one administered phone that supports Communication Manager features for both an office phone and one outside phone. Off-PBX Station application types allow users to receive and place office calls anywhere, any time. Application types are Extension to Cellular, Outboard Proxy SIP (OPS), Seamless Converged Communications Across Network (SCCAN), and Cellular Service Provider (CSP). Extension to Cellular extends office calls to a user's cellular phone. CSP performs the same functions as Extension to Cellular but is sold to a user by their cellular service provider. CSP differs from Extension to Cellular only in that a user of the CSP application type cannot disable the feature. OPS is used to administer a SIP phone. SCCAN offers voice and data access from a single SCCAN handset that is integrated with a desktop phone across the corporate Wireless Local Area Network (WLAN), public Global System for Mobile communication (GSM), and cellular networks. A user can have more than one application type per station.

Through all of these application types, people calling an office phone can reach users even if they are not in the office. With this added flexibility, you can access certain Communication Manager features from any phone device that is outside the office phone network.

You can administer the following types of Avaya phones as the host phone using Extension to Cellular and Off-PBX Station application types:

2402	**4606**	**4630**	**6408D**
2410	**4610**	**6402**	**6408D+**
2420	**4612**	**6402D**	**6416D+**
4601	**4620**	**6408**	**6424D+**
4602	**4624**	**6408+**	

The phone listed above support a number of wireless telephone devices including the Motorola CN620 Mobile Office Device. You can administer these phones on Communication Manager using the Administration with Hardware feature. In this way, the actual desk phone does not need to be physically connected.

Except for the purchase of cellular phones and cellular service by a public service provider, neither you nor users need any additional hardware for use of the Extension to Cellular/Off-PBX Station features. You simply administer the feature on the server running Communication Manager.

Avaya IP DECT Radio Base Station

The IP Digital Enhanced Cordless Telephony (DECT) Radio Base Stations are available with Avaya Aura® Communication Manager. Radio Base Station supports encryption of the communication between handset and base station and authentication of the handset against the base station.

IP DECT Radio Base Station for 3720 and 3725 Handsets

IP DECT radio base station for 3720 and 3725 handsets have the following characteristics:

- Handles up to eight concurrent calls
- Power over Ethernet or local power supply
- Supports Wireless networks of up to 1000 IP DECT Radio Base stations with up to 2000 DECT handsets
- Synchronization for seamless handover done over-the-air
- Master software can run on several base station and is required for Coordination of the over-the-air synchronization, LDAP phonebook access via AIWS, and VoIP interface to the PBX
- Several master software can run parallelly for redundancy, load balancing, and multi-site support
- Web Interface for configuration and software update
- Power over Ethernet or local power supply possible
- Supports two different versions of radio base station, one with internal antennas and one with external antenna
- Supports 3701/3711 IP DECT handsets only in CAP mode

IP DECT Radio Base Station for 3701 and 3711 Handsets

IP DECT radio base station for 3701 and 3711 handsets have the following characteristic:

- Supports RFP 32, RFP 34 Indoor, and Outdoor Base Station
- Uses Internal antenna (RFP32) and External antenna (RFP34)

- Wall mountable
- Supports 12 slots on the air and 8 channels
- Synchronization via air interface
- Supports generic access profile (GAP)
- Connection Handover according to GAP-standard
- DSAA Authentication of Base and Handset (DECT Standard Authentication Algorithm)
- DSC (DECT Standard Cypher) 64-bit “through-the-air” encryption
- Supports 802.3af standard-based PoE
- Optional region-specific AC and DC power supply

Appendix A: Specifications for Avaya Media Gateways

Environmental requirements

Altitude and air pressure

For altitudes above 5,000 feet (1,525 meters), you must reduce the maximum short-term temperature. Reduce this temperature limit by 1 ^{o}F (1.8 ^{o}C) for every 1,000 feet (304.8 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}C). 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 pounds per square-inch (psi) (648 to 1,048 millibars).

Cabinet dimensions and clearances

Floor plans usually allocate space around the front, ends, and rear of the cabinets for maintenance purposes. Floor area requirements vary between cabinets.

Floor load requirements

The equipment room floor must meet the commercial floor loading code of at least 50 pounds per square foot (242 kilograms per square meter). Floor plans usually allocate space around the front, the ends, and, if necessary, the rear for maintenance access of the Media Gateways. Additional floor support might be required if the floor load is greater than 50 pounds per square foot (242 kilograms per square meter). The following table contains information about weight and floor loading for the battery.

