What’s New in Avaya Communication Manager

for Release 4.0

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Chapter 1: What's New

This chapter presents highlights of features and enhancements as part of release 4.X of Avaya Communication Manager running on Avaya DEFINITY® servers, as well as the Avaya S8000-series media servers with associated Avaya media gateways.

The most current release of Communication Manager contains the features of prior releases.

> **Important:**
> Not all features in Communication Manager release 4.0 are administered the same way as in prior releases. For example, prior to release 4.0, you would administer logins through SAT screens. Beginning with Communication Manager release 4.0, those login SAT screens no longer exist. There is a new web page for login administration.

In this document, each Communication Manager feature or enhancement is listed alphabetically by release number.

Other documents for Avaya Communication Manager release 4.X can assist you with this What's New document:

- For an overview of the features of Communication Manager, see the *Overview for Avaya Communication Manager*.
- For a more complete description of the features of Communication Manager, see the *Feature Description and Implementation for Avaya Communication Manager*.
- For more information on how to administer all features, and for a description of new and changed screens for this release, see the *Administrator Guide for Avaya Communication Manager*.
- For a description of new and changed commands for this release, see the *Maintenance Commands for Avaya Communication Manager 4.0 Media Gateways and Servers*.
- For a description of new hardware and how Communication Manager 4.0 affects various endpoints, see the *Hardware Guide for Avaya Communication Manager 4.0 Media Gateways and Servers*.

Release 4.0 new features and enhancements

Avaya Communication Manager release 4.0 includes the following general telephony and system-wide features and enhancements.

Note:

Not all CM4 features are supported by all telephones and firmware. For a more complete reference, see the Hardware Guide for your particular telephone and firmware.

Access Security branch gateway authentication

Access security gateway authentication allows the Avaya Services organization to remotely login to Communication Manager media servers that are under service agreement. To enhance customer support in branch offices, ASG is enhanced to allow authentication capabilities to the G350/G250 gateway product line.

Announcements and tone detector enhancements

Starting with Communication Manager 4.0, there are several capacity increases with announcements and tone detectors.

The number of announcement files supported on the TN2501 VAL boards has increased from 256 to 1024 with firmware 17. This increase is available with all Communication Manager server platform configurations.

The following capacity increases are only available with the XL (extra large) S8720 server platform and supported LSP/ESS servers in an S8720XL main server configuration. Supported LSP/ESS servers include the S8720XL ESS, S8500C ESS/LSP or the S8500A/B ESS/LSP with 512 MB RAM.

- You can now have up to 128 TN2501 VAL port network boards (up from 10) within a single Communication Manager configuration. The number of VAL boards that can be installed is a right to use feature and requires a new license file for upgrades.
- The total supported announcement extensions that can be defined on the announcements/ audio sources screen has increased from 3,000 to 9,000. This applies to announcements stored on TN2501 VAL boards and the gateway embedded vVALs announcement sources. This does not apply to TN750 boards which are no longer supported.
- The total number of administered announcement files over all sources (VAL or vVAL) has increased from 3,000 to 12,000. With Locally Sourced Music and Announcements, an audio group listing a number of VAL/vVAL sources can be defined as the location for an announcement file. The file has to be stored in each source in the group so each is

counted as an administered announcement file towards the available capacity. Single sourced announcement defined extensions each use only one administered file.

Both the announcement extension and administered files system limits with use and available quantities are shown on the display capacity screen. For example, a configuration with 5 single sourced announcement extensions and 2 audio group sourced announcement extensions each group listing 10 sources, will use 7 announcement extensions and 25 administered announcement files.

- The number of VAL/vVAL sources that can be included in an audio group is 378, an increase from 250.
- The number of supported port network tone detector boards has been increased from 1,200 to 8,000 boards. This does not apply to the H.248 media gateways which have internal detector boards.

Authentication, Authorization, and Accounting Services

Authentication, Authorization and Accounting (AAA) services allow customers to store and maintain administrator account information on a central server. Communication Manager supports account information being stored on an external AAA server or locally on the Communication Manager server itself. Both types of accounts may be used at the same time.

AAA service interactions with Communication Manager use the Pluggable Authentication Module (PAM) and Name-Switch Service (NSS) features of Linux, which are provided on Linux-based Communication Manager servers.

External AAA support is a Linux process that is separate from Communication Manager, is not controlled by a license file, and is freely available to the customer. Customers can use the same AAA server for Communication Manager as is used by other servers on their network.

For more information, see the *Feature Description and Implementation for Avaya Communication Manager*.

Auto callback - QSIG call completion

Auto callback covers auto callback within a private corporate network only through QSIG. Auto callback provides an administrable option on the Trunk Group screen to allow users to specify the method of signaling connection the system uses while waiting for a busy station to become idle.

Automatic return to primary server

When an Enterprise Survivable Server (ESS) is in control due to a network fragmentation or catastrophic main server failure, the return to the primary (main) server can be handled in three ways:

- Scheduled
- Manual
- Automatic

A timer is associated with the Auto Return To Primary feature. The customer sets the timer after the Auto Return To Primary feature is activated to prevent recovery to the main server before the network is stable. The **Timer** field does not appear until the Auto Return To Primary option is selected. You must set the automatic option prior to any potential network fragmentation.

