Avaya Aura® Communication Manager Product Description

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Contents

Chapter 1: Introduction

Purpose

This document describes tested product characteristics and capabilities, including product overview and feature descriptions, and security and licensing requirements.

This document is intended for anyone who wants to gain a high-level understanding of the product features and functions.

Change history

Issue	Date	Summary of changes
1.0	August 2015	Initial release
2.0	May 2016	• Added the New in this release on page 10 section to add information about the new features. • Updated the Communication Manager OVA on page 15 section to add support for Dell™ PowerEdge™ R630 and HP ProLiant DL360 G9 common servers. • Updated the Port network and gateway connectivity on page 16 section to add support for Dell™ PowerEdge™ R630 and HP ProLiant DL360 G9 common servers. • Updated the Communication Manager intelligent networking on page 20 section to add support for Dell™ PowerEdge™ R630 and HP

Table continues…

Issue	Date	Summary of changes
		ProLiant DL360 G9 common servers. • Updated the Avaya Call Center on branch gateways on page 24 section to add support for Dell™ PowerEdge™ R630 and HP ProLiant DL360 G9 common servers. • Updated the Communication Manager deployment on page 13 to add support for VMware ESXi 6.0 version.

Warranty

Avaya provides a 90-day limited warranty on Communication Manager. To understand the terms of the limited warranty, see the sales agreement or other applicable documentation. In addition, the standard warranty of Avaya and the details regarding support for Communication Manager in the warranty period is available on the Avaya Support website at https://support.avaya.com under **Help & Policies> Policies & Legal > Maintenance and Warranty Information**. See also **Help & Policies > Policies & Legal > License Terms**.

Chapter 2: Communication Manager overview

Avaya Aura® Communication Manager is the open, highly-reliable and extensible IP Telephony foundation on which Avaya delivers intelligent communications to large and small enterprises. Communication Manager effectively scales from less than 100 users up to 36,000 users on a single system.

Communication Manager is an important component of the Avaya Aura® architecture, which consolidates several components into a holistic package that enterprises need for Unified Communications. Communication Manager software is part of all the Avaya Aura® editions. This software is available with a single-user licensing fee.

Communication Manager provides centralized call control for a distributed network of gateways and a wide range of analog, digital, and IP-based communication devices. Communication Manager comes with several built-in mobility applications, call center features, advanced conference calling, and E911 capabilities.

With support for SIP, H.323, and other industry-standard communications protocols, Communication Manager provides centralized voice mail and attendant operations to organizations and call centers, across multiple locations.

Communication Manager can be configured as an evolution server or a feature server. Communication Manager configured as an evolution server uses the full-call model to provide Communication Manager features to SIP and non-SIP endpoints. As a feature server, Communication Manager only supports SIP endpoints registered to Avaya Aura® Session Manager. Communication Manager configured as a feature server uses the IP Multimedia Subsystem (IMS) half-call model for full application sequencing.

New in this release

Multi-country tone support through Avaya Aura® Media Server (MS)

Avaya Aura® Media Server (MS), as a VoIP resource can provide tones as per user location. However, if more than one users are involved in a call from different locations, the system uses the Avaya Aura® MS native location that is configured on the SIP signaling group page.

Increased capacity for TLS user

Communication Manager Release 7.0.1 supports 18000 H.323 users when TLS mode is enabled.

Note:

TLS session should be configured to operate in TTS-TLS mode. This is accomplished by going to the **IP Network Region** form and going to the **H.323 Profile** and entering the value of **H323TLS**.

Commercialization of TLS

With Communication Manager Release 7.0.1, you can use TLS with or without enabling the FIPS mode.

MLPP support for Precedence Calling to SIP endpoints

With Communication Manager Release 7.0.1, the MLPP feature now includes dialing precedence calls to 17 SIP endpoints. However SIP endpoints cannot initiate Priority or above precedence calls. SIP endpoints can initiate Routine level calls though.

Support for Opus codecs

Communication Manager Release 7.0.1 supports Opus codec for SIP calls. Following Opus-codec sets are supported in the current release:

- OPUS-NB12K
- OPUS-NB16K
- OPUS-WB20K
- OPUS-SWB24K

Opus Codec for Inter-Gateway calls

The calls involving media gateways and media servers now supports Opus codec, an open source audio-codec. This feature is limited to inter-gateway and inter-media server calls where the media resource of each gateway or media-server is used to connect the call between the two gateways or media-servers.

Hunt Group Busy Position Button

The **Hunt Group Busy position (hntpos-bsy)** button facilitates non-ACD hunt group users to voluntarily opt-in or opt-out of hunt group calls. Currently, similar behavior can be achieved by **aux-work** button only on H.323 endpoints. Aux-work works with both ACD and non-ACD hunt. This new button is implemented for 96x1 SIP endpoints.

Streaming Music On Hold from an external source

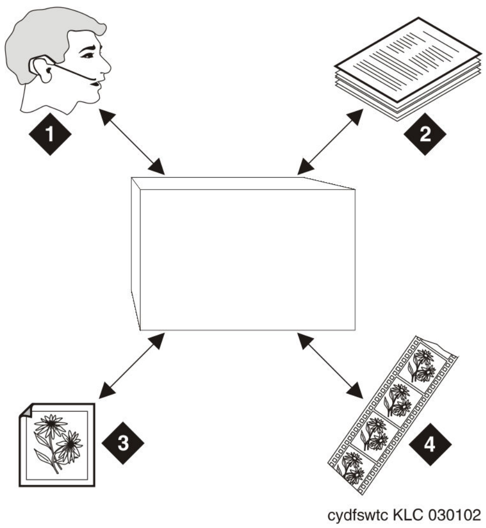
Communication Manager can be configured to source music from Avaya Aura® Media Server (MS). The Avaya Aura® Media Server (MS) is associated with external or remote servers and hosts media files using the new **Live Streaming Audio** feature. The external source can be an Internet source or a central server the announcement files are stored.

Enhanced interaction between Coverage Answer Group and Call Pickup Group

With Communication Manager Release 7.0.1, a new **Call Pickup for call to Coverage Answer Group** field is introduced. When the default value of this field is set to n and a call rings at a Coverage Answer Group member, Communication Manager does not trigger call pickup alerting if the Coverage Answer Group member is part of a Call Pickup group. This is applicable for the H.323, DCP, Analog, and SIP endpoints.

System running Communication Manager

Communication Manager provides user and system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking.



1	Voice
2	Data
3	Image
4	Video

Communication Manager software bundles

Communication Manager is available in two bundles, which satisfy most customer requirements.

Communication Manager Standard

Provides fully-converged telephony features; QSIG/DCS networking to interface with existing systems and centralized voice mail systems; and standard survivability at remote locations. Included with Avaya Aura® Standard Edition.

Communication Manager Enterprise

Includes everything in Communication Manager Standard plus multinational gateway support and high availability with 100% feature transparency at remote locations in survivable mode. Included with Avaya Aura® Enterprise Edition.

Chapter 3: Communication Manager deployment scenarios

Communication Manager deployment

Deployment

Communication Manager supports a wide range of devices, trunks, interfaces and ports. The System Manager and Communication Manager OVAs simplify deployment of Communication Manager across the organization.

Virtualization

Avaya Aura® uses standards-based virtualization technology for real-time communications. Virtualization of software allows a single piece of hardware to run multiple applications at the same time and improves portability, manageability and compatibility of applications.

Appliance Virtualization Platform is a unique, real-time virtualization technology that enables unmodified versions of Communication Manager, Voice Messaging, Session Manager, Application Enablement Services, Utility Services, and Media Services to be deployed on a single server.

Communication Manager can also be installed as an OVA on VMware vSphere 5.0, 5.1, 5.5, and 6.0. The Communication Manager VMware virtualization environment is packaged as a virtual appliance ready for deployment on VMware certified hardware.

For information about deploying Communication Manager on VMware, see *Deploying Avaya Aura® Communication Manager*.

Appliance Virtualization Platform

Appliance Virtualization Platform technology delivers simplified deployment of Unified Communications and Contact Center applications. This framework leverages virtualization technology, licensing, and support infrastructure.

The advantages of Appliance Virtualization Platform include:

- Easy installation of Avaya Aura® solution on a single server platform
- Simpler and faster deployment of applications and solutions
- Remote access and automated alarm reporting for Network Management Systems monitored by Avaya Services and Avaya Partners personnel

Appliance Virtualization Platform is a *virtual appliance* model. The model includes:

- An Avaya-defined common server platform
- An Operating System (O/S) for allocating and managing server hardware resources (CPU, memory, disk storage, and network interfaces) among virtual machine instances running on the server platform

Evolution server

An evolution server is similar to the traditional Communication Manager server. The evolution server provides Communication Manager features to both SIP and non-SIP endpoints. The evolution server uses the full-call model. The evolution server connects to Session Manager through a non-IMS Signaling group. Session Manager handles call routing for SIP endpoints and enables SIP endpoints to communicate with all other endpoints that are connected to the evolution server.