	Weight (pounds)	Floor loading (pounds per square foot)	Notes
Battery			
100-A	maximum 400 (181 kg)	180 (871.2 kg/m^2)	
200-A	maximum 815 (370 kg)	328 foot (1587.5 kg/m^2)	

	Weight (pounds)	Floor loading (pounds per square foot)	Notes
300-A	maximum 1480 (671 kg)	476 (2303.8 kg/m^2)	
400-A	maximum 1580 (717kg)	625 (3025 kg/m^2)	

Temperature and humidity

Install the DEFINITY equipment in a well-ventilated area. Maximum equipment performance is obtained at an ambient room temperature up to 110 ^{o}F (43 ^{o}C) for continuous operation and between 40 °F and 120 ^{o}F (4 ^{o}C and 49 ^{o}C) for short term operation. Short term operation is not more than 72 consecutive hours or 15 days in a year.

The relative humidity range is 10% to 95% at up to 84 ^{o}F (29 ^{o}C). Above 84 ^{o}F, the maximum relative humidity decreases from 95% to 32% at 120 ^{o}F (49 ^{o}C). Installations outside these limits might reduce system life or impede operations. The recommended temperature and humidity range is 65 ^{o}F to 85 ^{o}F (18^{o}C to 29 ^{o}C) at 20 to 60% relative humidity.

The following table correlates room temperature with allowable relative humidity.

Recommended room temperature (oF)	Recommended room temperature (oC)	Recommended 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

Recommended room temperature (°F)	Recommended room temperature (°C)	Recommended relative humidity (%)
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

System protection

The following types of system protection are provided to keep the switch active and online:

- Overvoltage
- Sneak current
- Lightning
- Earthquake

Protection from hazardous voltages

Protection from hazardous voltages and currents is required for all off-premises trunks, lines, and terminal installations. Both sneak current protection and overvoltage protection from lightning, power induction, and so on, are required.

Overvoltage protection

The following devices protect the system from overvoltage:

- Analog trunks use the 507B Sneak Protector. The local telephone company usually provides overvoltage protection.
- Analog voice and 2-wire DCP terminals can use one of the following types of combined protection against overvoltage and sneak current.

  The terminals can also use the equivalent of one of the following types:

  - 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.
- DS-1, E1, and T1 circuits require isolation from exposed facilities. A CSU (T1), lightwave integration unit (E1), or other equipment provides this isolation.

Sneak current protection

Extraneous power induces sneak current protection to protect building wiring with fuses. The fuses protect wiring between the network interface and trunk circuits. The fuses also protect the circuit packs.

All incoming trunks and outgoing trunks and off-premises station lines pass through the sneak fuses. 507B sneak fuse panels are installed on the system side of the network interface.

Sneak current protectors must be either UL-listed or CSA-certified or must comply with local safety standards. Sneak current protectors must have a maximum rating of 350 milliamperes (mA) and a minimum voltage rating of 600 volts, 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 with the wires. The mutual coupling reduces the voltage difference between the ground and the switch.

Ensure that the CBC connects to a telecommunications cable that is firmly connected to an approved ground. In multiplestory buildings, you must connect the CBC to an approved ground at each floor.

CBC can be any of the following configurations:

- a 10 AWG (5.3 millimeters 2/2.6 millimeters) ground wire
- a continuous cable sheath that surrounds wires within a cable
- six unused pairs of wire within a cable that are twisted and soldered together

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 millimeters^2/4.1 millimeters) 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, you must preserve ground integrity. Plug the power supply for this equipment into one of the two convenience outlets on the rear of the gateway. The convenience outlets are fused at 5 A. 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 might require additional bracing. Contact your Avaya representative for earthquake requirements at the location of the system installation.

Appendix B: Optional components for servers

Media gateways

Media Gateway	Supported Servers S8300D	S8510	S8800
Media gateways and integrated gateways	x	x	x
Avaya G350 Media Gateway	x	x	x
Avaya G430 Media Gateway	x	x	x
Avaya G450 Media Gateway	x	x	x
Avaya G650 Media Gateway		x	x
Avaya G700 Media Gateway	x	x	x
IG550 Integrated Gateway	x	x	x
G860 Media Gateway			x

Media modules

Media module	Supported Configurations S8300D with a Gxxx	S8510 with a Gxxx	S8800 with a Gxxx
MM312 DCP Media Module			
MM314 LAN Media Module			
MM316 LAN Media Module			
MM340 E1/T1 data WAN Media Module		x	x
MM342 USP data WAN Media Module		x	x
MM710 T1/E1 Media Module	x	x	x

Media module	Supported Configurations S8300D with a Gxxx	S8510 with a Gxxx	S8800 with a Gxxx
MM711 Analog Media Module	x	x	x
MM712 DCP Media Module	x	x	x
MM714 Analog Media Module	x	x	x
MM716 Analog Media Module	x	x	x
MM717 DCP Media Module	x	x	x
MM720 BRI Media Module	x	x	x
MM722 BRI Media Module	x	x	x
MM760 VoIP Media Module	x		x