Important:
Setting the timer for the auto return does not imply that the network will be stable when that timer expires. For example, if a customer chooses 3 minutes as the timer setting, the network may not be completely healthy by then — yet the return will be attempted. To prevent a premature recovery to the main, the manual option is the safest, followed by a scheduled return, then an automatic return. Now you can set the auto return timer for 2 hours (120 minutes), which would hopefully be sufficient time for the network to recover.

Avaya SoftConsole - RoadWarrior mode

Avaya SoftConsole, RoadWarrior mode, enables use of the full Avaya Communication Manager feature set from temporary remote locations anywhere in the world. The RoadWarrior application consists of two software applications running on a PC that is connected to Communication Manager over an IP network.

The single network connection between the PC and Communication Manager carries two channels, one for the signaling path and one for the voice path. On Communication Manager, the RoadWarrior application requires the CLAN circuit pack for signaling and the IP media processor for voice processing.

Avaya SoftConsole - Telecommuter mode

Avaya SoftConsole, Telecommuter mode, enables telecommuters to use the full Communication Manager feature set from home. It consists of a PC and a telephone with separate connections to Communication Manager. The PC provides the signaling path and the user interface for call control. A standard telephone provides a high-quality voice path. The Telecommuter application requires the CLAN circuit pack for signaling. The Telecommuter application does not use the IP media processor.

In telecommuter mode, both the SoftConsole and the physical handset telephone must be answered to answer the call.

Avaya Video Telephony Solution enhancements

The following components are part of the Avaya Video Telephony Solution feature for Communication Manager release 4.0:

- Polycom VSX3000, VSX7000, and VSX8000 conferencing systems with Release 8.03 or later.
- Polycom MGC video conferencing bridge platforms with Release 8.0.1.

  Note:
  Release 7.5 of the MGC is not supported.
- Third party gatekeepers, including Polycom Path Navigator.

To configure the Polycom MGC-25 Video Conferencing Bridge Platforms with Avaya S8300, S8500, and S87xx servers, see the *Video Telephone Solution R3.0 Quick Setup Guide*, 16-300310, Issue 3.

CDR format changes

CDR format fields are increased for Communication Manager release 4.0.

- The **attd-console** field length is increased to a maximum of 4 characters.
- The **ins** field length is increased to a maximum of 5 characters.
- The **calltype** field is a single character field that indicates whether or not the call was handled by call type digit analysis. The **calltype** field in only available in the customized format.

For Communication Manager release 3.X, the CDR format fields are as follows:

- The **attd-console** field length is a maximum of 2 characters.
- The **ins** field length is a maximum of 3 characters.

Most CDR adjunct vendors check the length of the CDR records to validate each CDR record received. If the length of a record is different from what is expected, the record is judged to be invalid and it is discarded. If customers upgrade to Communication Manager release 4.0 without adjusting their CDR formats, records could be lost.

To solve this problem, a new field, **Use Legacy CDR Formats**, is added to the **CDR System Parameters** SAT screen. The default choice is **y**.

Administrators can now select whether they want Communication Manager release 3.x or 4.0 CDR formats using the **Use Legacy CDR Formats** field. Using the default choice of **y**, Communication Manager uses release 3.x formats. If you change the value of the **Use Legacy CDR Formats** field to **n**, Communication Manager uses release 4.0 formats.

Complex private numbering plan support

Additional flexibility is provided for private-network numbering in support of customers' private networks.

Daylight Savings Time rules change

Communication Manager is updated to comply with the new Daylight Saving Time (DST) in the United States. DST is extended beginning in 2007. DST now starts the second Sunday of March, and ends on the first Sunday of November.

Dial Plan Expansion

With Communication Manager release 4.0, you can expand your dial plan to a maximum of 13 digits. This affects telephones, data modules, login IDs, and vectors.

Administrators have the flexibility to administer dial plans between 3 and 13 digits in length. Communication Manager supports mixed digit lengths in the same dial plan within a location and across a network of locations.

Customers upgrading to Communication Manager 4.0 can choose to migrate to the expanded 13-digit dial plan. Customers who choose not to migrate may convert their dial plans at a later date.

Distributed Communications System (DCS) protocol is limited to a dial plan of 3-5 digits. If your dial plan requires 6 or 7 digits, QSIG, which is the generic name for a family of signaling protocols, is required. Also, if Call Management System (CMS) is administered as a reporting adjunct, administration of extensions longer than 7 digits is not allowed.

Some features are not expanding to 13 digits. For a complete description of the Dial Plan Expansion feature, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Dial Plan Extension capacity increases

With S8720XL configurations, the total number of dial plan extensions capacity has been increased to 55,733, an increase from 49,733. The miscellaneous extensions capacity has been increased to 32,508, an increase from 26,508.

These increased limits are displayed on the **System Capacity** screen.

Dial Plan Transparency for LSP and ESS

The Dial Plan Transparency feature preserves the dialing patterns of users if a media gateway registers with a local survivable processor (LSP), or when a port network registers with an Enterprise Survivable Server (ESS).

