# Avaya Aura® Communication Manager 6.0.1 SP9

Release Notes

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## Contents

# Changes delivered to Communication Manager 6.0.1 SP9

## Communication Manager 6.0.1 SP9 Release Notes

Communication Manager service packs and releases are cumulative, and Communication Manager 6.0.1 SP9 includes the changes delivered to Communication Manager 6.0.1 SP0 and SP0.01, SP1 and SP1.01, SP2, SP3, SP4, SP5 and SP5.01, SP6, SP7, SP8 and SP8.01, and SP9. These changes are grouped as follows:

- Table 22: Fixes delivered to Communication Manager 6.0.1 SP9 on page 102
- Table 23: Known problems in Communication Manager 6.0.1 SP9 on page 109

For the supported upgrade paths between Communication Manager releases and service packs, see the latest Communication Manager Software & Firmware Compatibility Matrix at http://support.avaya.com. The supported upgrade paths account for both Communication Manager internal data translation records as well as 100% inclusion of bug fixes.

For security purposes, Avaya recommends changing Communication Manager account passwords at regular intervals, staying current on the latest available Communication Manager Service Pack, and reinstalling Authentication Files periodically to change the local craft password.

# Product Support Notices

Some problems are documented as Product Support Notices (PSN). To read the PSN descriptions online:

1. Go to http://support.avaya.com.
2. Enter your **Username** and **Password** and click LOG IN.
3. Begin to type **Communication Manager** in the **Get Started..** box toward the bottom of the page and when Avaya Aura® Communication Manager appears as a selection below, select it.
4. Select **6.0.x** from the **Choose Release** pull-down menu to the right, and the Communication Manager 6.0.x page is displayed.
5. Click **View All** under **NOTICES & RELEASE NOTES**. A list of available documents and a content filter are displayed.
6. Select **Product Support Notices** in the **Content Type** filter. Deselect any undesired content choices. The selected document types are automatically displayed.
7. Click the PSN title links of interest to open the notices for viewing.

# Communication Manager Messaging

For information regarding Communication Manager Messaging Service Packs (RFUs):

1. Go to http://support.avaya.com.
2. Enter your **Username** and **Password** and **LOG IN**.
3. Click **DOWNLOADS & DOCUMENTS** at the top of the page.

4. Begin to type **messaging** into the **Enter Your Product Here** box and when Avaya Aura® Communication Manager Messaging appears as a selection below, select it.
5. Select **6.0.x** from the **Choose Release** pull-down menu to the right.
6. Click **View Downloads**, if required.
7. Available downloads for Communication Manager Messaging are displayed. Click the links to see the details.

# Communication Manager Software

Communication Manager software includes certain third party and open source software packages, including software developed by the Apache Software Foundation http://www.apache.org. Communication Manager 6.0.1 includes open source licenses on the software DVD. To view the license files,

1. Insert the Avaya Aura® 6.0.1 Communication Manager Solution Templates DVD into the CD/DVD drive of a personal computer.
2. Browse the DVD content to find and open the folder **D:\Licenses**.
3. Within this folder are subfolders for Communication Manager, Installation Wizard, SAL-Gateway, Session Manager, and Utility Server that contain the license text files for each application.
4. Right click the license text file of interest and select Open With => WordPad. This information is only accessible on the Communication Manager software DVD and is not installed or viewable on the Communication Manager Server.

# Avaya Aura® Session Manager

For information regarding Session Manager updates:

1. Go to http://support.avaya.com.
2. Enter your **Username** and **Password** and **LOG IN**.
3. Click **DOWNLOADS & DOCUMENTS** at the top of the page.
4. Begin to type **session** into the **Enter Your Product Here** box and when Avaya Aura® Session Manager appears as a selection below, select it.
5. Select **6.1.x** from the **Choose Release** pull-down menu to the right.
6. Click **View downloads**, if required.

7. Available downloads for Session Manager are displayed. Click the links to see details.

## Avaya Video Conferencing Solutions

Communication Manager 6.0.1 SP8 supports the following video conferencing products that are part of the Avaya Video Conferencing Solutions 6.1 suite.

- Avaya 9600 Series IP Deskphones
- Avaya A175 Desktop Video Device
- Avaya Aura® Conferencing 6.0
- Avaya 1000 Series Video Conferencing Systems
- Avaya One-X® Communicator
- Polycom® HDX
- Polycom® RMX

For software and firmware compatibility information, refer to PSN003614u at http://support.avaya.com.

## System Platform

Communication Manager 6.0.x Service Packs are tested with specific versions and updates of System Platform 6.0. For more information, see Communication Manager Software & Firmware Compatibility Matrix at http://support.avaya.com or the appropriate Communication Manager Product Correction Notices.

# Enhancements in Communication Manager 6.0.1 SP1

New features and significant enhancements in Communication Manager 6.0.1 are described in the document titled Avaya Aura® Communication Manager Change Description for Release 6.0.1 which can be found at http://support.avaya.com.

The following enhancements were delivered to Communication Manager 6.0.1 SP1.

**Table 1: Enhancements in Communication Manager 6.0.1 SP1**

| Enhancement | Keywords | Workaround |
|---|---|---|
| New options for the logfilter command:<br>1. "-s" displays debug level and type for a process or "all".<br>2. "-d" sets the debug level and type to the original default for a process or all.<br>A new message that shows up when the debug level and/or type is changed, has also been added. The message prompts the user to continue by saying **Y** or **N** after it prints a message reminding the user to backup any logs they want to keep. | 083745 | |
| Adding cause values for **hold**, **call initiated**, and **call originated** messages to designate if these events occurred due to a transfer or conference button push. | 101924 | |
| Communication Manager now supports Annex H encryption for H.323 attendants. | 102617 | |
| Communication Manager now sends the octet 3a in the connected party number for CONNECT message. | 102899 | |
| Overlap signalling over QSIG is now possible when SA8887 **Hotline for IP Telephones** is enabled and the hotline feature is used to make calls. | 102927 | |
| SA8146 has been updated and now operates over QSIG trunks and displays non-english specific characters for forwarded calls. | 102971 | |
| The new MM721 BRI Media Module is supported with Communication Manager 6.0.1 and G430/450 Media Gateways. The minimum software and firmware versions required are Communication Manager 6.0.1 SP1 (18777) and MGP Firmware Vintage 31.18.1. | 102974,<br>110174. | |
| | | ***1 of 2*** |

**Table 1: Enhancements in Communication Manager 6.0.1 SP1**

| Enhancement | Keywords | Workaround |
|---|---|---|
| With some gateways, customers could not take full advantage of the number of channels that could terminate at the gateway. This was due to an administration restriction placed on H.323 signaling groups. The restriction prevented a second H.323 signaling group from being assigned to the same Near-end IP Address/Far-end IP Address/Far-end Port combination. This restriction is now removed. | 103064 | |
| Some far-end SIP applications such as Meeting Exchange do not supply a disconnect indication when the application is complete. In Communication Manager, SIP trunk groups are defaulted to expect the disconnect indication to come from the far-end. When this does not happen, the SIP trunks may not disconnect and will not be idled even though the trunk is no longer in use. Therefore, the Disconnect Supervision **In** and **Out** fields will now be administrable for SIP trunk groups. This will allow the disconnect process to be initiated by Communication Manager when necessary. | 103267 | |
| | | ***2 of 2*** |

# Enhancements in Communication Manager 6.0.1 SP2

**Table 2: Enhancements in Communication Manager 6.0.1 SP2**

| Enhancement | Keywords | Workaround |
|---|---|---|
| When a call arrives at a member of a pickup group, other pickup group members will not see the caller's identity when the **Caller ANI during pickup alert** field on the Class of Restriction form is set to N. | 103020 | |
| New GRIP 4359 - Service Observing Next Call Listen Only. The new feature supports following functionality with listen only mode.<br>1.If station/agent is already on call, and Service Observer dials the new FAC to monitor the station/agent then, Service Observer should not bridge-on to the active call. Instead Service observer is set to observe the next call.<br>2.There should not be any tone while service observer bridge-on to the call. | 103023 | |
| New Feature to add SA9118 - International QSIG Identification Numbers on 9th page of the form.system-parameters special-applications. When SA9118 is enabled, Communication Manager shall encode identification numbers as E.164/international when it generates a number using the Public numbering form. This shall apply only to ISDN (PRI, BRI, or H.323) trunk groups with these settings: Supplementary Service Protocol = b (QSIG), Service Type = tie, Numbering Format (page 3) = public. | 103246<br>103541 | |
| The Message Waiting Indication lamp did not light for a Siemens endpoint over QSIG with a voice mailbox on a SIP Integrated Modular Messaging System. | 110003 | |

# Enhancements in Communication Manager 6.0.1 SP3

**Table 3: Enhancements in Communication Manager 6.0.1 SP3**

| Enhancement | Keywords | Workaround |
|---|---|---|
| This enhancement improves the performance of the **`reset ip-stations`** command. | 102436 | |
| logmst does not show the patch activated on the MST trace. | 102551 | |
| A new special application SA9114 is implemented that will support different international access codes. The following changes are made to the locations administration form:<br>1. The Rule column is renamed DST.<br>2. The NPA column is renamed City/Area and will allow up to 5 digits if SA9114 is enabled.<br>3. A new field **Nth Am** (abbreviation for North America) is added. The values in this field will be Y(es) or N(o) with the default value being Y(es). If this field has y(es) then the number in the City/Area field is treated as a North American area code. | 102893,<br>102894,<br>110197,<br>110302. | |
| | | |