If you configure Communication Manager as an evolution server:

- H.323, digital, and analog endpoints register with Communication Manager.
- SIP endpoints register with Session Manager.
- All endpoints receive service from Communication Manager.

The evolution server supports a limited form of application sequencing.

Feature server

A feature server provides Communication Manager features to the SIP endpoints registered with Session Manager. The feature server uses the half-call model of IP Multimedia Subsystem (IMS). The feature server connects to Session Manager through an IMS-enabled SIP signaling group and an associated SIP trunk group.

The feature server supports full application sequencing.

The feature server has the following limitations:

- Does not support routing of PSTN calls directly to ISDN trunks for IMS users. You must administer the dial plan to route all PSTN calls to Session Manager over the IMS trunk group.
- Does not support traditional endpoints, such as DCP, H.323, ISDN, and analog.
- Does not support G650 Media Gateway.

Communication Manager OVA

Communication Manager is an OVA that can be deployed on Appliance Virtualization Platform. The Communication Manager OVA has all the features that Communication Manager supports, whether the OVA is on a duplicated server or a branch server.

Note:

The Communication Manager administration Web pages refer to Survivable Core as Enterprise Survivable Server (ESS) and Survivable Remote as Local Survivable Processor (LSP) respectively.

The following table provides the information about servers compatible with each OVA.

OVA type	Server configuration	Supported server
Simplex	• Main • Survivable Core • Survivable Remote	• S8300D • S8300E • Dell™ PowerEdge™ R610 • Dell™ PowerEdge™ R620 • Dell™ PowerEdge™ R630 • HP ProLiant DL360 G7 • HP ProLiant DL360 G9
Duplex	• Main • Survivable Core	• Dell™ PowerEdge™ R610 • Dell™ PowerEdge™ R620 • Dell™ PowerEdge™ R630 • HP ProLiant DL360 G7 • HP ProLiant DL360 G9

Note:

Survivable Core is not supported on S8300D and S8300E.

Communication Manager device support

Avaya Aura® Communication Manager provides for a resilient, distributed network of analog, digital and IP-based communication devices.

Communication Manager supports numerous communication devices. For example:

- Avaya IP Agent
- Avaya IP Softphone
- Avaya IP Softphone for pocket PC

- Communication Manager PC console
- Avaya one-X® Communicator
- Avaya one-X® Agent
- Avaya one-X® Portal
- Avaya SIP Softphone
- Avaya SoftConsole
- Avaya Communicator
- Scopia® client
- Scopia® environments
- Third-party video endpoints
- Polycom® VVX
- Polycom® DMA

  Note:

  Polycom® DMA replaces Polycom® CMA only for the gatekeeper functionality. The management application is provided by the Polycom® CMA gatekeeper.

For a full list of supported devices, see *Avaya Aura® Communication Manager Hardware Description and Reference*, 555-245-207.

Port network and gateway connectivity

Communication Manager supports the following connectivity features:

- Circuit switching: With the circuit switching feature, Communication Manager can connect unintelligent endpoints such as analog telephones to an intelligent network that contains IP or SIP endpoints.
- Call control signaling using H.248 branch gateway control protocol: Communication Manager uses standards based H.248 to perform call control to Avaya Branch Gateway, such as the G430. H.248 defines a framework of call control signaling between the Avaya S8300D, S8300E, Dell R610, Dell R620, Dell R630,HP DL360 G7, and HP DL360 G9 servers and multiple branch gateways.
- Separation of Bearer and Signaling: The Separation of Bearer and Signaling (SBS) feature provides a low cost virtual private network with high voice quality for customers who cannot afford private leased lines. SBS utilizes QSIG and replaces DCS + VPN for customers who need Dial Plan Expansion (DPE) functionality. SBS also utilizes QSIG to communicate between multiple Communication Manager systems.

Trunk connectivity

Communication Manager supports the following trunk connectivity features:

- Circuit switched DS1 trunk service — DS1 can be used for voice or voice-grade data, for data-transmission protocols, and for T1 and E1 services. For a full list of supported devices, see *Avaya Aura® Communication Manager Screen Reference*.
- Separate licensing for TDM stations and TDM trunks.
- Internet Protocol.
  - H.323 trunk. A TN802B in MedPro mode or a TN2302AP IP interface enables H.323 trunk service using IP connectivity between two systems running Communication Manager. The H.323 trunk groups can be configured as system-specific tie trunks, generic tie trunks, or direct-inward-dial (DID) public trunks. In addition, the H.323 trunks support ISDN features such as QSIG and BSR.
  - IP loss groups. A primary reason to accomplish a loss plan for voice communication systems is the desire to have the received speech and tone loudness at a comfortable listening level. This should be accomplished so that users can listen to each other without being concerned who or where the remote party is, or what kind of telephone equipment each may be using.
  - IP trunks. IP trunk groups may be defined as virtual private network tie lines between systems or ITS-E servers running Communication Manager. The benefits of IP trunk include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full function network with data and voice convergence and optimizing networks by using the available network resources.
  - IP trunk fallback to PSTN. The PSTN fallback of IP trunks feature refers to bypassing, or skipping over, IP trunks when IP network conditions make the voice quality of IP trunks unacceptable.
  - IP trunk link bounce. H.323 trunk link bounce provides customers with fewer call failures in the event of an IP network failure or disruption. This feature lessens the impact of IP network failures and disruptions by postponing corrective action after an H.323 signaling link failure.
  - Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network. For more information on SIP, see the documents from the Avaya Support website at http://support.avaya.com.
  - SIP trunking functionality:
    - Provides access to less expensive local and long distance telephone services, plus other hosted services from SIP service providers
    - Provides presence and availability information to members of the enterprise and authorized consumers outside the enterprise, including other enterprises and service providers
    - Facilitates SIP-enabled converged communications applications within the enterprise, such as the Seamless Service Experience.

- Auxiliary trunks connect devices in auxiliary cabinets with Communication Manager. Some of the features that are supported with this type of trunk are recorded announcements, telephone dictation service, malicious call trace, and loudspeaker paging.
- Central Office (CO) trunks connect Communication Manager to the local central office for incoming and outgoing calls.
- The digital multiplexed interface feature supports two signaling techniques: bit-oriented signaling and message-oriented signaling for direct connection to host computers.
- Direct Inward Dialing. Direct Inward Dialing (DID) trunks connect Communication Manager to the local central office for incoming calls dialed directly to stations without attendant assistance.
- Direct Inward/Outward Dialing. Traditionally, Central Office (CO) trunks and Direct Inward Dialing (DID) trunks interface an attendant console with a central office. A CO trunk services outgoing calls and accepts incoming calls that are terminated at the attendant. A Direct Inward/Outward Dialing (DIOD) trunk is used for calls that need to be terminated without an attendant interaction.
- E&M signaling - E&M trunks are used to provide analog communications links. Continuous and pulsed Continuous and pulsed E&M signaling is a modification to the E&M signaling used in the United States. Continuous E&M signaling is intended for use in Brazil, but can also be used in Hungary. Pulsed E&M signaling is intended for use in Brazil.
- E911 CAMA trunk group. This provides Caller Emergency Service Identification (CESID) information to the local enhanced 911 system through the local central office.
- Foreign Exchange. Foreign Exchange (FX) trunks connect Communication Manager to a Central Office other than to the local office.
- ISDN trunks. These give you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. Systems running Communication Manager can be connected to an ISDN using standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).
- Personal Central Office Line provides a dedicated trunk circuit between multi-appearance telephones and a CO or other switch via the network.
- Release Link Trunks (RLT) are used between switch locations to provide centralized attendant service or automatic call distribution group availability.
- Remote Access provides users with access to the system and system features from the public network. With Remote Access, users can make business calls from home or use Recorded Telephone Dictation Access to dictate a letter. An authorized user can also access system features from any onsite extension.
- Tie trunks carry communications between Communication Manager and other switches in a private network. Several types of trunks can be used, depending on the type of private network you establish.
- Timed automatic disconnect for outgoing trunk calls provides the capability to automatically disconnect an outgoing trunk call after an administrable amount of time. The amount of time that can elapse before the trunk is dropped can be specified, and can vary between 2 and 999 minutes.

- Wide Area Telecommunications Service (WATS) trunks allow you to place long-distance outgoing voice-grade calls to telephones in defined service areas. The calls are priced according to distance in the service area, length of the call, time of day, and the day of the week.
- Administrable Test Type 100 Timer. To test voice quality on a trunk set up at Central Office, you can administer the time length for which the test call must be active. Test Type 100 tests far-end to near-end loss and C-message. After the Test Type 100 line answers a call, Communication Manager sends a 1004 Hz tone at 0 dBm for 5.5 seconds, and then transitions to the silent mode until the call is disconnected.

Communication Manager public networking and connectivity

Communication Manager supports a wide range of public networking features, such as caller ID.