When a media gateway registers with a local survivable processor (LSP), or when a port network registers with an Enterprise Survivable Server (ESS), the Dial Plan Transparency feature routes calls over the public network when they cannot be routed over the IP network.

Note:

This feature does not provide alternate routing for calls between port networks connected through networks other than IP (e.g., ATM, CSS, or DS1C), and that request service from different ESS servers during a network outage.

This feature lets users continue their dialing patterns when LSP or ESS fragments exist, but it does not guarantee feature transparency for the calls. In most cases, only basic trunk features are available to the calling and called parties.

Enbloc Dialing and Call Type Digit Analysis

The Enbloc Dialing and Call Type Digit Analysis feature allows users to automatically place outgoing calls based on the telephone number information in the telephone's call log, or from the contacts or corporate directory, without the user having to modify the telephone number.

Enbloc Dialing is the ability of Communication Manager software to receive in one message all of the digits necessary to set up a call, or any time the terminal makes a request rather than receiving digits one at a time.

Call Type Digit Analysis is the ability for Communication Manager to decide how to route a call based on all of the digits in a telephone number in a call log, or from the contacts or corporate directory.

For a complete description of Enbloc Dialing and Call Type Digit Analysis, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Enhanced Call Forwarding

The Enhanced Call Forwarding (ECF) feature forwards incoming calls to different destinations depending on whether the incoming calls are from internal or external sources.

Enhanced security logging

Enhanced security logging increases the granularity of logging of user activity, and allows you to specify an external Linux syslog server to which to send a copy of system logs. Enhanced security logging consolidates several existing Communication Manager log files, and routes copies of the files to an industry standard external log server and the internal Linux syslog.

SAT activities are logged according to a logging level set by the administrator using the **SAT Logging Levels** screen.

On the Integrated Management Maintenance Web Pages, use the **Syslog Server** web screen to enable or disable the ability to send logs to an external server, and to specify the logs to be sent.

Enterprise Mobility User enhancements

Enhancements to the Enterprise Mobility User (EMU) feature supports calls to Extension to Cellular, and also allows the home station of a user registered as an EMU visitor elsewhere in the enterprise network to be visited by another EMU user.

For a complete description of the enhancements to Enterprise Mobility User, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Erase user data from DCP telephones

Administrators can delete local information that is stored in a DCP telephone when the telephone is reset by a PSA logoff or similar event. Depending on Communication Manager administration, this feature eliminates speed dial lists, call logs, or option settings that were defined by the user and may be of a personal nature. System administrators are able to re-assign terminals without the need to send a technician to user's desk to manually erase the data.

Upon the following events:

- PSA disassociate & TTI unmerge
- Use by a Enterprise Mobility User visiting extension
- A new 'erase terminal' SAT command

you can administer Communication Manager to erase the following local data items from R4 of the 2420 and R2 of the 2410 DCP sets:

- Don't erase anything
- Call Log
- Speed Dial List, Button Labels, User option settings
- Language

Note:

Communication Manager does not erase data from an IP terminal upon receiving an unregistration request from the terminal. Avaya IP terminals do not allow users access to local terminal information while the terminal is unregistered.

For more information, see the *Administrator Guide for Avaya Communication Manager*.

ETSI completion of calls

The auto-callback feature known as ETSI Completion of Calls to Busy Subscriber (CCBS) and Completion of Calls on No Reply (CCNR) works over the DSS1 network. An analog phone actives this feature by dialing the Feature Access Code (FAC). A multi-appearance voice terminal can use either the FAC or programmed buttons. By pressing an idle Automatic Callback button, a caller can activate either the SS-CCBS when the called station is busy, or the SS-CCNR when the called station does not answer.

The originating caller receives a callback when:

- The busy station becomes available after the user hangs up.
- An unanswered station becomes available after it is used for another call and then hangs up.

The call-back feature expires after a period of 40 minutes.

Extension to Cellular enhancements

The Extension to Cellular (EC500) feature for Communication Manager 4.0 supports Dial Plan expansion to 13-digits, and has new RTUs for Fixed Mobile Convergence (FMC) applications. In addition, the feature now supports digital R2FMC trunks, CLID support for origination mapping, and CTI support and Mobile Feature Extensions.

For a complete description of the enhancements to Extension to Cellular, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

G.722 shuffling over H.323/SIP trunks

During a typical phone call that goes over the PSTN, it is sometimes difficult to distinguish certain letter sounds, such as "f" and "s" or "t" and "d." The reason for this is that most of the speech energy that allows humans to distinguish consonants occurs above 3,000 Hz. The best digital PSTN phone call doesn't pass any energy above 3,300 Hz.

High-fidelity audio (sometimes called wideband audio) passes audio frequencies of up to 7,000 Hz. Research has shown that conversations over these higher frequencies dramatically improve intelligibility and reduce listener fatigue. They also make it much easier to understand accented speech.

G.722 is an audio coding system (50 to 7,000 Hz) that can be used in a variety of higher quality speech applications. This feature supports endpoints capable of handling G.722 codecs, but it is specifically intended for the Avaya 96xx H.323 telephones and G.722-capable conference bridge endpoints.