# Enhancements in Communication Manager 6.0.1 SP4

**Table 4: Enhancements in Communication Manager 6.0.1 SP4**

| Enhancement | Keywords | Workaround |
|---|---|---|
| This fix reduces the possibility of the var partition being filled by errant applications posting excessive messages on the logs. | 102142 | |
| SA9111 feature allows a blind/attended conference to be made before sending CONNECT message to the calling trunk party. Earlier with (SA8434) - Delay PSTN Connect on Agent Answer blind conference was not allowed for the agent who answered the incoming call over a trunk. This is now allowed with SA9111. SA911 can be administered as below.<br>Set the following fields on system-parameters special-applications to Y:<br>1. (SA9111) - Allow Conference with SA8434?<br>2. (SA8434) - Delay PSTN Connect on Agent Answer?<br>Turn on the "Delay PSTN Connect Message on Agent Answer" field on VDN screen. All the restrictions of SA8434 apply to this feature too. | 111011 | |
| An ESS is now capable of being a server which will trigger the Split Registration Prevention Feature. This feature previously allowed only LSP. | 110317 | |
| This is a new release of Message Tracer Analyzer. This Message Tracer release (6.4.2.10) has support for Interpretation of following:<br>1. New Internal Call Process fields<br>2. New Call Record Dump fields<br>3. New Denial Events added into Avaya Aura Communication Manager<br>4. Upgrade of Avaya SIP stack | 111395 | |
| | | |

# Enhancements in Communication Manager 6.0.1 SP5

**Table 5: Enhancements in Communication Manager 6.0.1 SP5**

| Enhancement | Keywords | Workaround |
|---|---|---|
| Communication Manager supports parsing of BFCP (Binary Floor Control Protocol) attributes in SIP. | 103206 | |
| Communication Manager will tandem the BFCP attribute that comes from one SIP endpoint to another. | 103315 | |
| Communication Manager supports SIP controlled transfers with ASAI. | 110006 | |
| Communication Manager supports BFCP cap neg tandeming. | 110066 | |
| BFCP parameters were missing in the session refresh INVITE from Communication Manager. | 111128 | |
| This fix reduces the time it takes to replace a CMM or MSG patch via the CDOM Patch Management interface. | 111732 | |
| This is a new version of Message Tracer Analyzer (6.4.3.0) which has parsing support for the new fields added to Communication Manager. | 112104 | |
| | | |

# Enhancements in Communication Manager 6.0.1 SP6

**Table 6: Enhancements in Communication Manager 6.0.1 SP6**

| Enhancement | Keywords | Workaround |
|---|---|---|
| The **`list trace station`** command showed incomplete data for Inter Gateway Communication calls. | 082739 | |
| H.323 and DCP endpoints could not be configured to operate like the CALLMASTER sets that were previously used for Call Center Elite agents. Active calls were dropped when the headset, speakerphone, and handset were turned off. With a new field on page 12 of the system-parameters features screen, customers can choose to use the CALLMASTER functionality with EAS Auto-Answer agents who log in on H.323 and DCP phones. | 111231 | |
| Customer could not administer a automatic message waiting (aut-msg-wt) button on a 96XX SIP station. | 112207 | |
| This is a new MTA version 6.4.3.3. There are some fields added into Internal Call Process message(VEDIOFLAG_UPDATE) and denail event message (DNY_NO_MULTILOC and DNY_BAD_SOLOC). This version of Message Tracer has an ability to parse these new fields. | 112480 | |
| | | |

# Enhancements in Communication Manager 6.0.1 SP8

**Table 7: Enhancements in Communication Manager 6.0.1 SP8**

| Enhancement | Keywords | Workaround |
|---|---|---|
| This modification affects the algorithms for selection of audio media processing resources. Specifically, when a network region contains both TN2302/2602 and H.248 media GWs, the system no longer selects the TN2302/2602 resources to the complete exclusion of the H.248 media gateways. Now selection is from both classes of resources according to the relative presence of each. If there are TN2302/2602 resources and H248 resources in the ratio of 10:1, then resources will be allocated in approximately the same ratio. The second significant change is that resources located in regions which are indirectly connected to the region of the requesting endpoint are no longer all grouped together in terms of preference. A resource in a closer network region is preferred over a resource in a further network region. | 112923 | |
| This is new Message Tracer release 6.4.4.1. We have added support for new Internal Call Process fields, Call Record Dump fields and one denial event. | 121004 | |
| | | |

# Enhancements in Communication Manager 6.0.1 SP9

**Table 8: Enhancements in Communication Manager 6.0.1 SP9**

| Enhancement | Keywords | Workaround |
|---|---|---|
| The upper limit of valid user IDs for the Communication Manager SMI was increased from 65535 to 2000000000. | 120997 | |
| This is a new MTA release 6.4.4.3. This release of MTA includes parsing support for following:<br>1. Language 23 CMS messages<br>2. Mute message<br>3. Agent ID message<br>4. New Capro fields, CRD fields and Denial Events<br>5. Avaya SIP Stack Upgrade<br>The decoding of above changes is not supported by earlier Message Tracer release. | 121888 | |
| | | |

# Problems fixed in Communication Manager 6.0.1 SP0

This release includes the following fixes delivered to Communication Manager 6.0.1 SP0.

**Table 9: Fixes delivered to Communication Manager 6.0.1 SP0**

| Problem | Keywords | Workaround |
|---|---|---|
| Fixes associated with the following keywords were also corrected in Communication Manager 6.0.1 SP0. | 102303, 103014, 103040, 103104. | |
| Under extreme traffic conditions, Communication Manager did not provide service from Branch Media Gateways for up to several minutes until system maintenance audited the Communication Manager internal data. | 102571 | |
| IP stations did not un-register after a network outage to migrate to a LSP or ESS when dial plan transparency was involved. | 102610 | |
| A SIP REFER message including a **rfrid= in the Refer-to** header was not forwarded in the corresponding INVITE. This resulted in failed calls. | 102639 | |
| **`List trace station`** and **`tac`** commands failed to record vector events when the traced call involved vectoring. | 102659 | |
| H.323 Stations on a Communication Manager Evolution Server with **H.323 Station Outgoing Direct Media** set to Y in the SIP signaling group form that called a SIP station on a Communication Manager Feature Server with a bridged appearance on a third SIP station got nontalk path. | 102677 | |
| Video call between Polycom HDX failed. | 102808 | |
| A SIP station bridging on (answer) to a Sip Direct Media established call resulted in the parties not hearing each other. | 102890 | |
| Call dropped when there was a glare condition with SIP. | 102933 | |
| High video call traffic resulted in server resets. | 102975 | |
| For Avaya 1050, hosting an eight party conference, three parties were dropped at session refresh time. | 102976, 103037. | |
| Initializing some internal variables helped avert possible H.323 call failures. | 103006 | |
| | | ***1 of 2*** |

**Table 9: Fixes delivered to Communication Manager 6.0.1 SP0**

| Problem | Keywords | Workaround |
|---|---|---|
| Under certain circumstances when calls were made over H.323 trunk, the Communication Manager went into rolling reboots. | 103007 | |
| There was no audible connection. When an IP station on an Evolution Server called a SIP 1X-Communicator which was in telecommuter mode and the call was answered by a SIP telecommuter station. | 103011 | |
| The direct IP-IP calls over SIP or H.323 trunk were dropped when Enterprise Survivable Server (ESS) tried to fall back to the main server. | 103016 | |
| A Communication Manager system was not able to provide VoIP resources from Branch media gateways when those media gateways lose communication with Communication Manager and then re-establish communication with Communication Manager. The loss of communication was typically one minute or longer. | 103024 | |
| Security Service Packs will be Cold and that will be displayed on the web screens. | 103058 | |
| High Video Call Traffic Resulting in Resets. | 103146 | |
| | | ***2 of 2*** |

# Problems fixed in Communication Manager 6.0.1 SP0.01

This release includes the following fixes delivered to Communication Manager 6.0.1 SP0.01

**Table 10: Fixes delivered to Communication Manager 6.0 SP0.01**

| Problem | Keywords | Workaround |
|---|---|---|
| A SIP station on a Communication Manager1, with Initial IP-IP Direct Media set to N, and Direct IP-IP Audio Connections set to Y, in the assigned SIP signaling group, called a SIP station on a second Communication Manager with Initial IP-IP Direct Media set to Y, in the assigned SIP signaling group via SIP trunk / Session Manager. That called SIP station had an EC500 extension pointing to any other station on the first Communication Manager routed via SIP trunk / Session Manager and EC500 was enabled. The EC500 extension answered the call, and after the call was established the principal SIP station bridged on the existing call. This resulted in the conference failing to establish. | 102794 | |
| On a Communication Manager system, completion of brief tones (ZIP tone, for example) was not reported properly to the call processing subsystem when the tones were generated on H.248 media gateways. Certain features rely on the tone completion event to stimulate further call progress. Agent/Caller Disconnect is one such feature. In this feature example, without the tone completion event, the agent and incoming caller did not disconnected properly in some circumstances. | 103029 | |
| A video call between 10x0 video endpoints was downgraded to audio call when there was not enough bandwidth to make it a video call. Downgraded audio call was dropped after a session refresh. | 103135 | |
| Fix 1-way talk path problem between 96xx end-points which had the DiffSevr Code Point (DSCP) capability. | 103290 | |
| | | |

# Problems fixed in Communication Manager 6.0.1 SP1

The following fixes were delivered to Communication Manager 6.0.1 SP1.