Public networking and connectivity features:

- Caller ID on analog trunks allows the system to accept calling name information from a Local Exchange Carrier (LEC) network that supports the Bellcore calling name specification.
- Caller ID on digital trunks. In the United States, the telephone of a user displays calling party information (if the telephone is a display telephone). Name and calling number are available from the US central offices.
- Flexible billing. The flexible billing feature allows Communication Manager or an adjunct to communicate with the public network using ISDN PRI messages to change the billing rate for an incoming 900-type call. Rate-change requests to specify a new billing rate can be made anytime after a call is answered and before it disconnects. Flexible billing is available in the U.S. for use with AT&T MultiQuest 900 Vari-A-Bill service. Flexible billing requires an adjunct switch application interface and other application software.
- Local exchange trunks. Local exchange trunks connect Communication Manager to a central office.
    - 800-service trunks let your business pay the charges for inbound long-distance calls so that callers can reach you toll-free.
    - Central Office (CO) trunks.
    - Circuit Switched DS1 Trunk Service
    - Direct Inward Dialing.
    - Direct Inward/Outward Dialing.
    - Wide Area Telecommunications Service.
- QSIG Supplementary Service - Advice of Charge. The QSIG Supplementary Service - Advice of Charge (SS-AOC) provides the capability to extend the public network charging information, provided by service providers in various countries, into users in a private network.

Related links

Trunk connectivity on page 17

Communication Manager intelligent networking

Intelligent networking and call routing lets organizations create a virtual fabric of many switches that can pass information and calls, opening new revenue opportunities and higher levels of customer service. Call routing features are also designed to reduce networking costs through effective use of IP Trunking over WAN or LAN links.

Communication Manager Intelligent networking features include:

- Avaya VoIP monitoring manager (VMON). This feature provides the ability to monitor voice over IP (VoIP) network quality. This web-based application receives QoS statistics from Avaya IP end points and displays the data via graphs and reports, so administrators can isolate voice quality problems and send traps when poor voice quality is detected.
- The Distributed Communications System (DCS) protocol allows you to configure two or more switches as if they were a single, large system. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the Communication Manager features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)
- In an Electronic Tandem Network (ETN) - also known as Private Network Access (PNA) - Communication Manager provides a variety of features on a network-wide basis. It allows calls to other systems in a private network. These calls do not use the public network. Instead, they are routed over your dedicated facilities.
- Extension number portability. When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even electronic tandem network and direct inward dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.
- Internet Protocol (IP): The capabilities and applications of Communication Manager are extended using IP. Communication Manager IP supports audio, voice, Fax, V.150.1 modem-relay packets over a LAN or WAN, and it ensures that remote workers have access to communication system features from their PCs. Communication Manager also provides standards-based control between Avaya S8300D, S8300E, Dell R610, Dell R620, Dell R630,HP DL360 G7, and HP DL360 G9 server and branch gateways allowing communications infrastructure to be distributed to the edge of the network.
    - Fax over IP: With the Fax over IP feature, enterprise networks interoperate with PSTN networks to transfer fax messages over IP. Only G430 and G450 gateways support the Fax over IP feature. If a media gateway uses the T.38 protocol to relay T.30 signaling between a fax machine and a far-end fax receiver, and the far-end fax receiver does not support T.38, then the call falls back to G.711. You can administer the feature on the Ip-codec-set form. This feature is only supported on G430 and G450 Media Gateways and only over Verizon SIP trunks. Communication Manager supports the transition of an existing audio call to a fax call. receives a request for establishing a fax call during an audio call. If T.38 is

administered, Communication Manager accepts the fax call and disconnects the audio call. Else, Communication Manager rejects the fax call. Communication Manager does not support a fax call and an audio call simultaneously

- V.150.1 Modem-over-IP: Modem devices use the V.150.1 protocol to transmit V-series modem signals between modems and telephony devices. The V.150.1 protocol is a standard recommended by International Telecommunication Union (ITU) to use a modem over IP networks that support dialup modem calls. The V.150.1 protocol defines how to transmit modem traffic between modems and telephony devices over an IP network. With the Modem-over-IP feature, secure terminals establish a secured connection over SIP and H.323 trunks and the Avaya proprietary Inter-gateway Connections (IGCs).

• QSIG support: QSIG is a global signaling and control standard for use in private corporate ISDN networks.

  - QSIG Supplementary Service - Advice of Charge. The QSIG Supplementary Service - Advice of Charge (SS-AOC) provides the capability to extend the public network charging information, provided by service providers in various countries, into users in a private network.
  - QSIG support for Unicode. The QSIG support for Unicode feature extends the Unicode support on a single server to multi-node Communication Manager networks. This feature allows Unicode support across large campus configurations.

• Uniform Dial Plan: A unique three to 13–digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the electronic tandem network to access that station. Communication Manager enhances the standard UDP with the unrestricted 13-digit Uniform Dial Plan, which allows up to five digits to be parsed for call routing.

  UDP provides extension-to-extension dialing between two or more private-switching systems.

• SIP and H.323 dual registration: With the SIP and H.323 dual registration feature, you can assign the same extension to H.323 and SIP endpoints. When you use the same extension to register a SIP endpoint to Session Manager and an H.323 endpoint to Communication Manager, an incoming call to that extension rings at both the endpoints. The user can answer the call either at the H.323 endpoint or at the SIP endpoint. You can create an extension of H.323 type by using System Manager. You can reassign the same extension as SIP by using the Stations with off-pbx telephone integration screen in Communication Manager SAT. This feature is supported by the following audio and video endpoints:

  - 96xx and 96x1 H.323 and SIP
  - 1XC H.323 and SIP
  - ADVD
  - Flare on iPad SIP Release 1.2
  - Flare on Windows SIP Release 1.2

• SIP Direct Media: Using the SIP Direct Media feature, SIP endpoints establish a direct communication path for subsequent calls, Extension to Cellular (EC500) calls, 3PCC calls, forked video calls, and forked calls to multiple devices (DAM). The direct communication path is established before the call connects between the endpoints. Communication Manager uses the TDM resources or loops the media back to the Communication Manager server only if required.

- SIP Dual Mode: With the SIP Dual Mode feature, the dual mode device can use the EC500 feature as well as the Wi-Fi and cellular networks to receive calls. The dual mode device is a combination of SIP WiFi and EC500 cellular wireless phone. This feature ensures that users receive an alert either by a VoIP call or a cellular call, but never both.

  In Release 6.3.6, this feature extends to Client Enablement Services (CES) users as well and is also known as EC500 Call Suppression.

Communication Manager data interfaces

Communication Manager data interface features include:

- Administered connections. This feature automatically establishes an end-to-end connection between two access or data endpoints based on administered attributes. It provides capabilities such as:
  - Alarm notification, including an administrable alarm type and threshold
  - Automatic restoration of connections established over a Software-Defined Data Network
  - ISDN-PRI trunk group [service may be referred to as ISDN-PRI (AC/AE) Service]
  - Scheduled as well as continuous connections; and administrable-retry interval for failed connection attempts
- Data call setup enables the setting up of data calls using a variety of methods, such as: keyboard dialing, telephone dialing, Hayes command dialing, permanent switched connections, administered connections, automatic calling unit interface, and Hot Line dialing. Data Call Setup is provided for both DCP and ISDN-BRI telephones.
- Data hot line provides for automatic placement of a data call when the originator hangs up. Data Hot Line may be used for security purposes. This feature offers fast and accurate call placement to commonly called data endpoints.
- Data Privacy protects analog data calls from being disturbed by any overriding or ringing features of the system. Data Privacy is activated when you dial an activation code at the beginning of the call.
- Data restriction protects analog data calls from being disturbed by any overriding or ringing features of the system. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.
- Default dialing. This feature provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. This feature enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a pre-administered destination in several different ways, depending on the type of data module.

- IP asynchronous links enable Communication Manager to transfer existing asynchronous adjunct connectivity to an Ethernet (TCP/IP) environment. IP asynchronous links support switch server applications, as well as client applications.
- The Multimedia Application Server Interface provides a link between Communication Manager and one or more Multimedia Communications eXchange nodes. A Multimedia Communications eXchange is a stand-alone multimedia call processor produced by Avaya.
- Multimedia calling. Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data.
- Pass advice of charge information to world-class BRI endpoints provides Advice of Charge (AOC) information to World Class BRI (WCBRI) endpoints. On a call using a WCBRI endpoint, AOC information will be displayed on the endpoint after the call has completed and the far end has hung up.

Chapter 4: Communication Manager functionality

Call Center

The Avaya Aura® Call Center provides a fully integrated telecommunications platform that supports a powerful assortment of features, capabilities, and applications designed to meet all of your customers' Call Center needs.

Call Center applications, such as Avaya Call Management System for real-time reporting and performance statistics, and Avaya Business Advocate for expert predictive routing based on incoming calls rather than historical data, are easily integrated.

Communication Manager supports the Agent ID feature using which telephones can retrieve specific agent greetings and play the greetings when calls are received.

Communication Manager also supports the Restrict Call Joining feature on Avaya Aura® Contact Center. If enabled, Communication Manager restricts the agents from initiating a transfer or a conference operation. The restriction is applicable only to outbound calls. With the Restrict Second Agent Consult feature, agents can use only one consult operation, transfer or conference, at a time.