Improved button downloads

The improved button download to IP endpoints feature greatly reduces the time needed for button labels to be sent to the IP telephone upon a new registration. This feature also reduces the traffic being sent to the IP telephones when button labels have not changed on a reregistration.

The Avaya H.323-based IP telephones request whether they need feature button information during registrations or re-registrations. If the IP telephones request the button information, or if Communication Manager is aware of an administration change that occurred to change button information, Communication Manager then downloads that button information, including the labels.

Increased attendant consoles

Depending on the platform, or server, you are using, Communication Manager supports an increase of attendant consoles. For example, an S8720XL platform supports 414 attendants, while a regular S8720 platform supports 128 attendants. This limit applies to both attendant console telephones with direct extension selectors (DXS), as well as soft attendant consoles.

The attendant consoles may collectively manage a single defined call load. Alternately, each attendant console may be defined so that it manages its own unique call load.

Increased trunk capacity

Communication Manager trunk capacities, which include all IP and TDM trunk types, are increased from 8,000 to 12,000 for any single Communication Manager server within a configuration. In some configurations, Communication Manager trunk capacities support 12,000 trunks, multiplied by the number of Communication Manager servers. The maximum trunk capacity is 96,000 (12,000 trunks x 8 Communication Manager servers).

Integrated Directory enhancements

The Integrated Directory feature is enhanced with Communication Manager release 4.0. One enhancement provides the ability to look up names with international characters (Latin letter with a diacritic) when you set **Display Character Set** = **roman**. Another enhancement is the ability to search on names using Cyrillic characters (**Display Character Set** = **cyrillic** or **ukrainian**).

Along with these enhancements, some changes were made to the Integrated Directory feature operation. The changes are:

- The ability to hide names (**Display Character Set** = **roman**) can only be accomplished if the name begins with two tildes (**~~**). Prior to Communication Manager release 4.0, you may have used other symbols to hide names. This practice is no longer supported.

  If your site was using a symbol other then the tilde to hide names, the recommended upgrade procedure is to export such names to a CSV file. Do a global search on the symbol that you used, and replace the symbol with two tildes. After the change, import the data back to your Communication Manager server.

- Prior to Communication Manager release 4.0, the asterisk was used as a wild card to skip the comma in integrated directory names such as “Doe, John”. The comma is now supported on button 1. When you want to move past the comma in any integrated directory name, press the **1** key instead of the asterisk key.

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.

IP trunk fallback to PSTN

Communication Manager for 4.0 includes a PSTN fallback for IP trunks feature for H.248 media gateways. This feature adds to the feature list of the PSTN fallback of IP trunks feature.

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. The skipping over of IP trunks allows existing route patterns to look at secondary trunk groups for outgoing calls. When these secondary trunk groups are not IP, the PSTN fallback feature applies. The key to the existing implementation is the evaluation of the current network condition.

Evaluation data is obtained from ICMP PING packets that are generated from the TN2302/TN2602 media processor circuit packs. Reports are generated from the circuit packs that aggregate many individual ICMP PING requests.

Before this feature, the existing family of H.248 media gateways (G250, G350, and G700) do not provide this functionality. In systems which do not contain TN media processors, or where those media processors don't have access to the network regions that are reachable by H.248 gateways, there is no PSTN fallback capability for IP trunks in the event of network degradation.

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. When a failure is detected, Communication Manager initiates a timer to allow time for the network to recover. If the network does recover within that period, calls are allowed to continue without interruption.

ISO 8859-1 encoding support

Communication Manager supports ISO 8859-1 encoding capability. For more information, see the *Administrator Guide for Avaya Communication Manager*.

Limit Number of Concurrent Calls

Beginning with Communication Manager 4.0, you can use the Limit Number of Concurrent Calls (LNCC) feature to restrict the number of calls that can terminate on an active multi-appearance terminal to one call.

A typical Communication Manager digital/IP station has three call appearances. The first two call appearances are for receiving incoming calls. If the first two call appearances are active, the last call appearance can be reserved for call origination or receiving priority, provided the **Restrict Last Appearance** field Detailed description of Call Coverage on the **Station** screen is set to **y**. With the LNCC feature enabled, and the user on a call, subsequent incoming calls receive a busy signal.

The LNCC feature does not restrict outgoing calls, incoming priority calls, 911 emergency callback, and crisis alert button activation. The LNCC feature applies to station directed calls only. That is, if LNCC is activated on a station by a logged in agent, LNCC is still active on the station after the agent logs off.

The LNCC feature is considered to be part of the station's attributes and is saved as part of the station record by a save translations. Subsequent resets will restore the LNCC state to what it was when the save translations was executed.

The feature is activated and deactivated by a new feature button (**limit-call**), or by using two new feature access codes (Limit Number of Concurrent Calls Activation/Deactivation). The **limit-call** button indicates whether the LNCC feature is active or not.

Mask CLI/Station Name for Internal Calls

A typical Communication Manager internal call generally provides calling/called party numbers and administered text name string, which are displayed on the involved parties' terminals.