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP1. | 093660, 102970, 102974, 102903, 103027, 103036, 103177, 103183, 103299, 103054, 103061, 103131, 103132, 103181, 103275, 103419, 110168. | |
| Customer did not see the button labels on the 2420 DCP terminals get updated immediately after the Personal Station Access (PSA) associate. | 093844 | |
| Incorrect CDRs were produced for some scenarios related to path replacement with QSIG or REFER with SIP. | 101102 | |
| Zero-sized SSH keys no longer prevent the correct startup of sshd. | 101331 | |
| There was no name or calling number displayed when a call was made using a bridged appearance via QSIG trunk. | 101410 | |
| A barge-in announcement was stuck in the PLAYING state. | 101532 | |
| Occasionally, there was no talk path and calling party heard dead air when a call was made over an IP trunk. | 101709, 102615. | |
| | | ***1 of 11*** |

## Table 11: Fixes delivered to Communication Manager 6.0.1 SP1

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, SIP endpoint subscription failed when Communication Manager had multiple TLS signaling groups to SES or Session Manager and some of them went out of service. | 101736 | |
| When Communication Manager sent offer in reinvite or in response(200 OK) it did not send its direction attribute as a=sendrecv even when the far end had gone on hold before. In certain call scenarios this could cause temporary loss of talk path. | 101781,<br>102344. | |
| Occasionally, custom labels were not displayed on the stations form when viewed from the System Access Terminal (SAT). | 102033 | |
| A call made by a station that supported Wideband over an H.323 trunk did not shuffle when wideband codec was administered.This behavior has been modified and calls get shuffled now. | 102238 | |
| A call was transferred to an attendant busy on another call. The call then went to another attendant in the group. The display on the second attendant would show "UNKNOWN NAME"instead of the party to which the call was transferred. | 102244 | |
| For the Sync Over IP feature, Media Gateways that were out of service due to a disabled network region showed a failure reason of **No Sync** rather than **NR Dis**. | 102322 | |
| Softkeys on a 96xx H.323 station were not updated when it cancelled the call transfer before the called party answered. This happened only when the softkeys for idle and ringback features on the 46xxsettings file were enabled. | 102333 | |
| On-hook dialing failed on DCP set after a no-hold-conf call was cancelled. | 102364 | |
| The caller was dropped and no busy treatment was given when a busy station on I55 was called from Communication Manager. | 102370 | |
| In case of far end hold for NCR and TDM scenarios, no talk path was observed when the far end changed keys or other media attributes. | 102438 | |
| | | ***2 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| On a system with trunk to trunk transfer set to restricted, and a COS that had trunk to trunk transfer override activated, a trunk to trunk transfer call was dropped when a party that was added by ASAI Single Step Conference (SSC) was dropped. There was an inbound trunk call answered by station A with a COS that had trunk to trunk transfer override activated. A second party B with the same COS was added to the call using the ASAI SSC feature. The call was then transferred to far end destination via an outbound trunk. The trunk to trunk transfer was dropped when station B was dropped. With this change the trunk to trunk transfer call remains up after station B is dropped. | 102487 | |
| The command **`list measurements ip voice-stats`** generated an error when Port Network 1 (PN1) was not assigned. | 102502 | |
| An incorrect number was displayed on the EC500 extension for an auto callback call. | 102516 | |
| Communication Manager closed the base ip station's TCP socket and the base station rebooted when one-x C portal phone tried to register in shared control mode through the Application Enablement Services (AES) link. | 102521 | |
| Ports that were administered for use as paging equipment did not play **pause** digits correctly causing calls to fail. | 102558 | |
| On a Communication Manager system, DTMF digits were not heard on the far-end of an IP trunk, including SIP trunks in which the IP trunk used a H.248 Media Gateway VoIP resource. The IP trunk originally used an alternate method for delivering DTMF digits (out-of-band, RFC2833) but changed the delivery method to in-band. When the IP trunk still used the same H.248 MG VoIP resource after changing the DTMF delivery method to in-band, digits were not delivered in-band. | 102559 | |
| list performance trunk-group report displayed a blank for the trunk group size for any trunk-group with a size of 255. | 102561 | |
| The calling party got a missed call log entry with its own number when a call was made to a station which was not in service and the call forwarded to another station. This problem was only reproducible when LOGUNSEEN was enabled on the Phone's 46xxsettings.txt file. | 102569 | |
| Occasionally, Mempool Errors involving ISDN connections showed up in the logs. | 102574 | |
| | | ***3 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, **find-me** calls from Modular Messaging to Communication Manager were auto-answered. | 102585 | |
| The UUIE information that was in the second line of the display was not cleared when a call was transferred to a Call-Master from an agent. Only the first line of the display was cleared. | 102605 | The workaround for this is to enable the special application SA7710 - Enhanced Display on Redirected Calls. |
| CDR was incorrectly created for unanswered SIP to R2MFC tandem trunk calls. | 102616 | |
| A station in another tenant was alerted incorrectly when the initial crisis alert was not responded to in the correct tenant. | 102634 | |
| Changing the field "Call Handling Preference" on the agent form resulted in an error saying "Entry is bad" even though the change was successful. | 102640 | |
| Coverage over multiple remote coverage points failed when look ahead routing was administered on the route pattern and the coverage points were reached over a SIP trunk. | 102738 | |
| The command to get measurement data from a DS1C board in slot 1 of a switch node carrier did not work. Entering **`list measurements ds1-fac log 1e01a`** on SAT caused an error indicating the entered location was bad. | 102741 | |
| Occasionally, invalid data was left in the Incoming Call Handling Table (ICHT) for SIP trunk groups. This invalid data caused incoming SIP trunk calls to fail. | 102784 | |
| The **Service Link Ext** field on the **status station** displayed an incorrect dial string when special characters were used in that dial string. | 102793 | |
| On-hook dialing on the hard-phone did not work after a successful call when 64xx phones were integrated with One-X Communicator in a shared-control mode. | 102800 | The workaround was to go off-hook/ on-hook. |
| | | ***4 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| The Media Gateways page of the status link command did not show registered Media Gateways. | 102805 | |
| Previously, a listen only party added via a Single Step Conference was counted as a non-trunk party when considering trunk to trunk transfer requests.This allowed trunk to trunk transfers that were meant to be restricted. Now, a listen only party will not be counted and the transferring station's COS will determine if the system trunk to trunk restrict administration is honored. | 102811 | |
| Attempts to change the Survivable Processor via the change survivable-processor command failed to display the pages containing that data when a Survivable Processor was administered before its associated IP Service and Processor Channel data. | 102847 | |
| Communication Manager sent an UPDATE message in Allow header of 200 OK in response to REINVITE for a public-network SIP trunk. | 102851 | |
| Generic greeting was heard when a call was made from an Agent to a station that covered to SIP Modular Messaging. | 102855 | |
| Line selection for 9620, 9630, and 9650 terminals changed when another call was on hold after background maintenance was run. This caused changes in the soft feature buttons that were displayed. | 102859 | |
| The caller could activate an Automatic Call Back on hearing the busy tone when a call was made to a station which was not in service, but had an Extension to Cellular (EC500) mapping. | 102860 | |
| Calls on a SIP trunk from Communication Manager to Voice Portal or a SIP station were not tracked by IQ/CMS. | 102861 | |
| ANI (Automatic Number Identification) was not captured in CDR for incoming DTMF calls. | 102864 | |
| Occasionally, a third record displayed in the **`list ip-interface`** all output when only the PROCR ip-interface was administered. This caused the command to scroll without completion. | 102869 | |
| Attendant was able to transfer parties in different tenants which did not have permissions to talk to each other when integrated-music was used. | 102884 | |
| | | ***5 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| Customer could not make changes from SAT to certain 46xx stations because validation for the IP HOT LINE feature blocked the **submit** button of the Station form, placing the cursor on blank space beneath the Abbreviated Dialing List fields and displaying the following error message: **Required abbreviated dialing list not assigned.** | 102885 | |
| The history-info SIP header for the first VDN was lost when a call arrived at a VDN on a SIP trunk and was sent to a second VDN (for instance, by vector processing). This caused problems when the VDN information was subsequently expected by an IVR, display, or other applications. | 102886 | |
| An external call transferred by a SIP user was not re-routed via ASAI call deflection to a remote switch. | 102907 | |
| Using IAS (Idle Appearance Select) FNE (Feature Name Extension), the call-appearance on the principal OPTIM (Off-PBX Telephone Integration and Mobility) station was locked for 30 seconds when an invalid number was dialed from an EC500 phone. This issue occurred only when SA9106 was enabled. | 102937 | |
| When the **Timed After-Call Work Mode for Held Calls** feature was enabled and an agent pressed the **after-call work** button with an active call and then initiates a call transfer, that agent will enter TIMED After Call Work mode, instead of entering After Call Work mode as intended, when the now-held caller dropped before the transfer was completed. | 102954 | |
| Occasionally, the system crashed on a server interchange. | 102994 | |
| An invalid error message was displayed when running "set sync" on a Media gateway. | 102996 | |
| A parked call was dropped when a station was being recorded and had its music source administered as integrated music. | 103003 | |
| Busy Indicator for Trunk Access Code (TAC) to call paging system failed for a station and the attendant. | 103005 | |
| | | ***6 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| An ACD call on an analog type port, including a DS1FD VRU port, was put on conference/transfer hold by flashing the switch hook (real or simulated) and was dropped before a valid extension was dialed on the second leg call. This was reported to IQ/CMS as an ACD call rather than an extension-out call. | 103025 | |
| Only 1 CDR was produced instead of 2 when an incoming trunk call was answered by Voice Portal and transferred to another agent. | 103032 | |
| Agent was occasionally unable to login when the **date-time** button was already activated. | 103035 | |
| When **Initial IP-IP Direct Media** was enabled on a Feature Server or an Evolution Server and a SIP station called another SIP station which had enabled call forwarding to a SIP station outside of this server and the external station did forward the call again there was an audible connection which was cleared after some 30 seconds. | 103051 | |
| An inbound SIP trunk hung when Communication Manager was working as an Evolution Server when the original half call doesn't sent a re-INVITE message to the term half call by any failure. | 103055 | |
| Internal system errors were logged for very high traffic systems. A high enough number of calls could be made to overrun the maximum BW recorded amount of 65 Mbits when a network region pair had no Bandwidth limits. | 103056 | |
| Problems with cron jobs that resulted in messaging email notifications not being sent, scheduled backups not running and log files were not being rotated. | 103070 | |
| Communication Manager system repeatedly selected the same H.248 media gateway for VoIP resources even when the H.248 MG had no more VoIP resources available. | 103085 | |
| An IQ/CMS measured trunk call was abandoned from hold and was reported as abandoned to a Pre-R15 CMS. The reason for a held call being dropped should only be reported to CMS R15 and later. | 103087 | |
| Customers calling into call center agents on ISDN trunks heard the intercept tone when the called agent disconnects before the caller if the new **Agent Caller Disconnect Tones** feature was enabled. | 103101 | |
| | | ***7 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| A duplicated Communication Manager system chose to select VoIP resources from a H.248 Media Gateway after a server interchange but before the media gateway had re-established communication with the newly active duplicated server. | 103115 | |
| An ESS would wait for the Port Network Cold reset timer plus 10 seconds when the last MG left control before forcing the unregistration of the IP phones when a system consisted of both Port networks and Media Gateways. Now the wait is eliminated speeding up recovery if all the port-networks have already returned to the Main server. | 103116 | |
| Communication Manager reset during heavy SIP traffic when the CLAN over which the SIP trunk traffic was run was busied/released and then again busied out. | 103129 | |
| There was no audible connection when the called station answered the call when **Initial IP-IP Direct Media** was enabled on a Feature Server or an Evolution Server and a SIP station called another SIP station which had a call on hold. | 103143 | |
| Six parties were dropped from the conference at session refresh time on an Avaya 1050 video endpoint hosting an eight party conference. | 103144 | |
| Calls queued against an agent with skills 1 and 2 while the agent was in aux mode. Even after the agent went into auto-in mode, the calls were stuck in the queue. | 103156 | |
| IP direct calls over H.323 trunks dropped when resources being controlled on an ESS fell back to the main server. Also, on any server, an H.323 trunk call that was IP direct, dropped when all VoIP media resources were removed or disappeared for the associated region. | 103161 | |
| The system did not select media resources from remote regions evenly in a system where IP endpoints were located in a region with no PN or GW media resources, and PN/GW media resources were distributed across many other network regions. | 103164 | |
| For large configurations, an ESS took several hours to force the IP phones off the system after the last port network and/or Media Gateway returned to the main server. Now that time has been reduced so the unregistration of IP phones happens after the last port network is down and/or Media gateway's link loss delay timer has expired. | 103167 | |
| | | ***8 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| A member of Attendant group blind transferred a call to a Vector Directory Number (VDN), which subsequently queued the call to a hunt-group and when all the agents were busy, the call was returned to the VDN which then routed the call to SIP Modular Messaging, where a wrong greeting (generic) was played instead of VDN's greetings. | 103168 | |
| A LSP sent subscribe messages to MM for inactive stations. | 103169 | |
| The system had a seg fault when using system monitoring tools to run many list measurements commands in sequence. | 103175 | |
| Auto Callback failed when the user who initiated ACB over a QSIG trunk had an entry in the off-pbx station mapping table. | 103202 | |
| Hundreds of failed SIP calls on a system with network problems caused system restart due to the speed at which time available audits run. The time available audit found left up connections that the audit then force a cleanup on. The clean up of hundreds of calls too quickly caused the system restart due to massive interprocess communications. | 103209 | |
| Occasionally, calls on which announcements were played got dropped. | 103218 | |
| Occasionally SIP trunks would not return from the Communication Manager survivable server to the main server leaving the SIP trunks registered with the survivable server. The survivable server could also reset when this happened. | 103222 | |
| Communication Manager was not able to parse generic parameters in a Record Route Header causing inter operation issues with CS1000. | 103223 | |
| Communication Manager went into overload when the branch parameter in a SIP message exceeded 512 characters. | 103245 | |
| Communication Manager returned 503 Service Unavailable when it received a MWI NOTIFY message on a signalling group which had no free channels. This caused MWI failures on the stations associated with the voice mail system. | 103283 | |
| Communication Manager reset when it received any SIP message without User-Agent or Server header. | 103308,<br>103378. | |
| For IP endpoints registered via Processor Ethernet, some extensions could be out of service after a server interchange. | 103320 | |
| | | ***9 of 11*** |