For a complete description of Call Center features for Communication Manager, see the following documents:

- *Avaya Aura® Call Center Overview*
- *Planning an Avaya Aura® Call Center Implementation*
- *Administering Avaya Aura® Call Center Features*
- *Avaya Aura® Call Center Feature Reference*
- *Programming Call Vectoring Features in Avaya Aura® Call Center*

Avaya Call Center on branch gateways

Avaya Call Center functionality is supported on branch gateways with Communication Manager evolution server configuration, with an S8300D Server, S8300E Server, Dell™ PowerEdge™ R610, Dell™ PowerEdge™ R620, Dell™ PowerEdge™ R630, HP ProLiant DL360 G7, or HP ProLiant DL360 G9 Server, and the G650 port network gateway with the Dell™ PowerEdge™ R610, Dell™ PowerEdge™ R620, Dell™ PowerEdge™ R630, HP ProLiant DL360 G7, or HP ProLiant DL360 G9 Server.

Avaya Call Center Basic software is included with Communication Manager capability along with optional Computer Telephony Integration (CTI). This package provides a low-cost call center solution for small or branch offices.

More robust call center capabilities are provided with the optional Avaya Call Center Elite, which features Avaya Expert Agent Selection and services as the foundational software for the optional Avaya Business Advocate and Avaya Dynamic Advocate software.

The call center capabilities found in the Elite Call Center software package allows Communication Manager Call Center customers to enhance their customer service, help desk, travel, and other operations by providing powerful, integrated call routing through call vectoring and resources selection.

Computer Telephony Integration

Computer Telephony Integration (CTI) enables Communication Manager features to be controlled by external applications, and allows integration of customer databases of information with call control features.

Avaya Computer Telephony is server software that integrates the premium call control features of Communication Manager with customer information in customers' databases. It is a local area network (LAN)-based CTI solution consisting of server software that runs in a client/server configuration. Avaya Computer Telephony delivers the CTI architecture and platform that supports contact center application requirements, along with emerging applications programming interfaces (APIs). For more information, see *Avaya Aura® Application Enablement Services Overview*.

Communication Manager Automatic Call Distribution

Automatic Call Distribution (ACD) is the basic building block for call center applications. ACD offers you a method for distributing incoming calls efficiently and equitably among available agents. With ACD, incoming calls can be directed to the first idle or most idle agent within a group of agents. ACD along with Call Center Elite provides a very feature rich complement of routing and call handling capabilities. For detail information, see the *Avaya Aura® Call Center Overview* and *Avaya Aura®Call Center Feature Reference* guides.

Avaya Basic Call Management System

The Avaya Basic Call Management System (BCMS) helps you fine tune your call center operation by providing reports with the data necessary to measure your call center agents performance integrated with Communication Manager software.

The BCMS feature offers call management control and reporting at a low cost for call centers of up to 3000 agents. BCMS collects and processes ACD call data (up to seven days) within the system; an adjunct processor is not required to produce call management reports.

Communication Manager can generate real-time and historical reports.

Avaya Business Advocate

Avaya Business Advocate is the collection of features that provide flexibility in the way a call is selected for an agent in a call surplus situation, and in the way an agent is selected for a call in agent surplus situations. Instead of the traditional “first in, first out” approach, the needs of the caller, potential business value, and the desire to wait are calculated. The system then decides what agents should be matched to the callers.

The Avaya Business Advocate features include:

- Auto reserve agents. Auto reserve agents allows the system to use the percent allocation distribution feature for agent skills.
- Call selection override per skill. Call selection override is determined by skill. Call center supervisors can override the normal call handling activity either on particular skills only, or for the entire call center.
- Dynamic percentage adjustment. The dynamic percentage adjustment feature allows the system to compare actual levels of service with service targets. The system can then adjust the service target so that the overall use of the skill is more efficient.
- Dynamic queue position. Dynamic queue position allows the system to put calls from multiple vector directory numbers (VDNs) into a skill queue. This feature ensures balanced call handling across VDNs.
- Dynamic threshold adjustment. Dynamic threshold adjustment allows the system to compare actual levels of service with service targets, and to adjust overload thresholds. This feature makes the use of overload agents more efficient.
- Logged-in advocate agent counting. The logged-in advocate agent counting feature counts agents toward the advocate agent limit if a service objective, percent allocation, or a reserved skill is assigned to the agent login ID, or if one of the agent skills is assigned least occupied agent or service level supervisor.
- Percent allocation distribution. Percent allocation distribution allows the system to distribute calls to auto reserve agents by comparing a reserve agent work time in a skill with the target allocation for that skill.
- Reserve agent time in queue activation. This feature activates a reserve agent either if the expected wait time (EWT) exceeds a pre-determined threshold, or if the call time in the queue exceeds the administered service level supervisor threshold.

Communication Manager mobility

Communication Manager supports extensive mobility features — Extensive in-building or in/out building wireless choices and hot desking features like Extension to Cellular (EC500), Personal Station Access (PSA) and Automatic Customer Telephone Rearrangement (ACTR) extend Communication Manager features to users, no matter where they are working.

Communication Manager mobility features include:

- Administration Without Hardware allows you to administer telephones that are not yet physically present on the system. This greatly facilitates the speed of setting up and making changes to the telephones on the system.
- Automatic Customer Telephone Rearrangement (ACTR) allows a telephone to be unplugged from one location and moved to a different location without additional switch administration. The switch automatically associates the extension to the new port.
- Avaya Wireless Telephone Solutions (AWTS) is fully integrated with Communication Manager, and allows a user full access to Communication Manager features from a mobile telephone.

  Note:

  Avaya Wireless Telephone Solutions (AWTS) replaces the DEFINITY Wireless Business System (DWBS).
- The Avaya Extension to Cellular (EC500) feature provides the expansion of mobile services, including one-number availability, increased user capacities, flexibility across facilities and hardware, more control over unauthorized usage, enhanced enable/disable capability, increased serviceability, and support of IP trunk facilities. To define call treatment options for EC500 calls, you can use up to 99 configuration sets that are defined in the system. If you set the **Cellular voice mail detection** field, an EC500 call does not cover to the cellular voice mail. When the call server detects that the call is covered to the cellular voice mail, the call server returns the call to the server.

  Note:

  In the One-X mobile environment, you can edit the values of only the **Cellular voice mail detection** field and the **Call log notify** field. All other fields are read-only.

  Communication Manager 6.3.2 introduces additional security for the EC500/One-X Mobile Lite call (AEFSC) feature. With this feature, when a user makes an FNE call from a cellular phone, the system authenticates the call with the station security code (SSC). The call fails without the valid SSC. When a caller wants to make an EC500 call, the caller must dial the SSC after the FNE number. For example, <FNE> [Dial tone] <SSC> # [Dial tone or confirmation tone] <Subsequent digit or extension>.

  The integration of Microsoft Office Communicator (MOC) with Communication Manager through ASAI supports bridging, that is, having two user functions simultaneously. For example, the user can be on an active call on a desk phone and, at the same time, be on an active call on an off-PBX destination, such as a mobile phone.
- E911 ELIN for IP wired extensions automates the process of assigning an emergency location information number (ELIN) through an IP subnetwork (“subnet”) during a 911 call. The ELIN is

then sent over either CAMA or ISDN PRI trunks to the emergency services network when 911 is dialed.

- The Personal Station Access (PSA) feature allows you to transfer your telephone station preferences and permissions to any other compatible telephone. PSA has several telecommuting applications. For example, several telecommuting employees can share the same office on different days of the week. The employees can easily and remotely make the shared telephone “theirs” for the day.
- The SIP Visiting User (SIP VU) feature enables users with the 9620 or 9630 SIP telephone to log in to any SIP telephone in the enterprise and receive their own individualized services, including menus, contacts, and buddy lists.

  The SIP Visiting User feature relies on specialized firmware on the telephone, and also requires SIP VU administration.
- Use the Terminal Translation Initialization (TTI) feature to merge an X-ported extension to a valid port, or to separate an extension from a port. You usually use TTI to move telephones. However, you can also use TTI to connect and move attendants and data modules. Terminal Translation Initialization (TTI) also works with Administration Without Hardware (AWOH).
- The TransTalk 9000 is a single-zone or dual-zone, in-building wireless system that provides a mobility solution on Communication Manager-based systems. It delivers the benefits and accessibility of a wireless telephone with all the power and functionality of a wired desk telephone.
- X-station mobility allows remote users to access switch features. That is, X-station mobility allows certain OEM wireless telephones remoted over a PRI trunk interface to be controlled by Communication Manager as if the telephones were directly connected to the switch.
- With the Multiple Device Access (MDA) feature, a SIP user can register more than one SIP device with a single extension. For example, a user has ADVD at his desk, 96X1 in his lab, and one-X® Communicator on his laptop and all the devices are registered with the same extension 123456. When a call arrives at extension 123456, all the devices are alerted. The user can answer the call from any one of the devices. If required, the user can bridge on to the call from one of the idle devices by using the Simulated Bridge Appearance (SBA) feature. Therefore, the call can be handed off between devices without parking the call.

Collaboration

Communication Manager contains a variety of features aimed at providing easy ways to collaborate with groups of peers, customers, and partners such as executives, sales people, and professional specialists. These key work groups require a high level of effective interaction, and Communication Manager delivers.