Beginning with Communication Manager 4.0, the Mask CLI/Station Name for Internal Calls feature provides a new station capability (via proper COR administration) of masking off such information and replacing it with a "hard-coded," system-wide text string, "Info Restricted," which will be displayed on the party's terminal. In addition, the feature shall provide a means of administering that the CPN/Name information not be masked-off when the call is made.

Before this feature, Communication Manager can block calling party number and name to be sent to the network with proper trunk/station administration. This new feature takes a similar approach to these existing Communication Manager capabilities whenever it is applicable for internal calls.

The Class of Restriction screen has a new field called Mask CPN/NAME for Internal Calls. Enter y to hide the display of calling/called party numbers and administered name on internal calls.

The Class of Service screen also has a new field called Masking CPN/Name Override. Setting y on this field allows users to override the Mask CLI/Station Name for Internal Calls capability.

For more information on these fields, see the Screen Reference chapter of the *Administrator Guide for Avaya Communication Manager*.

Merge of IP-PNC and fiber-PNC configurations

Communication Manager uses web pages to configure media servers. The customer can now choose how to configure each Ethernet Network Interfaces (NIC) based on their network and policies. This feature applies to all Linux-based Communication Manager servers that have more than two Ethernet Network Interfaces (NICs).

A customer can assign the following functions to any server NIC:

- Enterprise LAN
- Duplication LAN (duplicated servers only)
- Control Network A
- Control Network B (duplicated control networks only)
- Services Laptop

This feature eliminates all reference to, and the distinction between, S87XX fiber-PNC and IP-PNC. Eliminated, too, are all restrictions and rules on Ethernet port assignment except that no other functions may be assigned to the NIC used for the Services Laptop.

The duplication LAN must use a gigabit NIC. Avaya highly recommends that no other functions be assigned to the NIC used for duplication.

IP-PNC/GW can only initiate DPT calls to the fiber-PNC fragment that contains the trunks that are associated with the IGAR LDN.

Name display on unsupervised transfer

The name of a transferred-to station appears on the display of the initiating station. Here is how this feature works:

Station A calls station B. The call to station B is then transferred to station C. Station B has established a connection through QSIG to station A, and an "enquire" connection (intern or through QSIG) to station C.

Station B transfers station C to station A. The name of station C (not answered enquire call) is sent after call transfer to station A when the **Send Name** field on the **Trunk Group** screen is set to **y** at the QSIG trunk side to the primary PINX. The name of the transferred-to station C appears on the display of the station A in the primary connection immediately after the call transfer.

QSIG Call Completion - Administrable TSC signaling connection

This capability covers auto callback within a private corporate network only through QSIG. Communication Manager 4.0 provides an administrable option on the **Trunk Group** screen to allow users to specify the method of signaling connection that the system uses while waiting for a busy station to become idle.

The **Retain NCA-TSC for Auto Callback?** field on the **QSIG Trunk Group Options** page of the **Trunk Group** screen allows customers to administer whether or not to retain the Temporary Signaling Connection (TSC) when QSIG Call Completion (QSIG-CC) is activated with non-Communication Manager servers (like the I55 or Siemens Hicom).

Older QSIG-CC standards require Path Reservation, which is not supported by Communication Manager. QSIG-CC can fail if the QSIG TSC is dropped when connected to a server that does not support the latest version of the QSIG-CC standards. In order for Communication Manager to work with servers based on old QSIG-CC standards, this retaining option is needed.

Currently, the preferred QSIG-CC signaling connection method used by Communication Manager is the signaling connection release. In other words, Communication Manager populates the element retain-sig-connection in CCBS/CCNR invoke APDUs with the hard-coded value FALSE.

Starting with Communication Manager 4.0, the new administration option on the **Trunk Group** screen controls the:

- Element retain-sig-connection in CCBS/CCNR invoke APDUs, when Communication Manager is the originating PINX.
- Signaling connection method for the QSIG-TSC when Communication Manager is the terminating or the outgoing gateway PINX.

For more information, see the Screen Reference chapter of the *Administrator Guide for Avaya Communication Manager*.

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. This is done by transiting the charge information from a gateway enterprise system to the end user's enterprise system and desktop display.

This support is for both the AOC-D (charging information during the call), and the AOC-E (charging information at the end of the call).

SRTP media encryption

SRTP is a media encryption standard. An end-to-end SRTP implementation includes the following media processing platforms:

- SIP telephones (SRTP video encryption for SIP is not covered)
- TN2602AP circuit pack
- MM760 media module in a G700 Media Gateway
- G700 VoIP on-board element
- G450 VoIP element
- G350 VoIP element
- G250 VoIP element

Note:

All of these platforms, except for SIP telephones, also support AES and AEA media encryption. For SIP telephones, only the 9620 and the 9630 support SRTP.

SRTP is supported on H.323 endpoints, including the Avaya one-X Deskphone Edition IP telephones, including:

- 9610
- 9620
- 9630
- 9640
- 9650

Communication Manager release 4.0 or later is required to support three SBM-24 button modules on the 9600- series telephone, up to 72 additional call appearances.