Table 11: Fixes delivered to Communication Manager 6.0.1 SP1

| Problem | Keywords | Workaround |
|---|---|---|
| Incoming SIP trunk calls failed after a busy/release of the SIP trunk group. | 103332 | |
| An Avaya A175 Desktop Video Device did not successfully enter the necessary passcode when dialing into an RMX hosted audio or video conference. | 103336 | |
| The ASAI software version query responded with a release of 0 for Communication Manager 6.0 when it should have responded with a release of 16. | 103384 | |
| **Status Station** command showed the mac address as **unavailable** even when the station had a valid mac address. | 103385 | |
| Resource Priority Header was inserted in the UPDATE message for precedence call transfer. | 103389 | |
| A Communication Manager system took 40 to 60 seconds to detect an H.248 Media Gateway link outage. This resulted in dropped or unusable calls during this detection time. | 110002 | |
| 20-30 seconds after answer, the call dropped when a SIP originated trunk call had a trailing '#' after the called number and the termed to party was a SIP trunk as well. Both SIP signalling groups in this call had been administered with **Initial IP-IP Direct Media** set to Y. | 110008 | |
| The announcement was not played when an internal announcement audit was executed. | 110067 | |
| Station to station calls across a SIP trunk did not have audio when the stations were enabled for video softphone. | 110015 | |
| Communication Manager rebooted when a call between two Session Initiation Protocol (SIP) endpoints was dropped or shuffled. These were registered on two different Communication Managers with same Session Manager. | 110116 | |
| The call dropped when a video call was placed on hold and then resumed. | 110173 | |
| Incorrect CPN was captured in CDR for unformatted format after upgrading Communication Manager to 6.0.1 from 6.0 SP0. | 110226 | |
| | | ***10 of 11*** |

**Table 11: Fixes delivered to Communication Manager 6.0.1 SP1**

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, Communication Manager responded incorrectly to Message Manager which caused message waiting indication to behave erratically. | 110288 | |
| Main Server processor channel data was overwritten by Survivable Processor translations after it rebooted. | 110379 | |
| | | ***11 of 11*** |

# Problems fixed in Communication Manager 6.0.1 SP1.01

This release includes the following fixes delivered to Communication Manager 6.0.1 SP1.01.

**Table 12: Fixes delivered to Communication Manager 6.0.1 SP1.01**

| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager could reset in systems that have TN 2501 "VAL" circuit packs in a port network with H.323 stations/trunks or SIP endpoints in the same network region as the port network. This only happened when there were no TN2302 "Medpro" circuit packs in the port network that were assigned to the same network region as the TN 2501 "VAL" circuit packs and the trunks/endpoints. | 110705 | |
| | | |

# Problems fixed in Communication Manager 6.0.1 SP2

This release includes the following fixes delivered to Communication Manager 6.0.1 SP2