Conferencing:

- Abort conference. When you press the conference button and for any reason you hang up before you complete the conference, you will cancel the conference. The original call that was put on soft-hold is put on hard-hold

- Conference - three party. The conference button allows single-line telephone users to make up to three-party conference calls without attendant assistance.
- Conference - six party. The conference button allows multi-appearance telephone users to make up to six-party conference calls without attendant assistance.
- Conference/transfer display prompts are based on the display prompts are based on the user class of restriction (COR), independent of the select line appearance conferencing and no-dial-tone conferencing feature.
- The conference/transfer toggle/swap feature allows users to toggle between two parties in the middle of setting up a conference call prior to connecting all parties together, or to consult with both parties prior to transferring a call.
- The group listen feature simultaneously activates your speakerphone in listen-only mode, and your handset or headset in listen-and-speak mode. This allows you to serve as spokesperson for a group. You can participate in a conversation while everyone else in the room is listening to what is said.

  Note:

  This feature is not supported on IP telephones.
- Hold/unhold conference allows a user to use the Hold button to bring the held party back to the conversation.

  Note:

  This feature is not available for BRI stations or attendant consoles.
- The Meet-me Conferencing feature allows a person to set up a dial-in conference of up to six parties. The Meet-me Conferencing feature uses call vectoring to process the setup of the conference call.
- Expanded Meet-me Conferencing. Use the Expanded Meet-me Conferencing application to set up multi-party conferences consisting of more than six parties. The Expanded Meet-me Conferencing application supports up to 300 parties.
- No dial tone conferencing. This feature can eliminate user confusion over receiving dial tone when trying to conference two existing calls.
- No hold conference. This feature allows a user to automatically add another party to a conference call while continuing the conversation of the existing call.
- Select line appearance conferencing. If you are in a conversation on line “b”, and another line is on hold or an incoming call is alerting on line “a”, then pressing the CONF button bridges the calls together. Using the select line appearance feature on Communication Manager, the user has the option of pressing a line appearance button to complete a conference instead of pressing CONF a second time.
- The selective conference party display feature allows any user on a digital station with display or on an attendant console to use the display to identify all of the other parties on a two-party or conference call.

- Selective party drop allows a user to selectively drop the party currently shown on the display with a single button push. This can be useful during conference calls when adding a party that does not answer and the call goes to voice mail.
- Selective conference mute allows a conference call participant, who has a display station, to mute a noisy trunk line. Selective conference mute is also known as far end mute.
- Enhanced SIP Signaling. Using the Enhanced SIP Signaling feature, you can:
  - see a roster of conference participants and drop the selected participants for Communication Manager-based conferences.
  - enable audio conferences, facilitated by Avaya Aura® Conferencing Release 7.0.
  - enhance the behavior of sequenced applications in a Communication Manager Feature Server environment.

Multimedia calling:

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data.

- Multimedia Application Server Interface. The multimedia Application Server Interface (ASA) provides a link between Communication Manager and one or more multimedia communications eXchange nodes. A Multimedia Communications Exchange (MMCX) is a stand-alone multimedia call processor produced by Avaya.
- Multimedia call early answer on vectors and stations. Early answer is a feature applied to multimedia calls in conjunction with conversion to voice.
- Multimedia Call Handling (MMCH) enables you to control voice, video, and data transmissions using your telephone set. The feature buttons on a multi-function telephone enable you to conduct video conferences, and forward, cover, hold, or park multimedia calls much as you would a standard voice call.
- Multimedia call redirection to multimedia endpoint. A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.
- Multimedia data conferencing (T.120) through an ESM. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). For more information on ESM, see *Installation for Adjuncts and Peripherals for Avaya Aura™ Communication Manager.*
- Multimedia hold, conference, transfer, and drop. Station users can activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.
- Multimedia queuing with voice announcement. When multimedia callers queue for an available member of a hunt group, they are able to hear an audio announcement.

Paging and intercom:

- Code calling access allows attendants, users, and tie trunk users to page with coded chime signals.

- Group paging allows a user to make an announcement to a group of people using speakerphones. The speakerphones are automatically turned on when the user begins the announcement.
- Intercom - automatic. With this feature, users who frequently call each other can do so by pressing one button instead of dialing an extension number.
- Intercom - dial. This feature allows multi-appearance telephone users to easily call others within an administered group. The calling user lifts the handset, presses the dial intercom button, and dials the one-digit or two-digit code assigned to the desired party.
- Loudspeaker paging access provides attendants and telephone users dial access to voice paging equipment. As many as nine paging zones can be provided by the system, and one zone can be provided that activates all zones at the same time.
- Manual signaling allows one user to signal another user. The receiving user hears a two-second ring. The signal is sent each time the button is pressed by the signaling user. The meaning of the signal is prearranged between the sender and the receiver. Manual signaling is denied if the receiving telephone is already ringing from an incoming call.
- Whisper page allows an assistant or colleague to bridge onto your telephone conversation and give you a message without being heard by the other party or parties you are talking to. Whisper page works only on certain types of telephones.

Team button:

The Team button feature is used to monitor members of a team of stations. Monitoring station is notified about the general redirection state of the monitored station. Starting Release 6.3.6 of Communication Manager, direct transfer, transfer upon hang-up, and override SAC/CFWD/EC features can be used with the **Team** button feature.

Communication Manager call routing

Call routing features are designed to reduce networking costs through effective use of IP Trunking over WAN or LAN links.

Call Routing features include:

- Automatic routing: Communication Manager provides a variety of automatic routing features for public and private networks. Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are the foundation for these automatic-routing features. They route calls based on the preferred (normally the least expensive) route available at the time the call is placed.
- Enbloc Dialing and Call Type Digit Analysis: With this feature, users can automatically place outgoing calls based on the telephone number information in the telephone’s call log, without the user having to modify the telephone number.
- Generalized route selection: This feature provides voice and data call-routing capabilities. You use it to select not only the least-cost routing, but also optimal routing over the appropriate facilities. It enhances AAR and ARS by providing additional parameters in the routing decision and maximizing the chance of using the right facility to route the call.

- Multiple Location Support: This feature enables local user time, local ARS Public Analysis Tables for local trunking, automatic Daylight Savings Time, enhances shared resource algorithms (touch tone receivers), and other features, when Remote Expansion Port Networks (EPNs), ATM Port Networks, and Avaya Media Gateways are remoted off of a central server at a different location.
- Alternate facility restriction levels: These levels allow Communication Manager to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, Alternate Facility Restriction Levels are also assigned.
- Traveling Class Marks: A mechanism for passing the facility restriction level of a caller from one Electronic Tandem Network switch to another. Traveling Class Marks allow privilege checking to be passed across switches through the Electronic Tandem Network.
- Answer detection: For purposes of Call-Detail Recording (CDR), it is important to know when the called party answers a call. Communication Manager provides three ways to determine whether the called party has answered an outgoing call — answer supervision by time-out, call-classifier board and network answer supervision.
- Source-based routing: With the source-based routing feature, Communication Manager sends the location information of H.323, DCP, and analog stations to Session Manager. Session Manager uses the IP address to select the matching trunk or route pattern and then routes the call to destination stations.
- With the Multiple Call Handling feature, the rerouted or the forwarded-switched calls use the call coverage path of the diverted-to party. Based on the Communication Manager configuration, the greeting of the administered party is played to the caller.
- Delayed drop: With Communication Manager Release 6.3.6, you can use the **Interworking of ISDN Clearing with In-Band Tones** field on the SIP Trunk form to communicate the reason of the call drop to the caller. After knowing that the called party will not answer the call, the caller or the Voice Portal agent can decide whether to wait for the announcement to complete or drop the call.
- Inter-Gateway Alternate Routing: IGAR provides enhanced Quality of Service (QoS) to large distributed single-server configurations. You can use IGAR for configurations where the IP network is not reliable enough to carry bearer traffic. If you have more than one IP network available, you can use H.323 or SIP trunks for IGAR instead of the PSTN. Communication Manager Release 6.3.5 and earlier supported IGAR for analog, DCP, and H.323 endpoints. From Release 6.3.6 onwards, IGAR support is extended to SIP endpoints.

Telecommuting and Remote Office

Telecommuter capabilities route calls appropriately and give employees access to the full Avaya Aura Communication Manager feature set whether working at home, in the office or on the road.

Communication Manager supports the following telecommuting features:

- Coverage of calls redirected off-net. Coverage of calls redirected off-net (CCRON) allows calls that have been redirected to locations outside of the switch to return to the switch for further processing.
- Extended user administration of redirected calls (telecommuting access). Extended user administration of redirected calls (also called telecommuting access) allows you to change the lead call coverage path or forwarding extension from any on-site or off-site location.
- Off-premises station. A trunk-data module connects off-premises private-line trunk facilities and Communication Manager.
- Remote access permits authorized callers from remote locations to access the system via the public network and then use its features and services. There are a variety of ways of accessing the feature.