Standard Local Survivability

Standard Local Survivability (SLS) provides a local Avaya G250/G350 Media Gateway and Juniper J4350/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.

SLS provides:

- Calling capability among analog, DCP, and IP telephones
- ISDN BRI/PRI trunk interfaces supported on the G250-DS1, G250-BRI, G350, and Juniper J4350/J6350 gateways

- Non-ISDN digital DS1 trunk interfaces supported on the G250-DS1, G350 and Juniper J4350/J6350 gateways
- Outbound dialing through the local PSTN (local trunk gateway) from analog, DCP, and IP telephones
- Inbound calls from each trunk to pre-configured local analog or IP telephones that have registered
- Direct Inward Dialing (SLS)
- Multiple call appearances
- Hold (SLS) and Call Transfer (SLS) functions
- Contact closure feature
- Local call progress tones (dial tone, busy, etc.)
- Emergency Transfer in survivable mode on the media gateway hardware in cases of power loss
- Auto Fallback to Primary Server
- IP station registration
- Expanded dial extension numbering for a maximum of 13 digits

Station lock by time of day

Beginning with Communication Manager 4.0, you can now lock stations using a Time of Day (TOD) schedule. To engage the TOD station lock/unlock feature, you do not have to dial the station lock/unlock Feature Access Code (FAC), or use the **stn-lock** button push.

For a complete description of Station Lock by time of day, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Survivable CDR

Use the Survivable CDR feature to preserve the CDR records associated with calls that occur while a gateway is under the control of a Local Survivable Processor (LSP). When administered with the Survivable CDR feature enabled, the LSP saves the CDR information in files that are stored in a special directory on the local hard drive until such time as the CDR adjunct logs into the LSP and copies the files on to its own storage devices.

The LSP has the capacity to store up to a total of 20 CDR data files. Once the adjunct has read the data files, it can delete them. When the LSP attempts to create the 21st CDR data file, the oldest CDR data file on the LSP disk is automatically removed.

CAUTION:

Cautions associated with using the CM 4.0 Survivable CDR feature:

- The special login used by the CDR adjunct will have the same rights and permissions as Customer logins.
- Actions performed by the CDR adjunct through SFTP (Secure File Transfer Protocol) will not be logged.

For a complete description of Survivable CDR, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Team button

The team button feature is a standard feature starting with Communication Manager 4.0 and enables you to monitor members of a team of stations. Members of a team can be any station type with multiple call appearance displays and administrable feature buttons. Attendant consoles, remote stations, BRI-stations and SIP-Phones are excluded as valid stations, either as monitoring station or as monitored station.

For a complete description of the team button feature, see the following documents:

- *Feature Description and Implementation for Avaya Communication Manager*
- *Administrator Guide for Avaya Communication Manager*

Chapter 2: Hardware

This chapter presents highlights of any hardware as part of the most current releases of Avaya Communication Manager running on Avaya DEFINITY® servers, as well as the Avaya S8000 series Media Servers (with associated Avaya Media Gateways).

Release 4.0 hardware additions

Avaya Communication Manager release 4.0 includes the following general hardware additions.

Avaya one-X Deskphone Edition IP telephones

With its wideband audio capability in combination with noise reduction, the 96xx-series telephones provide high fidelity with crystal clear sound. These telephones, when natively administered in Communication Manager, support 3 SBM24 and 96 buttons in Unicode.

9610 IP telephone

The Avaya 9610 IP telephone is specifically designed for use in common areas such as building lobbies or hallways. In most instances, the 9610 telephone will not belong to a single user, but rather will be shared by visitors or building employees. Through its integrated web browser and backlit display, the 9610 supports productivity-enhancing phone applications such as access to building directories, visitor information, news, and events.

For more information on features, see:

- *Hardware Description and Reference for Avaya Communication Manager*
- Avaya one-X™ Deskphone Edition for 9610 IP Telephone User Guide

9620 IP telephone

The Avaya 9620 IP telephone, released in July 2006, is specifically designed for the everyday telephone user — those who rely on multiple communications tools such as email and instant messaging, yet still require a high-quality and intuitive telephone for voice communications. The high fidelity audio of the 9620 IP telephone provides crystal clear sound, eliminating background noise. The backlit display and intuitive interface simplifies access to critical telephone features, such as one-touch dialing from the contact list and accessing recent call information from call logs.

Through its integrated web browser and application interface, the 9620 supports productivity-enhancing phone applications such as corporate LDAP directories and the receipt of broadcast reminders and alerts.

For more information on features, see:

- *Hardware Description and Reference for Avaya Communication Manager*
- Avaya one-X™ Deskphone Edition for 9620 IP Telephone User Guide

9630 IP telephone

The Avaya 9630 IP telephone, released in July 2006, delivers advanced communications capabilities in a solution designed for those who are absolutely dependent on voice communications for their business operations.

The 9630 IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The backlit display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants.

Through its integrated web browser and application interface, the 9630 supports productivity enhancing phone applications such as LDAP corporate directories and integration with Microsoft Outlook calendars.