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP2. | 102940, 103200, 110033, 110136, 110148, 110195, 110343, 110427. | |
| The transferring party saw the calling party's information when a call was made using the Send-NN button and the called station did an internal transfer. | 100007 | |
| The call remained active and no one could make another call to the group page when the originator of that group page did not drop the call. | 100345 | |
| The **Run Now** function of a scheduled backup did not work with multiple data sets. | 101490 | |
| No Talk Path was observed when a call was made using mixed direct media settings where the originating SIP trunk had direct media on and the terminating SIP trunk had direct media off. | 101554 | |
| This fix prevents the segmentation fault which occured when an Application Enablement Services (AES) server sent traffic to Communication Manager and there were severe network impairments/errors (the kind of network impairments/errors that also cause multiple port networks and media gateways to go down). | 102133 | |
| With SA9106 enabled, user was able to bridge-on to call at an OPTIM (Off PBX Telephone Integration and Mobility) station when the call was on hold at PVFMC (Private Fixed Mobile Convergence) endpoint. | 102153 | |
| Call transfer did not abort when the Abort Transfer feature was enabled and a user pressed a different call appearance button, while the transfer was in progress. | 102519 | |
| | | ***1 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| It was possible for a customer to administer a media gateway using the pre-populated location 1 even when that location was not administered. This occurred without giving the administrator a warning. | 102612 | |
| Occasionally, announcements did not work on G450/G430 media gateways after a gateway unregistered and did not return until after the link loss delay timer expired. | 102622 | |
| Communication Manager should run glare timer between 0-2s as per spec RFC-3261 when Communication Manager is not the dialog owner of that call and glare is observed. | 102674 | |
| Communication Manager failed to forward a call to voicemail and rerouting failed when an outgoing qsig call was made and far end sent rerouting request for this call with short digit extension of voicemail hunt group, and these hunt group digits were administered as type extension in the calltype analysis. | 102717 | |
| Occasionally, Communication Manager dropped SIP calls when inter working with some 3rd party SIP telephones and when the SIP trunk session refresh interval expired. | 102806 | |
| Automatic Number Identification (ANI) that reaches the transferred-to station was a trunk number instead of calling party info when a SIP trunk transferred a call to a H323 trunk. | 102960 | |
| Special Application 9020 allows periodic test packets to measure round trip delay and packet loss between network regions. Based on these results, the system can deny IP connections between network regions. If IGAR has been configured and enabled between these same network regions, IGAR can be used to establish the inter-region bearer connection using non-IP trunk facilities. If the originator of a call connects to an H.248 gateway, the system will not use IGAR for bearer establishment when SA9020 has indicated poor IP network performance. The call would have been denied. | 102986 | |
| VDN name was not displayed on ACD agent after BSR poll over IP. | 102991 | |
| For SIP signaling groups, when the Layer 3 test was off, and a call failure on the signaling group's trunks occurred, an alarm was generated after a couple of minutes if the far-end did not respond to the ping test. After the far-end condition has cleared, the signaling group went back in service, but the alarm was never retired. | 103000 | Workaround: Turn on the L3 test. It should be on anyway. |
| | | ***2 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| The aux_work button lamp did not go off when the agent went form aux_work mode to auto in mode even though the agent received calls. | 103012 | |
| Contact header was truncated when a call was made to/from a SIP endpoint that used a phone-context parameter such as Nortel endpoints. | 103043 | |
| The attendant on held state received an alert when the Single Step Confernce(listen-only) party was dropped from the call. | 103053 | |
| Initial invite went with PAI containing only the extension and no display name for a call made over DCS trunk followed by SIP trunk. After answer, re- invite message for display update had PAI containing just display name and no extension. | 103063 | |
| LAR(Look Ahead Routing) failed when AAR/ARS FAC was not administered on the feature-access code form and UDP(Uniform Dial Plan) was used to route calls. | 103083 | |
| An IPv6 trunk call did not establish successfully in an environment with a mix of IPv4 and IPv6 capable H.248 controlled gateways. | 103133 | |
| In the following fields on page 4 of the **system-parameters features** screen, the extension field overwrote the last character when either the announcement or the extension was entered.<br>● **Controlled Outward Restriction Intercept Treatment**<br>● **Controlled Termination Restriction (Do Not Disturb)**<br>● **Controlled Station to Station Restriction** | 103148 | |
| Team button to analog phone failed to answer the call when the ring type was set to ring, delayed ring, icom or abbreviated ring. | 103155 | Workaround to this issue is to use bridge appearance feature instead of team button. |
| The loss of communication with a H.248 Media Gateway resulted in the loss of use of SIP trunks that were using VoIP resources on that Media Gateway. | 103166 | |
| | | ***3 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| The **call appr** and display remained active even when the call had dropped when the agent/caller disconnect tone option was enabled and the call dropped (before call disconnect tone could play) from an ACD/DAC call. This occurred only when there was an agent or a caller and no third party on the call. | 103170 | |
| Some calls did not play announcements under heavy traffic conditions. | 103196 | |
| Calls which were landing on logged off Internet Protocol(IP) or Terminal Translation Initialization (TTI) or Personal Station Access(PSA) phones due to call coverage or call forward, did not follow the criteria for Logged off IP/PSA/TTI phones. | 103199 | |
| The alias node name used to reference a duplex-ESS survivable processor was removed from the node-names IP form while still being used by the survivable processor. | 103215 | |
| Calls measured by IQ/CMS were not tracked by IQ/CMS when they were transferred during the Coverage Response Interval (CRI), to stations of Type "virtual" which covered to a hunt-group with Message Center qsig-mwi which were then redirected to an outgoing trunk group. | 103229 | |
| CDR did not capture resource flag of 4 for video calls. | 103238 | |
| Phone with a bridge-appearance of calling party got an invalid display of **f** when the call was forwarded via SIP trunk. | 103240 | |
| The keyword **i-silent** could not be entered in a vector **`wait-time`** command even when it was displayed as a valid option in the help message. | 103253 | |
| The new label displayed on the phone appended several characters from the old label when a customized button label was edited by the end users using the local phone application and the old label had 11 or more characters. | 103254 | |
| No talk path on calls from/to Multitech endpoints. | 103260 | |
| The **Queue** field on the announcement form was allowed to be changed from **b** to **y** when an announcement was played back using the announcement access code. | 103265 | |
| | | ***4 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| Earlier, the `list trace vdn x` command displayed call flow trace data when the specified vdn number was dialed. In this release, the command displays that data even when the specified vdn is routed-to / from another vdn, rather than just dialed directly. | 103271 | |
| Though the conference was ended, user could still see "conference" on the phone display. This happened when station A called station B over a SIP trunk and station B established a conference with station C again over a SIP trunk. When station A hung up, station B still displayed "conference" followed by the remote phone number. | 103277 | |
| Agent display showed **UNKNOWN NAME** when an incoming trunk call was transferred over a SIP trunk and the outgoing call was finally redirected to a VDN which queued the call to an agent. | 103285 | |
| Network Regions without Media Gateways could not be disabled using the disable nr-registration command. | 103300 | |
| Communication Manager 6.x did not disable SIP Direct Media when **Avaya IE** (IEEx, a former Tenovis product) was involved in call origination. | 103328 | |
| Activating Network Call Redirection through route-to-number failed to go to the next step when the far end station was busy. | 103330 | |
| When using Avaya Site Administration (ASA) to administer an Agent-LoginID form, the tool allowed a blank Native Name when the Native Name Script was non-NULL or a NULL Native Name Script when the Native Name was not blank. When this mismatch occurred the administration record for the Agent-LoginID was corrupted. | 103337 | |
| Occasionally, call failures occurred during heavy call traffic through an Acme Packet Session Border Controller. | 103373 | |
| The **Logged-In ACD Agents** and **Logged-In IP Softphone Agents** fields on the **system-parameters custom-options** screen displayed a default value of **0** instead of **5200** when there were no licensed Elite Agents. | 103379 | |
| Occasionally, a reset board command initiated from SAT for an MM710B, failed. | 103383 | |
| | | ***5 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| The system did not always select an announcement board to play an announcement from a GW which was reachable by the listener when an audio group was configured to include many GWs in many different NRs. "Reachable" defined as having an administered ip-codec-set between the region of the GW and the region of the listener. | 103387 | |
| Call was dropped when an EC500 mobile answered the call that had covered out of Modular Messaging (MM). This EC500 was activated for the station in coverage-answergroup to which call had covered. | 103388 | |
| Occasionally, Communication Manager reset. | 103402 | |
| Users were not able to change their destination cell number from any station or cell phone using SAFE code for EC500. The SAFE feature failed when ARS was done through partition-route-table. | 103415 | |
| The customer could not submit **`duplicate station`** command on SAT when there was a **send-nn** button on the station being duplicated. The customer saw the following error message: Error encountered, can't complete request; check errors before retrying The customer would have to remove the button before duplicating the station. | 103416 | |
| For calls on IQ/CMS measured trunks which are queued to multiple skills through vectoring multiple-skill queueing, a DEQUEUE event was not being reported to IQ/CMS for skills in which the call was NOT answered. The result of this was the appearance of a stuck call in queue on IQ. | 103426 | |
| Three agents, all using the A175 were labeled phone A, B, and C. A and C were released from a call when B, while on a call with A attempted to transfer the call to C when C's contact was in E.164 format. | 103428 | |
| Calls over IP trunk via PROCR failed after WARM restart. The condition was triggered when an IP trunk call was recieved via PROCR while the system was executing a WARM restart. IP trunk calls then continued to fail after the restart was completed. | 103436 | |
| Remote service observer over a SIP trunk dropped after receiving 200 OK in response to a re-INVITE. | 103445 | |
| Previously, under heavy traffic, a CLAN audit would inadvertently tear down a listen socket. In this release, the audit process has been corrected to avoid this error. | 103446 | |
| | | ***6 of 10*** |