Communication Manager telephony

Communication Manager provides comprehensive end user telephony features (such as, auto attendant, call transfer, call forward, and so on) that facilitate effective communications among employees, customers and partners.

Mid-call features:

Communication Manager ensures that mid-call call telephony features work when Avaya endpoints establish video calls with Radvision endpoints. Customers can use video mute and unmute, transfers, and conferences during a video call.

Exclusion:

Users can maintain privacy of their telephonic conversations and ensure that unwanted parties will be unable to join the call. You can use Exclusion with Extension To Cellular, Bridge Call Appearance, and Service Observing.

Concurrent call management:

If the Limit Number of Concurrent Calls (LNCC) feature is enabled on a station, Communication Manager restricts the number of incoming calls to one call at a time. If the user is busy, the subsequent incoming calls receive a busy tone. Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP telephones. Communication Manager Release 6.3 extends this support to SIP telephones.

Call log support

Communication Manager records all missed calls in the missed call log of 94xx deskphones.

Call log support for busy 94xx deskphones

Communication Manager 6.3.2 records all incoming calls when a 94xx deskphone is busy because:

- All but one call appearances reserved for incoming calls are in the non-idle state. The last call appearance is reserved for outgoing calls.
- All call appearances are in the non-idle state.
- The Do Not Disturb feature is active on the endpoint.
- One call appearance is busy on a call because a remote user has put the call on hold or started a transfer or a conference call.

Supported number of digits in a call log

For a direct incoming external call from an ISDN or a SIP network, Communication Manager displays up to 21 digits of the calling-party number on a DCP, an H.323, or a SIP endpoint. Earlier, Communication Manager displayed only 7 digits of the calling-party number.

The missed call log and the answered call log of the endpoints display all 21 digits. Communication Manager also stores all 21 digits of an incoming external call from an ISDN or SIP network that is redirected by coverage, forwarding, bridging, or a similar feature in the missed call log and the answered call log of the endpoints

Online/Offline Call Journal (Call History)

With the Online/Offline Call Journal (Call History) feature, the SIP and H.323 phone users can view the call log entries when the user logs in from a different H.323 or SIP device. The SIP and H.323 users receive the logs for all answered and unanswered calls while the phones were in the logged-out state. With this enhancement, the H.323 and SIP desk phones back up all the call logs and restore them when the user logs in.

Communication Manager backs up up to 10 calls for the logged out H.323 users. Communication Manager does not back up or restore the log for calls that are answered or unanswered by the H. 323 phones when in the logged-in state. The H.323 phones continue to use HTTP for this purpose.

Call notification

SIP undelivered call notification:

The SIP undelivered call notification feature provides a notification about the undelivered call to the endpoint. Communication Manager initiates the SIP undelivered call notification feature when a SIP endpoint receives a call in one of the following situations:

- All call appearances are busy.
- LNCC is activated and the endpoint is busy.
- Call Forward Busy or Call Forward All is enabled.
- Enhanced Call Forward (ECF) unconditional or ECF busy is enabled.
- Cover All Calls is enabled.

Codec support

Communication Manager supports G.722 wideband audio codec between H.323 endpoints and SIP video and audio endpoints.

Chapter 5: Communication Manager features

Administration features

Communication Manager supports several administration interfaces for ease of use. See *Administering Avaya Aura® Communication Manager* for more information.

- System Access Terminal (SAT) uses a Command Line Interface (CLI) interface for telephony administration. SAT is available through the Avaya Site Administration package. The system-level limit on the number of concurrent SAT sessions is 22. This limit is only for login profiles 18 to 69 and not for system logins. A user can have up to 5 concurrent SAT sessions.
- System Management interface.
- System Manager.
- Solution Deployment Manager: The Solution Deployment Manager utility resides in System Manager. With Solution Deployment Manager, you can install the Avaya OVAs, and perform administrative activities.

Communication Manager labels each point-to-point session with a globally unique identifier by generating a 128-bit identifier and inserting the identifier in the Global Session ID (GSID) header of the request. To troubleshoot call flows, you can use a tracing tool and filter GSIDs from the relevant logged messages.

Communication Manager attendant features

Communication Manager contains many features that provide easy ways to communicate through your telephone system attendant (operator). In addition, attendants can connect to their console (switchboard) from other telephones in your system, thereby expanding the attendant capabilities.

- Attendant backup. The attendant backup feature allows you to access most attendant console features from one or more specially-administered backup telephones. This allows you to answer calls more promptly, thus providing better service to your guests and prospective clients.
- Attendant room status. Communication Manager allows an attendant to see whether a room is vacant or occupied, and what the housekeeping status of each room is.

Note:

This feature is available only when you have enhanced hospitality enabled for your system.

- Attendant functions using Distributed Communications System protocol.
  - Control of trunk group access allows an attendant at any node in the Distributed Communications System (DCS) to take control of any outgoing trunk group at an adjacent node.
  - Direct trunk group selection allows the attendant direct access to an idle outgoing trunk in a local or remote trunk group by pressing the button assigned to that trunk group.
  - Inter-PBX attendant calls allows attendants for multiple branches to be concentrated at a main location.
- Call handling.
  - Attendant Intrusion. Use the Attendant Intrusion feature to allow an attendant to intrude on an existing call. The Attendant Intrusion feature is also called Call Offer.
  - Attendant lockout - privacy. This feature prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.
  - Attendant split swap. The attendant split swap feature allows the attendant to alternate between active and split calls. This operation may be useful if the attendant needs to transfer a call but first must talk independently with each party before completing the transfer.
  - Attendant vectoring. Attendant vectoring provides a highly flexible approach for managing incoming calls to an attendant. For example, with current night service operation, calls redirected from the attendant console to a night station can ring only at that station and will not follow any coverage path.
  - Automated attendant. Automated attendant allows the calling party to enter the number of any extension on the system. The call is then routed to the extension. This allows you to reduce cost by reducing the need for live attendants.
  - Backup alerting. The backup alerting feature notifies backup attendants that the primary attendant cannot pick up a call.
  - Call waiting. Call waiting allows an attendant to let a single-line telephone user who is on the telephone know that a call is waiting. The attendant is then free to answer other calls. The attendant hears a call waiting ringback tone and the busy telephone user hears a call waiting tone. This tone is heard only by the called telephone user.
  - Calling of inward restricted stations. A telephone with a class of restriction (COR) that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. This feature allows you to override this restriction.
  - Conference. The conference feature allows an attendant to set up a conference call for as many as six conferees, including the attendant. Conferences from inside and outside the system can be added to the conference call.
  - Enhanced Return Call to (same) Attendant. Communication Manager provides individual queuing functions for each attendant supporting a multiplicity of waiting calls at a given time.

- Listed directory number. Allows outside callers to access your attendant group in two ways, depending on the type of trunk used for the incoming call.
- Override of diversion features. The override of diversion feature allows an attendant to bypass diversion features such as send all calls and call coverage by putting a call through to an extension even when these diversion features are on. This feature, together with attendant intrusion, can be used to get an emergency or urgent call through to a telephone user.
- Priority queue. Priority queue places incoming calls to the attendant in an orderly queue when these calls cannot go immediately to the attendant.
- Release loop operation. Release loop operation allows the attendant to hold a call at the console if the call cannot immediately go through to the person being called. A timed reminder begins once the call is on hold.
- Selective conference mute. Selective conference mute allows a conference call participant, who has a display station, to mute a noisy trunk line. Selective conference mute is also known as far end mute.
- Serial calling. The serial calling feature enables an attendant to transfer trunk calls that return to the same attendant after the called party hangs up. The returned call can then transfer to another station within the switch. This feature is useful if trunks are scarce and direct inward dialing services are unavailable.
- Timed reminder and attendant timers. Attendant timers automatically alert the attendant after an administered time interval for the certain types of calls.

• Centralized Attendant Service. Centralized Attendant Service (CAS) enables attendant services in a private network to be concentrated at a central location. Each branch in a centralized attendant service has its own listed directory number or other type of access from the public network. Incoming calls to the branch, as well as calls made by users directly to the attendants, are routed to the centralized attendants over release link trunks.

• Display. The display feature shows call-related information that helps the attendant to operate the console. This feature also shows personal service and message information.

• Making calls.
  - Auto Start and Do Not Split. The Auto Start feature allows the attendant to make a telephone call without pushing the start button first. If the attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the second call.
  - Auto Manual Splitting. Auto Manual Splitting allows an attendant to announce a call or consult privately with the called party without being heard by the calling party on the call. It splits the calling party away so the attendant can confidentially determine if the called party can accept the call.

• Monitoring calls.
  - Attendant control of trunk group access. Use the Attendant Control of Trunk Group Access feature to allow the attendant to control outgoing and two-way trunk groups.