For more information on features, see:

- *Hardware Description and Reference for Avaya Communication Manager*
- Avaya one-X™ Deskphone Edition for 9630 IP Telephone User Guide

9640 IP telephone

The Avaya 9640 IP telephone delivers advanced communications capabilities in a solution designed for those who are absolutely dependent on voice communications for their business operations.

The 9640 IP telephone supports a high-resolution color display with integrated web browser and application interface. It is the ideal telephone to support productivity enhancing phone applications such as LDAP corporate directories, integration with Microsoft Outlook calendars, and surveillance cameras/web cams (refreshed still images).

The 9640 IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The color display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants.

For more information on features, see:

- *Hardware Description and Reference for Avaya Communication Manager*
- Avaya one-X™ Deskphone Edition for 9640 IP Telephone User Guide

9650 IP telephone

The Avaya 9650 IP telephone delivers advanced communications capabilities in a solution designed specifically for those who are employed to speak on the telephone for the majority of the day — positions such as building receptionists and executive administrative assistants.

The 9650 IP telephone supports higher quality wideband audio in both the handset and speakerphone, which provides crystal clear audio with the elimination of background noise. The backlit display and intuitive interface simplifies access to Avaya Communication features, such as simultaneously managing multiple calls and selectively muting and dropping conference call participants. The 9650 supports built-in button module functionality (16 buttons) with the user interface enhanced to provide simple one-touch access to bridged appearances, speed dials and feature keys.

Through its integrated web browser and application interface, the 9650 supports productivity enhancing phone applications such as LDAP corporate directories and integration with Microsoft Outlook calendars.

For more information on features, see:

- *Hardware Description and Reference for Avaya Communication Manager*
- Avaya one-X™ Deskphone Edition for 9650 IP Telephone User Guide

DAL 2 duplication memory card

The DAL2 is a duplication memory card that is new with Communication Manager 4.0. The card is available only on S8720 Media Server systems. The DAL2 card has 512 MB of memory, compared to 256 MB of memory on the DAL1 duplication memory card. The extra memory is needed to accommodate the increase in shadowed memory size for S8720 systems which have the S8720 in XL Configuration and hardware duplication.

The DAL2 cards are used in pairs, one in the Active server and another in the Standby server. The DAL2 operates in two modes, Active mode and Standby mode. The DAL2 supports one-way memory shadowing. Critical data is transferred from one DAL2 to another via a fiber link. When a processor writes into the memory of the DAL2 in the active server, the write is also sent over the fiber link to the other DAL2 card in the standby server. The DAL2 is field replaceable.

For more information, see:

- *ESS User Guide*
- *Upgrading, Migrating and Converting Media Servers and Gateways*

G250 Media Gateway

The Avaya G250 Media Gateway supports the connection of PCs, LAN switches, IP telephones, analog telephones, and trunks, via fixed analog and PoE ports on the chassis. A media module slot supports either of two WAN media modules, for connection to a WAN. There are several models of the G250, with various port combinations for support of analog, BRI, or T1/E1 trunks or DCP telephones, as described below.

The G250 is available in the following models:

- Analog model (G250-Analog). The G250-Analog includes four analog trunk ports, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.
- BRI model (G250-BRI). The G250-BRI includes two ISDN BRI trunk ports, one analog trunk port, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.
- DCP model (G250-DCP). The G250-DCP provides twelve DCP (Digital Communications Protocol) ports, as well as four analog trunk ports, two analog line ports, a Fast Ethernet WAN port, and two LAN ports.
- DS1 model (G250-DS1). The G250-DS1 provides a T1/E1 and a PRI trunk port, enabling support of fractional T1/E1 and PRI. The G250-DS1 also includes one analog trunk port, two analog line ports, a Fast Ethernet WAN port, and eight PoE LAN ports.

G350 Media Gateway

The Avaya G350 Media Gateway is a modular device, adaptable to support different combinations of endpoint devices.

Pluggable media modules provide interfaces for different types of telephones and trunks. A combination is selected to suit the needs of the branch. A LAN media module with PoE standard compliant Ethernet ports provides support for IP telephones as well as all other types of data devices. A range of telephony modules provides full support for legacy equipment such as analog and digital telephones.

G860 Media Gateway

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

For more information on components and configurations, see the *Hardware Description and Reference for Avaya Communication Manager*.

TP-6310 Trunk Media Processing Module

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

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

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

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

IG550 Integrated Gateway

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

The IG550 consists of the TGM550 Telephony Gateway Module (TGM550) and Telephony Interface Modules (TIMs). The IG550 supports the:

- TIM514 Analog Telephony Interface Module
- TIM510 E1/T1 Telephony Interface Module
- TIM521 BRI Telephony Interface Module

The IG550 is inserted into either a Juniper J4350 or J6350 Services Router. The IG550 is also connected over a LAN or WAN to an Avaya media server running Communication Manager. Therefore, Avaya S8700, S8710, S8720, S8500, S8400, and S8300 Media Servers are able to provide the same top quality telephony services to the small branch office as to the headquarters of the organization. As a result, the IG550 provides full feature support for IP and analog telephones.

The IG550 is designed for use in a 20-to-100 user environment, with optimal performance at branch offices with 20 to 80 telephones.