Table 13: Fixes delivered to Communication Manager 6.0.1 SP2

| Problem | Keywords | Workaround |
|---|---|---|
| Remote Service Observing call over a SIP trunk dropped when it stayed up more than 25 minutes. | 103447 | |
| For some call scenarios, a call may not properly cover to voice mail. | 103457 | |
| Group page call failed when group page had maximum number of group pages. | 103466 | |
| A new incoming SIP call attempted to request service from a media gateway when all media processing resources were made up of H.248 media gateways and all the media gateways were at capacity. The request was denied as expected. However, the system did not reject the incomingSIP call and the SIP trunk remained active for several minutes. Now the incoming SIP trunk is dropped immediately. | 103471 | |
| Restarted http with **graceful** option when httpd log was rotated. | 103472 | |
| Sometimes generic greeting was heard when a call was made from a bridge appearance of an x-ported station to voice mail. | 103478 | |
| SIP mutated dsv-format caused Communication Manager to reset. | 103479 | |
| SIP Mutated media-field dsv caused Communication Manager to reset. | 103480 | |
| Phone display changed to SIP trunk name after shuffling. | 103481 | |
| SIP Mutated sdp time-field value caused Communication Manager to restart. | 103482 | |
| The Avaya One-X Agent application didn't start a video window when the wrong network region was used in ProcRRQ when getSupportedFeatureS was invoked. | 103489 | |
| The customer was not able to make internal or external calls when the time of day chart was set to 11 for the stations COR (Class Of Restriction) when SA9050 - Increased TOD (Time Of Day) Routing Tables/Partition Grp Num was enabled on system-parameters special-applications. | 103492 | |
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**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| The values entered for the fields Display Language and Multiple Unicode Message File Support in the add station screen reset to blank after the user navigated to the next page and navigated back. The field Display Language was initially set to unicode2/3/4 and Multiple Unicode Message File Support was set to Y and station of type 1408/1416 was added using the add station command for SA8942. | 103495 | |
| An IP extension was forced to log off when One-X communicator logged in, in a shared control mode and another IP station logged into the same extension with an unnamed registration. | 103538 | |
| Using IAS (Idle Appearance Select) FNE (Feature Name Extension), the call-appearance on the principal OPTIM(Off-PBX Telephone Integration and Mobility) station was locked for 30 to 45 seconds when an invalid number was dialed from an EC500 phone. This issue occurred only when SA9106 was enabled. | 103563 | |
| When systems that did not have the PNC Duplication feature enabled tried to add a new ATM PNC connection or change an existing ATM PNC connection using the atm pnc form, the user may have seen the following error message: "Error encountered, can't complete request; check errors before retrying". | 110028 | |
| A Network Region with no Media Gateways could not be disabled and an error message was displayed. | 110040 | |
| Occasionally, the subnet mask was incorrectly displayed on the **display IP-interface PROCR** screen of SAT after a change was made to the subnet mask on the Ethernet interface used for the PROCR interface. | 110051 | |
| Originator did not hear ringback for a call that involved R2MFC and a H.323 trunk. | 110061 | |
| Communication Manager-ESS sent subscribe messages to Messaging Manager for stations in the inactive state. | 110110 | |
| | | ***8 of 10*** |

**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| When Communication Manager was a tandem switch between two SIP trunks, a call made between the two trunks failed when the following conditions were all true:<br>1) The incoming SIP trunk's signaling group is administered with the field **Initial IP-IP Direct Media** set to Y<br>2) The outgoing SIP trunk's signaling group is administered with the field **Initial IP-IP Direct Media** set to N<br>3) The far end switch responds with SDP in the progress message | 110117 | |
| After a SAT-initiated "reset system 4", ACD agents could not login. Login attempts generated denial events 1040 and 2126. | 110160 | |
| Under certain IP network conditions, Communication Manager system temporarily lost memory when the network conditions impaired communications with H.248 Media Gateways. | 110163 | |
| The call dropped when a video call was placed on hold and then resumed. | 110173 | |
| Occasionally, use of GIP internal socket 0 caused a new call to fail. Socket 0 is no longer made available to be used. | 110206 | |
| Communication Manager did not play local ringback for ISDN PRI to a SIP tandem call when the outgoing SIP trunk call used media resource which was capable of early media detection. | 110207 | |
| Calls failed on Communication Manager for a brief period of time (up to one minute) when a H.248 media gateway connection was temporarily lost. This occurred when the media gateway connection was lost for a time, up to, but not greater than the H.248 link loss delay timer setting. | 110222 | |
| On a Communication Manager system, a call made between a station on a H.248 media gateway and a station on a legacy port network lost talkpath. The loss of talkpath occurred when the H.248 media gateway lost connectivity with the main server, failed over to a LSP or ESS and then re-connected with the main server. | 110253 | |
| SOSM passive monitors did not send events after a conference or transfer. | 110289 | |
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**Table 13: Fixes delivered to Communication Manager 6.0.1 SP2**

| Problem | Keywords | Workaround |
|---|---|---|
| Traffic to some H.323 IP stations failed when Communication Manager PE (Processor Ethernet) server interchanged or failed over to ESS (Enterprise Survivable Sever). | 110324 | |
| During a conference call between a SIP phone and two H.323 phones (controller and originator of both calls was SIP phone), after SIP phones pressed join to conference, there was no talk path. The conference dropped when any key was pressed on a H.323 phone. SIP trunk to Session Manager had Direct Media enabled and H.323 phone had Direct IP enabled. | 110352 | |
| The **`list cor`** command repeatedly displayed 0 as its first entry in the output instead of incrementing by one and displaying the range of 0 through 995. | 110381 | |
| SIP calls with Audiocodes M3K gateway failed. | 110447 | |
| **in-trk-code** field of CDR was not captured correctly for custom CDR format. | 110457 | |
| A Memory leak associated with SIP MWI processing (NOTIFY from SIP Modular Messaging to Communication Manager via Session Manager) resulted in a system reset. | 110580 | |
| | | ***10 of 10*** |