- Attendant direct extension selection. This feature allows the attendant to keep track of extension status - whether the extension is idle or busy - and to place or extend calls to extension numbers without having to dial the extension number.
- Attendant direct trunk group selection. With this feature, the attendant directs access to an idle outgoing trunk by pressing the button assigned to the trunk group. This feature eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups.
- Crisis alerts to an attendant console. Crisis alert uses both audible and visual alerting to notify attendant consoles when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and displays the caller's name and extension (or room).
- Trunk group busy/warning indicators to attendant. This feature provides the attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy. This feature is particularly helpful to show the attendant that the attendant control of trunk group access feature needs to be invoked.
- Trunk identification by attendant. Trunk identification allows an attendant or display-equipped telephone user to identify a specific trunk being used on a call. This capability is provided by assigning a trunk ID button to the attendant console or telephone. This feature is particularly helpful for identifying a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.
- Visually Impaired Attendant Service. Visually Impaired Attendant Service (VIAS) provides voice feedback to a visually impaired attendant. Each voice phrase is a sequence of one or more single-voiced messages. This feature defines six attendant buttons to aid visually impaired attendants.

Communication Manager customization features

Communication Manager allows you to customize interfaces with Avaya and third-party adjuncts and solutions.

- An Application Programming Interface (API) allows numerous software applications to work with Communication Manager. APIs also allow a client programmer to create their own applications that work with Communication Manager.
- Application Enablement Services (AE Services) is a connector that provides connectivity between applications and Communication Manager. This connector allows development of new applications and new features without having to modify Communication Manager or expose its proprietary interfaces.

  Note:

  AE Services has its own set of customer documentation, including an overview. This Overview of Communication Manager does not outline the changes to AE Services.

- Device and media control API. Device and media control API provides a connector to Communication Manager that allows clients to develop applications that provide first party call control. Applications can register as IP extensions on Communication Manager and then monitor and control those extensions.

  Device and media control API consists of connector server software and a connector client API library. The connector server software runs on a hardware server that is independent from Communication Manager. That is, device and media control API does not run co-resident with Communication Manager.

  Tip:

  Ask your Avaya representative for a complete list of device and media control API documentation.
- Co-resident branch Gateway. In simplest terms, the branch Gateway is an application that enables communications between TCP/IP clients and Communication Manager call processing. In more technical terms, the application is software that both routes internetwork messages from one protocol to another (ISDN to TCP/IP) and bridges all ASAI message traffic by way of a TCP/IP tunnel protocol.
- Java telephony application programming interface (JTAPI) is an open API supported by Avaya computer telephony that enables integration to Communication Manager ASAI.
- Telephony Services Application Programming Interface (TSAPI) is an open API supported by Avaya computer telephony that allows integration to Communication Manager ASAI. TSAPI is based on international standards for CTI telephony services. Specifically, the European Computer Manufacturers Association (ECMA) CTI standard definition of Computer-Supported Telecommunications Applications (CSTA) is the foundation for TSAPI.
- Use the Automatic Number Identification (ANI) feature to display the telephone number of the calling party on your display telephone. The system uses ANI to interpret calling party information that is signaled over multifrequency (MF) or Session Initiation Protocol (SIP) trunks. Any display telephone can use ANI.
- For H.323 and DCP endpoints, the caller information on the bridged call appearances can be set to be the same as the caller information on the principal station. To enable this feature, set the Match BCA Display to Principal field on page 2 of the Class of Service screen to y.

Scalability

System capacities have been expanded for many products and features.

For the entire list of updated capacities, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Communication Manager reliability

Communication Manager supports a wide variety of servers, gateways and survivability features enabling maximum availability for any customer. The software is capable of mirroring processor functions, providing alternate gatekeepers, supporting multiple network interfaces and ensuring survivability at remote and central locations.

Communication Manager reliability features include:

- Alternate gatekeeper. The alternate gatekeeper feature can provide survivability between Communication Manager and IP communications devices such as IP telephones and IP softphones.
- Auto fallback to primary for branch gateways. This feature automatically returns a fragmented network, where a number of branch gateways are being serviced by one or more Communication Manager Survivable Remote sites, to the primary (main) server. This feature is targeted to branch gateways only.
- Connection preserving failover/failback for branch gateways. The Connection Preserving Migration (CPM) feature preserves existing bearer (voice) connections while an branch gateways migrates from one Communication Manager server to another. Migration might be caused by a network or server failure.
- Connection preserving upgrades for duplex servers. The connection preserving upgrades for duplex servers feature provides connection preservation on upgrades of duplex servers for:
    - connections involving IP telephones
    - connections involving TDM connections on port networks
    - connections on branch gateways
    - IP connections between port networks and branch gateways
- Communication Manager Survivable Core provides survivability by allowing backup servers to be placed in various locations in the customer network. The backup servers supply service to port networks in the case where the main server or server pair fails, or connectivity to the main server or server pair is lost.
    - Automatic return to primary (main) server. When the Survivable Core is in control due to a network fragmentation or catastrophic main server failure, the return to the main server is predicated by the scheduled, manual, and automatic options.
    - The Dial Plan Transparency for Survivable Remote and Survivable Core preserves users' dialing patterns if a branch gateway registers with Survivable Remote, or when a port network registers with Survivable Core.
- IP bearer duplication using the TN2602AP circuit pack. The TN2602AP IP Media Resource 320 circuit pack provides high-capacity voice over Internet protocol (VoIP) audio access to the switch for local stations and outside trunks.
    - 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.

- Bearer signal duplication. Two TN2602AP circuit packs may be installed in a single port network for bearer signal duplication. In this configuration, one TN2602AP is an active IP media processor and one is a standby IP media processor.

- IP endpoint Time-to-Service The IP endpoint time-to-service (TTS) feature improves a customer's IP endpoint time to service, especially in cases where the system has a lot of IP endpoints trying to register or re-register. With this feature, the system considers that IP endpoints are in-service immediately after they register.
- A survivable processor is an Internal Call Controller (ICC) with an integral branch gateway, in which the ICC is administered to behave as a spare processor rather than as the main processor. The standby Avaya S8300 Server runs in standby mode with the main server ready to take control in the event of a outage with no loss of communication.
- Handling of split registrations. Split registrations occur when resources on one network region are registered to different servers. For example, after an outage activates the Survivable Remote server (Local Survivable Processors) or Survivable Core server (Enterprise Survivable Server), telephones in a network region register to the main server, while the branch gateways in that network region are registered with the Survivable Remote server. The telephones registered with the main server are isolated from their trunk resources. Communication Manager detects a split registration and moves telephones to a server that has trunk resources.
- Power failure transfer provides service to and from the local telephone company central office (CO), including wide area telecommunications system, during a power failure. This allows you to make or answer important or emergency calls during a power failure. This feature is also called emergency transfer.
- Standard Local Survivability. Standard Local Survivability (SLS) provides a local Avaya G430 or G450 Branch Gateway and Juniper J4350 or J6350 gateways with a limited subset of Communication Manager functionality when there is no IP-routed WAN link available to the main server or when the main server is unavailable.
- Communication Manager supports SRTP for video call flows. This support is available only when the call-originating and the receiving endpoints are SIP-registered and the IP-codec-set administration on Communication Manager is SRTP. SRTP for video does not work for H.323 signaling. H.323-registered endpoints always send video RTP. SIP-H.323 interworking with video encryption is not supported and video is blocked in this case. However, if the SIP signaling follows the Best effort SRTP mode, Communication Manager allows video RTP to pass through in SIP to H.323 interworking.

Communication Manager security, privacy and safety

Communication Manager provides security features for detecting probable breaches, taking measures to protect the system, notification and tracking activities. It also provides real-time media encryption for environments where enhanced voice privacy over a LAN/WAN is required.

Communication Manager supports:

- Industry Standard STRP (Secure Real Time Protocol) for authentication and media encryption,
- Real Time Media and Signaling Encryption
- Access Security Gateway
- Malicious Call Tracking
- Toll Fraud protection
- Emergency Calling Services (E911)

You can isolate Communication Manager telephony servers from the rest of the enterprise network to safeguard them from viruses, worms, DoS (Denial of Service) and other attacks. It uses the minimum number of services and access ports to reduce susceptibility to malicious attacks and employs encryption between servers, gateways and endpoints to secure the voice stream and signaling channels.

See *Avaya Aura® Communication Manager Security Design* for further information.

Related links

NIST support on page 43

NIST support

The National Institute of Standards and Technology (NIST) develops cryptographic standards for the United States government. NIST recommends that starting in 2014, the digital signatures of Identity Certificates use SHA2 hashing and 2048–bit encryption keys. NIST requires at least 2048 bit keys.

With theCommunication Manager Release 6.3.6 security service patch, you can receive and validate the certificate that uses the SHA-2 signing algorithm and 2048 bit RSA keys. Using Communication Manager System Management Interface, you can import the third-party trusted certificate that uses the SHA-2 signing algorithm.

Related links

Communication Manager security, privacy and safety on page 42

Communication Manager localization

Communication Manager supports a range of language features, such as administrable language displays and country-specific localization.

Communication Manager localization features:

- Administrable language displays. This feature allows messages that appear on telephone display units to be shown in the language spoken by the user. These messages are available in English (the default), French, Italian, Spanish, user-defined, or Unicode; where user-defined can be almost any language using the Latin, Russian or Katakana writing scripts, and Unicode can be almost any language in the world. The language for display messages is selected for each user by the administrator. The feature requires 40-character display telephones.