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

In addition to advanced and comprehensive telephony services that are provided by the TGM550, the Juniper J-series Router, either the J4350 or J6350, provides full data networking services, precluding the need for a WAN router. The J-series routers use Juniper Physical Interface Modules (PIMs) for the hardware components to support network and routing

features. The J-series routers also provide Ethernet connections to a separate Ethernet switch that IP phones connect to.

For more information about features of the IG550 Integrated Gateway and features of the J4350/J6350 Services Router, see 03-601548, *Overview of the Avaya IG550 Integrated Gateway*.

TIM510 E1/T1 Telephony Interface Module

The TIM510 T1/E1 Telephony Interface Module terminates a T1 or E1 trunk. The TIM510 has a built-in Channel Service Unit (CSU) so an external CSU is not necessary. The CSU is only used for the T1 circuit.

The TIM510 features:

- DS1 level support for a variety of E1 and T1 trunk types
- Trunk signaling to support US and International CO or tie trunks
- Echo cancellation in either direction

TIM514 Analog Media Module

The TIM514 Analog Telephony Interface Module provides four analog telephone ports and four analog trunk ports.

Note:

The four analog trunk ports, ports 5 through 8, *cannot* be used for analog DID trunks. Instead, the four analog line ports, ports 1 through 4, must be used.

The TIM514 provides the capability to configure any of the four trunk ports as:

- A loop start or a ground start central office trunk with a loop current of 18 to 120 mA
- A two-wire analog Outgoing CAMA E911 trunk, for connectivity to the PSTN. MF signaling is supported for CAMA ports.

The TIM514 provides the capability to configure any of the four line ports as:

- A wink start or an immediate start DID trunk
- Analog tip/ring devices such as single-line telephones with or without LED message waiting indication

The TIM514 also supports:

- Three ringer loads, which is the ringer equivalency number for up to 2,000 feet (610 meters) for all eight ports
- Up to four simultaneously-ringing ports
- Type 1 caller ID and Type 2 caller ID
- Ring voltage generation for a variety of international frequencies and cadences

TIM521 BRI Telephony Interface Module

The TIM521 BRI Telephony Interface Module provides four ports with RJ-45 jacks that can be administered as BRI trunk connections.

The TIM521 supports up to four BRI interfaces to the central office at the ISDN T reference point. Information is communicated over each port in two ways:

- Over two 64-kbps channels, called B1 and B2, that can be circuit-switched simultaneously
- Over a 16-kbps channel, called the D-channel, that is used for signaling. The TIM521 occupies one time slot for D-channel use, regardless of whether one, two, three, or four D channels are in use.

The circuit-switched connections have an A- or Mu-law option for voice operation. The circuit-switched connections operate as 64-kbps clear channels when in the data mode.

Each port interfaces to the central office at the ISDN T reference point.

Note:

The TIM521 module does not support BRI stations or combining both B channels together to form a 128-kbps channel.

S8300 Media Server version C

An S8300 Media Server (version C) is an Intel Celeron-based processor that runs the Linux operating system. The S8300 Media Server resides in Slot V1 of a G700 Media Gateway and includes:

- A 60-GB hard disk
- 1 GB RAM (with one 1 GB DIMM)
- Three USB ports and a 10/100 Base-T port
  - One USB port supports a readable DVD/CD-ROM drive, which is used for system installations and upgrades.
  - One USB port can be used for a USB modem.
  - Another USB port can be used for a Compact Flash drive.
- One services port.
- One internal Compact Flash drive, which is used as the primary reboot device and provides additional reliability with the RAMdisk and hard drive

For more information, see the *Hardware Description and Reference for Avaya Communication Manager*.

S8720 Media Server

The S8720 media server is available in two configurations:

- Standard configuration
- Extra large configuration, available with Communication Manager 4.0 and later releases. This configuration requires the DAL2 card.

The capacity increases are as follows:

- Trunks from 8000 to 12,000
- Port network VAL circuit packs from 10 to 128
- Announcement files from 256 to 1024 per PN VAL
- Tone Detector Support from 1200 to 8000
- Announcements per Communication Manager from 3000 to 9000
- Agents logged-in to a skill
- Logged-in agents to 7000
- Logged-in agent-skill pairs to 100,000
- VDN variables to 9
- Attendant consoles
- EC500 Enhanced CLID Support

For more information, see the *Hardware Description and Reference for Avaya Communication Manager*.

Samsung Ubigate iBG-3026 router

The Samsung Ubigate iBG-3026 Router in a branch office works with Avaya Communication Manager release 4.0 and later, and Avaya SIP Enablement Services (SES) as part of an end-to-end SIP solution. Avaya SIP phones can be connected to the router at the branch, and connectivity is provided to Avaya Communication Manager through a SES server at the main office. Only the combination SES home server is supported.

Supported configurations include the following media servers:

- S8710
- S8720
- S8500

A G650 media gateway is required.

The Samsung Ubigate iBG-3026 Router is currently only available from Avaya in China, Germany, India, and Korea.

For information on the Samsung Ubigate iBG-3026 Router, see the Samsung documentation, available at http://www.samsungen.com.

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