# Problems fixed in Communication Manager 6.0.1 SP3

**Table 14: Fixes delivered to Communication Manager 6.0.1 SP3**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP3. | 103539,<br>110014,<br>110165,<br>110286,<br>110315,<br>110378,<br>110438,<br>110547,<br>111359. | |
| Internal processes in Communication Manager ran out of resources and reset when a large number of administration commands were run. | 101167 | |
| Communication Manager attempted to handle the case where the far end reduces the number of "m=" in the SIP session descriptor which could cause dropped calls if IP video was enabled on the endpoint. | 102293,<br>110164. | |
| Communication Manager assigned announcement resources from a remote network region to a held call when local announcement resources could have been used instead. | 102343 | |
| On a SIP call made from a one-X Communicator to Polycom RMX, video was lost when the one-X Communicator transferred the call to an Avaya 1000 series endpoint. Audio was not affected by this problem. | 102661 | |
| SIP Session Refresh timer caused calls to drop when Communication Manager underwent a server interchange with a large number of SIP calls were active on it. | 102823 | |
| An incoming LAI (Look Ahead Interflow) trunk call to an agent was not logged on the phone. | 102825 | |
| Port networks stayed out of service after a network outage driven interchange. The system was restarted for recovery. | 102875 | |
| A call dropped and an invalid avext parameter was inserted when the contact header contained an alpha user par in a third party call control with SIP REFER. | 102878 | |
| | | ***1 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| A descriptive error was not displayed when a call made on an Avaya 1000 series endpoint was denied due to insufficient bandwidth. | 102883 | |
| More than 40,000 stations could not be added. | 102935 | |
| Network problems that caused momentary differences in the number of IPSIs connected to each server in a duplicated server pair caused a server interchange that did not preserve existing calls. This occurred since the operation of the recovery software that waits for the connections to settle down on one server did not take into account the state of the recovery software that waits for the connections to stabilize on the other server. | 103017 | |
| The number displayed on a caller phone was incorrect when a call was made over a SIP trunk and was covered to voice mail. | 103071 | |
| Call to third party T3 Telecom messaging system failed because a SIP 200 OK message had an extra cr/lf at the end of the Session Description Protocol. | 103077 | |
| An incoming analog trunk call with busy tone detection did not disconnect when the call was answered by voice mail. | 103147 | |
| During heavy video traffic, MCSNIC (Mask Calling Number/ Station Name for Internal Calls) code for SIP stations caused the system to reset. | 103178 | |
| H.323 link loss delay timer was not applied for non Avaya endpoints. | 103197 | |
| Incoming calls to a VDN (Vector Directory Number) extension were sent to a vector. The vector sent the caller to an attendant console group where the answering operator was unable to transfer the call from the Softconsole. | 103272 | |
| Some digits were truncated and sent over ASAI when the extension size was larger than what ASAI supports. The redirecting numbers were expanded incorrectly while sending them over ASAI. | 103306 | |
| When the B-side IPSI was active, changing the A-side IPSI **Preference** to Disabled caused the IPSI daily interchange to stop working. Now, when IPSI **Preference** is disabled, IPSI interchange will work regardless of which IPSI is active. When IPSI **Preference** is enabled, daily IPSI interchange works as before. | 103324 | |
| | | ***2 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| SIP calls involving the REFER message resulted in stuck trunk members. | 103363 | |
| The BTD trunks were not released when the caller dropped and the incoming BTD trunk call on PN was transferred over another BTD trunk on PN. | 103417 | |
| Only the far end number for a VDN call that was routed to a Cisco extension over a SIP trunk was observed on the originating SIP station. | 103450 | |
| Customers observed survivable server keep-alive alarms when a new survivable server was administered. | 103458 | |
| Incorrect socket peg counts and usage measurements were observed on the **`list meas CLAN socket`** reports. | 103474 | |
| An incorrect display beginning with an ***** and ending with a **0**, was displayed at the originating station when a **200 OK** message with **+** as the leading digit was observed in the P-Asserted Identity Header. | 103532 | |
| A call hung in vector processing on the originating server and was torn down after two minutes when a vector route-to step initiated LAI (Look Ahead Interflow) and the receiving communication server deflected the call back to the originating server via NCR (Network Call Redirection), on SIP trunks, before the call gave answer supervision on the receiving server, however, after the call was deflected back to the originating server. | 110019 | |
| The top line of a 96xx phone displayed the old extension instead of the new one when the **`change extension-station`** command was run. | 110022 | |
| ARS digit conversion did not work for more than 18 digits. | 110024 | |
| Digit conversion in Calltype Analysis form was ignored when the call was routed via a second (or later) preference due to LAR on the first preference. | 110049 | |
| Occasionally, the calling party number truncated when a prefix was inserted to a large calling party number extension. | 110060 | |
| When a call was made from Avaya SIP phone to CISCO phone over a SIP trunk with shuffling enabled and direct media disabled, after CISCO phone did hold and unhold operation, call did not shuffle to IP. Because of this media resources were not released. | 110062 | |
| | | ***3 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| Video was lost when Avaya one-X® Communicator transferred a call to an Avaya 1000 Series Video endpoint in a Communication Manager Evolution Server environment. | 110103 | |
| Occasionally, all sockets on PROCR stopped working shortly after a server interchange. After a few minutes, the system restarted and recovered the PROCR. | 110171 | |
| dtmf detection on Communication Manager failed when the field **system parameters - ip-options**, set "Prefer use of G.711 by IP Endpoints Listening to Announcements" was set to Y. | 110184 | |
| Occasionally, DSP resources were tied up on incoming SIP trunk call and were not released. | 110185 | |
| IP DECT (Digitally Enhanced Cordless Telephone) did not activate Auto-Callback after rejecting an auto-callback call. | 110191 | |
| Under high traffic conditions, customers with a large number of Media Gateways registering through CLANs suffered from poor Media Gateway performance. | 110230 | |
| No talk path was observed after a call made from Xlite phone to an Avaya SIP phone over SIP trunk with SA8965 and shuffling enabled was answered. | 110236 | |
| **`list trace station`** and **`tac`** would output SIP messages for only one SIP endpoint on a call when it should handle multiple SIP endpoints. | 110277 | |
| The call was not routed to the correct mailbox and the voicemail server responded with 302 when the call covered to SIP voicemail server. | 110284 | |
| SA8439 Forward Held-Call CPN feature did not work. | 110290 | |
| Removing and adding an active CTI link caused the CTI link to have the wrong attributes. | 110293 | |
| Activation of PLAT-rhel4-1010 failed because the NTP RPM took too long to install (> 60s). This left the system with two NTP RPMs installed that again resulted in failure. Manual intervention was required to correct the situation. | 110305 | |
| The caller saw a garbled display for an outbound call over an ISDN trunk that was diverted, when **SA8146 Update Display for Redirected Calls** on change system-parameters special applications screen was set to **Y**. | 110314 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| The SA8312 Meet-Me Paging feature did not work for the first 30 minutes after the upgrade. | 110319 | |
| The setup_hp-sshd script generated errors about chkconfig not found when activating/deactivating a Communication Manager update/service pack. | 110325 | |
| Occasionally, calls dropped on the Media Gateway fallback to the main server from an ESS. | 110331 | |
| "list trace ras forced_urqs" could not be used to trace if a URQ was sent for the IP stations when doing "reset ip-stations all-ip ip-network-region xx". | 110362 | |
| IP softphones in share control mode got a latent on-hook that caused the ASAI to drop. | 110364 | |
| With SPI language version 23 (new in Communication Manager 6.0), agents making or receiving personal calls while in Aux Work with a reason code other than 0 showed a change in the Aux Work Reason code in the CMS Reports. | 110402 | |
| SOSM supplementary feature query did not report stations that were in pickup groups. | 110406 | |
| On a call from an Avaya 323 phone to an InTouch phone over a SIP trunk, the InTouch phone put the call on hold for longer than session refresh timer of the SIP trunk. There was no talk path when the InTouch phone un-held the call. | 110414 | |
| IAC (International access code) was not inserted before the calling number that started out as national or local before being sent to ASAI. | 110419 | |
| Communication Manager sent a 500 Server Internal Error in response to m-line with a=inactive in the initial invite sent and dropped the call. | 110441 | |
| The system did not send alarms after invalid login attempts. | 110442 | |
| A call placed by SIP station to a SAC enabled H.323 VPN station, failed. | 110448 | |
| No announcement was heard after transferring a held call to the announcement extension. | 110453 | |
| **`list trace station`** button option did not work with SIP stations. | 110454 | |
| ASAI notified the application that a work mode change took effect immediately even when the request was still pending. | 110471 | |
| | | ***5 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, Communication Manager did not communicate with H.248 media gateways. | 110483 | |
| The MVSubAgent was crashing when walking the G3-MIB. | 110497 | |
| The **`list ip-interface clan`** command returned garbage data causing Session Manager initialization to fail. | 110503 | |
| After a transfer, station display showed INT instead of INTL. | 110504 | |
| EC500 station was not displayed in a conference party display. | 110523 | |
| It was not possible to blind transfer a call to a station with 2 lines busy when the call originated from outside and **External Ringing for Calls with Trunks** was set to **all-calls**. | 110525 | |
| Communication Manager could not parse a SIP NOTIFY without the "Message-Account" header. | 110545 | |
| Under load, the H.323 signaling group socket closed prematurely, and some calls were mishandled. | 110549 | |
| Communication Manager could not establish H.245 connection to CIE (Customer Interaction Express). | 110559 | |
| Three party ad-hoc conference calls had all parties connected to the same ad-hoc conference on the Avaya Aura Conferencing. | 110586 | |
| Internal calls using ARS could not be Service Observed. | 110625 | |
| Avaya Site Administration GEDI feature and Native Configuration Manager session connections to a server would drop when the following measurement commands were executed: **`list measurements ip dsp-resource summary last, list measurements ip dsp-resource hourly 1`** | 110642 | |
| History information headers for subsequent coverage points were not seen in the INVITE message when a remote coverage point was present in a coverage path. | 110643 | |
| Customers with translations that contained more than 128 ip-interfaces with the first administered interface being PROCR experienced a SAT lockup when attempting to add a new or change an existing ip-interface. | 110650 | |
| Communication Manager deleted one via header and sent one via header while sending a ReINVITE message which caused the call to drop. | 110654 | |
| | | ***6 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| `list trace station` and `tac` commands failed to output location information when the call was routed using the AAR/ ARS feature. | 110657 | |
| Agents had to push the call-appearance button manually to answer incoming calls. | 110708 | |
| CPN (Calling Party Number) was not captured in CDR for incoming SIP trunk call when CPN was prefixed with a **+**. | 110732 | |
| Communication Manager returned 503 response code for the first incoming call and 403 for the subsequent calls when all SIP trunks were busy. | 110735,<br>110737. | |
| A call transfer or conference (REFER) made from a SIP station to a non SIP station whose extension did not have a mapping in the public unknown number table (or the private table if that is used for Session Manager number transformations) failed. All extensions on Communication Manager must have public numbering table patterns. This problem occurred when Communication Manager was used as a trunk gateway and the refer-to destination was off-PBX. | 110755 | |
| All soft buttons and calling party information on the phone was lost when the EXCLUSION button was deactivated on 96xx phones. | 110767 | |
| A call transfer or conference (REFER) made from a SIP station to a non SIP station whose extension did not have a mapping in the public unknown number table (or the private table if that is used for Session Manager number transformations) failed. All extensions on Communication Manager must have public numbering table patterns. This problem occurred when Communication Manager was used as a trunk gateway and the refer-to destination was off-PBX. | 110775 | |
| Only some of the Call Record fields were parsed using previous Message Tracer Analyzer version. Message Tracer Analyzer 6.4.2.8 has support for parsing all Call Record fields. | 110820 | |
| List trace station failed to record ASAI calls. | 110835 | |
| No talk path was observed at party B when a conference call was made between three SIP phones (A, B and C) with SDM/ shuffling ON for trunk towards Session Manager. | 110874 | |
| | | ***7 of 8*** |

| Problem | Keywords | Workaround |
|---|---|---|
| The Per Button Ring Control feature did not work with the correct values when the Rg field on the station form of an analog station was set to A (abbreviated), D(delayed) or N(no-ring) and a call was made to the analog station, the above administration was reset to R(ring). | 110876 | |
| SIP A called SIP B. SIP trunk towards Session Manager had SDM enabled. DTMF was not heard after answer. | 110891 | |
| IPv6 phones on a remote subnet did not come into service for several minutes after a server interchange because the Neighbor Cache in the near-end router was not updated after the interchange. | 110936 | |
| Media Gateway message rates were unduly restricted. | 110984 | |
| Occasionally, outbound calls made from one-x communicator through a session border controller failed because of extra characters inserted in the avext parameter by Communication Manager. | 111117 | |
| Communication Manager reset when an Avaya 1040/1050 Video Multipoint Control Unit attempted to conference in a 5th party. | 111180 | |
| | | ***8 of 8*** |

# Problems fixed in Communication Manager 6.0.1 SP4

This release includes the following fixes delivered to Communication Manager 6.0.1 SP4.