Circuit packs

Power circuit packs

Circuit Packs	Supported Servers S8510	S8800
655A power supply	x	x
650A AC power unit		
The 120A CSU is supported on DEFINITY, Multivantage, and Communication Manager servers that support TN circuit packs.	x	x
TN2202 ring generator	x	x
TN755B neon power unit	x	x

Line circuit packs

Circuit Pack Name	Supported Servers S8510	S8800
TN479 analog line (16 ports)	x	x
TN556D ISDN-BRI 4-wire S/T-NT interface (12 ports)	x	x
TN746B analog line (16 ports)	x	x
TN754C DCP digital line (4-wire, 8 ports)	x	x
TN762B hybrid line (8 ports)	x	x
TN769 analog line (8 ports)	x	x
TN791 analog guest line (16 ports)	x	x
TN793CP analog line with Caller ID for multiple countries (24 ports)	x	x
TN797 analog trunk or line circuit pack (8 ports)	x	x
TN2181 DCP digital line (2-wire, 16 ports)	x	x
TN2183/TN2215 analog line for multiple countries (16 ports)	x	x
TN2185B ISDN-BRI S/T-TE interface (4-wire, 8 ports)	x	x
TN2198 ISDN-BRI U interface (2-wire, 12 ports)	x	x
TN2224CP DCP digital line (2-wire, 24 ports)	x	x
TN2215/TN2183 analog line for multiple countries (16 ports) (international offers or Offer B only for US and Canada)	x	x
TN2224CP DCP digital line (2-wire, 24 ports)	x	x

Trunk circuit packs

Circuit Pack Name	Supported Servers S8510	S8800
TN429D incoming call line identification (ICLID)	x	x
TN459B direct inward dialing trunk (8 ports)	x	x

Circuit Pack Name	Supported Servers	
	S8510	S8800
TN436B direct inward dialing trunk (8 ports)	x	x
TN464HP DS-1 interface, T1 (24 channels) or E1 (32 channels)	x	x
TN465C central office trunk (8 ports)	x	x
TN747B central office trunk (8 ports)	x	x
TN753B direct inward dialing trunk (8 ports)	x	x
TN760E tie trunk (4-wire, 4 ports)	x	x
TN763D auxiliary trunk (4 ports)	x	x
TN767E DS-1 interface, T1 (24 channels)	x	x
TN1654 DS-1 converter, T1 (24 channels) and E1 (32 channels)	x	x
TN2140B tie trunk (4-wire, 4 ports)	x	x
TN2146 direct inward dialing trunk (8 ports)	x	x
TN2147C central office trunk (8 ports)	x	x
TN2184 DIOD trunk (4 ports)	x	x
TN2199 central office trunk (3-wire, 4 ports)	x	x
TN2207 DS-1 interface, T1 (24 channels) and E1 (32 channels)	x	x
TN2209 tie trunk (4-wire, 4 ports)	x	x
TN2242 digital trunk	x	x
TN2308 direct inward dialing trunk (8 ports)	x	x
TN2313AP DS-1 interface (24 channels)	x	x
TN2464CP DS-1 interface with echo cancellation, T1/E1	x	x

Control circuit packs

Circuit Pack Name	Supported Servers S8510	S8800
TN744E call classifier and tone detector (8 ports)	x	x
TN771DP maintenance and test	x	x
TN775C maintenance	x	x
TN799DP control LAN (C-LAN) interface	x	x
TN2182C tone clock, tone detector, and call classifier (8 ports)	x	x
TN2302AP IP media processor	x	x
TN2312BP IP server interface	x	x
TN2464CP DS-1 interface with echo cancellation, T1/E1		
TN2464CP DS-1 interface with echo cancellation, T1/E1		
TN2602AP IP Media Resource 320	x	x

Service circuit packs

Circuit Packs	Supported Servers S8510	S8800
TN433 speech synthesizer	x	x
TN725B speech synthesizer	x	x
TN787K multimedia interface		
TN788C multimedia voice conditioner		
TNCCSC-1 PRI to DASS converter	x	x
TNCCSC-2 PRI to DPNSS converter	x	x
TNCCSC-3 PRI to DPNSS converter	x	x

Circuit Packs	Supported Servers S8510	S8800
TN-C7 PRI to SS7 converter	x	x
TN-CIN voice, fax, and data multiplexer	x	x

Application circuit packs

Circuit Packs	Supported Servers S8510	S8800
TN750C recorded announcement (16 channels)		
TN801B MAPD (LAN gateway interface)	x	x
TN2501AP voice announcements over LAN (VAL)	x	x

Wireless circuit packs

Circuit Packs	Supported Servers S8510	S8800
TN789B radio controller	x	x

Avaya telephone devices

All telephones listed in Deskphones and softphones can be used with any server that supports Communication Manager Release 3.0 and later.

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