- Administrable loss plan. The administrable loss plan provides the ability to administer signal loss and gain for telephone calls. This capability is necessary because the amount of loss allowed on voice calls can vary by country.
- Bellcore calling name ID. This feature allows the system to accept calling name information from a Local Exchange Carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore calling name ID is administered. The following caller ID protocols are supported.
    - Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol).
    - V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).
- Busy tone disconnect. In some regions of the world, the CO sends a busy tone for the disconnect message. With busy tone disconnect, the switch disconnects analog loop-start CO trunks when a busy tone is sent from the CO.
- Country-specific localization
    - Brazil. Block collect call. This feature blocks collect calls on class-of-restriction basis. This feature is available for any switch that uses the Brazil country code.
    - Italy. Distributed Communications Systems protocol Italian DCS adds features to the existing DCS capabilities and requires the use of Italian TGU/TGE tie trunks.
    - Japan.
        - National private networking provides support for Japanese private ISDN networks.
        - Katakana character set Communication Manager supports the Katakana character set .
    - Russia
        - Central Office support on branch gateways. Communication Manager supports central office (CO) trunks in Russia using Avaya branch gateways.
        - ISDN/DATS network support. This feature supports ISDN/DATS trunk networks when the tone generated field is set to 15 (Russia) on the system-parameters tone—generation screen. It modifies the overlap sending delay and ISDN T302 and T304 timers to support the Russian trunk network.
        - Multi-Frequency Packet signaling. Multi-Frequency Packet (MFP) address signaling is provided in Russia on outgoing CO trunks. Calling party number and dialed number information is sent on outgoing links between local and toll switches.
- E&M signaling - E&M trunks are used to provide analog communications links. Continuous and pulsed Continuous and pulsed E&M signaling is a modification to the E&M signaling used in the United States. Continuous E&M signaling is intended for use in Brazil, but can also be used in Hungary. Pulsed E&M signaling is intended for use in Brazil.
- Multinational Locations. For customers who operate in more than one country, the Multinational Locations feature provides the ability to use a single Enterprise Communication Server (ECS) across multiple countries.

- Public network call priority provides call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to switches on public networks. Different countries frequently refer to these capabilities by different names.
- QSIG support for Unicode. The QSIG support for Unicode feature extends the Unicode support on a single server to multi-node Communication Manager networks. This feature allows Unicode support across large campus configurations.
- World class tone detection. World class tone detection enables Communication Manager to identify and handle different types of call progress tones, depending on the system administration.
- XOIP Tone Detection Bypass. The X over IP Tone Detection Bypass feature (where X = modem, fax, TTY-TDD, and so on) serves customers using older or non-standard external equipment such as modems, fax, TTY devices which are not easily recognized by VoIP resources within Communication Manager.

Chapter 6: Resources

Documentation

The following table lists the documents related to this product. Download the documents from the Avaya Support website at http://support.avaya.com.

Title	Description	Audience
Design		
Avaya Aura® Communication Manager Security Design, 03-601973	Describes security-related issues and security features of Communication Manager.	Sales Engineers, Solution Architects
Avaya Aura® Solution Design Considerations and Guidelines, 03-603978	Describes the Avaya Aura® solution, IP and SIP telephony product deployment, and network requirements for integrating IP and SIP telephony products with an IP network.	Sales Engineers, Solution Architects
Avaya Aura® Communication Manager System Capacities Table, 03-300511	Describes the system capacities for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects
Maintenance and Troubleshooting		
Avaya Aura® Communication Manager Reports, 555-233-505	Describes the reports for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Maintenance Alarms for Avaya Aura® Communication Manager, Branch Gateways Servers, 03-300430	Provides procedures to monitor, test, and maintain an Avaya server or Media Gateway.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers, 03-300431	Provides information to monitor, test, and maintain hardware components of an Avaya servers or Gateways.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Avaya Aura® Communication Manager Server Alarms, 03-602798	Provides procedures to monitor, test, and maintain an Avaya servers.	Sales Engineers, Solution Architects, Implementation

Table continues…

Title	Description	Audience
		Engineers, Support Personnel
Avaya Aura® Communication Manager Denial Events, 03-602793	Describes the denial events listed on the Events Report form.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Avaya Aura® Toll Fraud and Security Handbook, 555-025-600	Describes the security risks and measures that can help prevent external telecommunications fraud involving Avaya products.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Administration		
Administering Avaya Aura® Communication Manager, 03-300509	Describes the procedures and screens for administering Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
Administering Network Connectivity on Avaya Aura® Communication Manager, 555-233-504	Describes the network connectivity for Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
Administering Avaya Aura® System Manager for Release 7.0.1	Describes procedures for managing the features that are part of Solution Deployment Manager for Communication Manager.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Implementation and Upgrading		
Deploying Avaya Aura® Communication Manager	Describes the implementation instructions while deploying Communication Manager on VMware.	Implementation Engineers, Support Personnel, Solution Architects
Deploying Avaya Aura® applications from System Manager	Describes the implementation instructions while deploying and configuring Solution Deployment Manager for Communication Manager.	Implementation Engineers, Support Personnel, Solution Architects
Upgrading and Migrating Avaya Aura® applications to Release 7.0.1 from System Manager	Describes the implementation instructions while deploying and configuring Solution Deployment Manager for Communication Manager.	Implementation Engineers, Support Personnel, Solution Architects
Understanding		
Avaya Aura® Communication Manager Feature Description and Implementation, 555-245-205	Describes the features that you can administer using Communication Manager.	Sales Engineers, Solution Architects, Support Personnel

Table continues…

Title	Description	Audience
Avaya Aura® Communication Manager Screen Reference, 03-602878	Describes the screens that you can administer using Communication Manager.	Sales Engineers, Solution Architects, Support Personnel
Avaya Aura® Call Center Elite Overview and Specification	Describes tested product characteristics and capabilities, including product overview and feature descriptions, interoperability, performance specifications, security, and licensing requirements.	Sales Engineers, Solution Architects, Support Personnel
What's New in Avaya Aura® Release 7.0.1, 03-601818	Describes the new features for the current release of Avaya Aura®.	Sales Engineers, Solution Architects, Support Personnel
Avaya Aura® Communication Manager Special Application Features	Describes the special features that are requested by specific customers for their specific requirement.	Sales Engineers, Solution Architects, Avaya Business Partners, Support Personnel

Training

The following courses are available on the Avaya Learning website at www.avaya-learning.com. After logging into the website, enter the course code or the course title in the **Search** field and click **Go** to search for the course.

Course code	Course title
Understanding	
1A00234E	Avaya Aura® Fundamental Technology
AVA00383WEN	Avaya Aura® Communication Manager Overview
ATI01672VEN, AVA00832WEN, AVA00832VEN	Avaya Aura® Communication Manager Fundamentals
2007V	What is New in Avaya Aura® 7.0
2009V	What is New in Avaya Aura® Communication Manager 7.0
2011V	What is New in Avaya Aura® System Manager & Avaya Aura® Session Manager 7.0
2009T	What is New in Avaya Aura® Communication Manager 7.0 Online Test
2013V	Avaya Aura® 7.0 Solution Management
5U00060E	Knowledge Access: ACSS - Avaya Aura® Communication Manager and CM Messaging Embedded Support (6 months)
Implementation and Upgrading	

Table continues…

Course code	Course title
4U00030E	Avaya Aura® Communication Manager and CM Messaging Implementation
ATC00838VEN	Avaya Media Servers and Implementation Workshop Labs
AVA00838H00	Avaya Media Servers and Media Gateways Implementation Workshop
ATC00838VEN	Avaya Media Servers and Gateways Implementation Workshop Labs
2012V	Migrating and Upgrading to Avaya Aura® 7.0
Administration	
AVA00279WEN	Communication Manager - Configuring Basic Features
AVA00836H00	Communication Manager Basic Administration
AVA00835WEN	Avaya Communication Manager Trunk and Routing Administration
5U00041I	Avaya Aura® Communication Manager Administration
AVA00833WEN	Avaya Communication Manager - Call Permissions
AVA00834WEN	Avaya Communication Manager - System Features and Administration
5U00051E	Knowledge Access: Avaya Aura® Communication Manager Administration

Viewing Avaya Mentor videos

Avaya Mentor videos provide technical content on how to install, configure, and troubleshoot Avaya products.

About this task

Videos are available on the Avaya Support website, listed under the video document type, and on the Avaya-run channel on YouTube.

Procedure

- To find videos on the Avaya Support website, go to http://support.avaya.com and perform one of the following actions:
  - In **Search**, type `Avaya Mentor Videos` to see a list of the available videos.
  - In **Search**, type the product name. On the Search Results page, select **Video** in the **Content Type** column on the left.
- To find the Avaya Mentor videos on YouTube, go to www.youtube.com/AvayaMentor and perform one of the following actions:
  - Enter a key word or key words in the **Search Channel** to search for a specific product or topic.
  - Scroll down Playlists, and click the name of a topic to see the available list of videos posted on the website.

Note:

Videos are not available for all products.

Support

Go to the Avaya Support website at http://support.avaya.com for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

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