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***1 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP4. | 100727,<br>110149,<br>110501,<br>110684,<br>110859,<br>110879,<br>110899,<br>110946,<br>110990<br>111002,<br>111025,<br>111119,<br>111129<br>111244,<br>111270,<br>111372,<br>111388,<br>111421,<br>111501,<br>111521,<br>111812,<br>111814. | |
| 2 QSIG trunks were found active even after the transfer was completed. This happened because QSIG Path Replacement failed after an incoming trunk call to a VDN was routed to an EC500 station via a HUNT group and the call was answered with a cellular phone and transfered back to Communication Manager via a QSIG trunk. | 100189 | |
| | | ***1 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***2 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| The inter region bandwidth utilization for calls made between SIP phones belonging to 2 different Network Regions was not handled correctly.<br>**Note:**<br>If any change is made to the Network Region of a SIP endpoint, then the first call to that endpoint will follow the old call topology. All the subsequent calls will work fine. | 101586 | |
| This fix reduced the possibility of the var partition being filled by errant applications that posted excessive messages on the logs. | 102142 | |
| Video calls made from SIP to H.323 interworking endpoints did not support high definition video. | 102695 | |
| An incomplete abbreviated dialing button was observed on button 5 of a station when the set type of the station was changed from analog set type to non-analog set type and the analog station had a Hot Line Destination. | 102874 | |
| An EC500 user calling another user in a network of Communication Managers was unable to reach the voice mailbox of the called endpoint when the call was covered to voicemail. Instead, the calling EC500 user was prompted to enter the password to access the user’s own voice mailbox. | 103190 | |
| Incorrect name was displayed on the calling party endpoint after a blind transfer to PSTN over a SIP trunk. | 103294 | |
| One way audio was observed on a call made from a Polycom VVX video phone to an Avaya voice only endpoint. | 103326 | |
| The Active server was restarted after registering an IP phone to an extension that was administered with softkeys and was Unicode-enabled. | 103413 | |
| Users were unable to SSH into a TN2501AP VAL board. | 110026 | |
| The caller had to enter more digits than required by the MLDP (Multi-Location Dial Plan) feature for calls to route correctly. This only happened when the call routed to a VDN through attendant night service. | 110039 | |
| Log External Calls was not activated for a call that terminated into a VDN and routed to a station. | 110137 | |
| | | ***2 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *3 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| The Duplicate vector SAT command failed and displayed the error `Native Name should be specified for Script Tag` after a vector was assigned to a VDN. | 110330 | |
| H.323 video calls dropped during hold and unhold operations after the calls were tandemed through Communication Manager via H.323 trunks. | 110347 | |
| A 1608 IP phone displayed `System Busy` because it was not registered. | 110351 | |
| Call forwarding busy don't answer did not work for an endpoint that was active on a call. | 110389 | |
| A call, made from an external video phone to an Avaya One-XA via Ingate SBC on a SIP trunk that had shuffling enabled, dropped after it was answered. | 110431 | |
| History Info Header was inserted in INVITE messages for Survivable mode remote coverage point even when Communication Manager was not in the LSP or ESS mode. | 110449 | |
| One-XM mobile phone rang even after it was disconnected from One-XM server. | 110524 | |
| A generic greeting was heard when a call was re-routed over QSIG to a QSIG MWI hunt group. | 110546 | |
| The call forward feature could not be deactivated when the feature button was administered after the call forward feature. | 110555 | |
| Communication Manager updated the team buttons label:<br>1. When the name of a station changed & the other station, whose team button label had to be updated, was in service mode, without further user or administrator action. The update could be delayed depending on how busy the system is at the time of the name change.<br>2. When the other station (on which the update should take place) was in service mode after the name of a station had changed. | 110568 | |
| Calls to SIP endpoints including Modular Messaging from stations with a tilde (~) in their name field on Communication Manager failed. | 110570 | |
| | | ***3 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***4 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| Calls that were answered by voice mail after coverage were shown as answered calls in the call log of the principle termination after the **Maintain SBA At Principal** field on **change system-parameters coverage-forwarding** screen is set to **Y**. | 110582 | |
| MWI (Message Waiting Indication) audit was not completed when Communication Manager had a high number of users. | 110585 | |
| There was no talkpath when a call was made over a SIP trunk and the far end sent codec that was second in the list on the **IP-codec-set** screen to the near end. | 110603 | Workaround: Do not have codec negotiation. |
| SIP trunk was held during NCR after the SIP station did a blind transfer while shuffling was ON. | 110615 | |
| History Info header was not formatted properly. | 110646 | |
| The SAT **`duplicate station`** command displayed `Error encountered, can't complete request; check errors before retrying` after a **`list station`** command was executed. | 110662 | |
| Automatic Call Back feature over two SIP endpoints, that were configured for more than 11 digits on two different switches/servers and managed by a common System Manager, did not work. | 110674 | |
| The OPTIM station was stuck in the in-service/active state after a conf-on-ans call was dropped. | 110701 | |
| CDR did not capture the agent extension that answered the call after it traversed through Voice Portal and conversed on vector steps. | 110734 | |
| Call recording failed when the call recorder was added to an incoming SIP trunk call and an existing SIP transaction was in progress. | 110754 | |
| No audio and video was observed when a video call was transfered to AAC (Avaya Aura Conferencing). | 110757 | |
| Agents using Avaya one-X® Agent were unable to log in until a call was made. | 110758 | |
| A memory leak was observed when the system had the MLPP (Multiple Level Precedence Preemption) feature enabled and the 96xx IP station was provisioned with the DSCP (Differentiated Services Code Point) capability. | 110760 | |
| | | ***4 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *5 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, BRI station with **bridge** button posted an internal stim to drop a call even when it was not active on that call. | 110761 | |
| SYNC alarms were observed on an inactive H.248 LSP. | 110764 | |
| User-to-User Information, received as escaped parameter, was sent twice in subsequent messages. | 110772 | |
| A call, held and unheld by ADVD (Avaya Desktop Video Device), was dropped. | 110787 | |
| The analog line boards were sent Bellcore provisioning when the **Analog Terminals** field in the **change system-parameters** screen was set to **V23-Bell**. This caused the boards to use incorrect caller ID protocol. | 110791 | |
| Incoming caller ID over analog trunks did not work correctly when the **country code** was set to **UK** on the **trunk group** screen. The trunk group data was displayed instead of the actual caller ID. | 110816 | |
| Calls made by customers to agent endpoints with extensions that were longer that 7 digits, with CMS R.16 and newer and IQ 5.1 and newer, and measured by the reporting adjunct, hung in the ringing state on abandoned extension-in calls. | 110826 | |
| A duplicated ESS survivable processor could not use Processor Ethernet to connect with CMS or IQ. The same restriction applied to a simplex ESS survivable processor. LSP survivable processors could not be defined with IPv6 Processor Ethernet connectivity. | 110827 | |
| The 6th and 7th **NPA** field value on the **locations** screen changed after a patch was installed and the system was reset. | 110831 | |
| The **Alarm Groups 601-666** field on the **set options** screen was blank. | 110840,<br>111706. | |
| The system was reset when the length of the incoming SIP message was greater than 9216. | 110856 | |
| Avaya Desktop Video Device could not join the adhoc conference on Meeting Exchange. | 110857 | |
| The **Transfer** softkey was not visible when a call transferred to a busy station was cancelled. | 110863 | |
| | | ***5 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***6 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| When a 96xx phone logged in as an agent and transferred an ACD call directed to a VDN with VDN of Origin Announcement, **Drop** softkey was seen instead of the **Release** softkey after the transfer was cancelled. | 110875 | |
| Port network H.323 IP stations and DCP stations acting as listen only service observers could not hear DTMF tones generated using out-of-band methods and produced by other parties on the call. | 110878 | |
| Incoming R2MFC trunk calls on MG failed with a mixed dial plan. | 110886 | |
| In a system that had BCMS measurement enabled and did not have CMS configured, a series of calls that utilized path replacement and were subsequently transferred by a BCMS measured agent caused BCMS to stop tracking calls. | 110895 | |
| Other parties in a conference call, that was hosted by an Avaya 1040/1050 MCU (Multipoint Control Unit) and had video and content activated simultaneously, dropped when the fifth party was added. | 110896,<br>110958. | |
| HistoryInfo header with port and transport did not parse. | 110899 | |
| An AES DMCC endpoint registered at a Communication Manager extension that had EC500 enabled did not receive RTP streams when the user was on a call. | 110904 | |
| MWI update to OneX agent (in the Desk Phone mode) was delayed. | 110909 | |
| Outbound calls enabled with SA9114 and marked with TON subscriber or national were not presented on ASAI/PRI. | 110910 | |
| Echo was heard by the user in the case of QSIG Path Replacement. | 110931 | |
| Alert type in the INVITE to SIP Modular Messaging was incorrect for the incoming ISDN PRI calls that covered to SIP Modular Messaging via Session Manager. | 110941 | |
| There was no talk path after a Communication Manager with Port Networks and TN 2302 "Medpro" Circuit Packs completed a SIP request to blind transfer an existing call. | 110943 | |
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**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *7 of 12***

| **Problem** | **Keywords** | **Workaround** |
|---|---|---|
| Translations for many vector steps, in Communication Manager systems newer than Communication Manager 3.1 and older than Communication Manager 5.2.1, did not upgrade correctly to Communication Manager 6.0 and above. The incorrectly upgraded vector steps also logged proc_errs in the error logs. | 110948 | |
| The ifDescr OID in the MIB-2 interfaces MIB Group was not properly populated with the correct hardware interface string information. | 110971 | |
| MWI indicator did not light up after leaving a message. | 110980 | |
| CLAN boards were constantly resetting even after multiple resets and a sever interchange. | 110987 | |
| A SIP message that contained multipart/mixed content appeared in the log file on the server: capro 27537 MED MIME Content Parse Can't find starting boundary. | 110990 | |
| Communication Manager did not allow SIP INVITEs without media, negatively impacting features like Callback Assist. | 111001 | |
| A call to a CMS measured agent that was subsequently conferenced or transferred to a CMS measured trunk caused CMS to abort tracking the call when it was dropped. | 111008 | |
| The VDN return destination failed when Avaya ACR and Avaya Quality monitoring recorded a call. | 111013 | |
| On Communication Manager system, intra-H.248 media gateway calls dropped after the media gateway failed over to an LSP or ESS or re-established communication with the main server. The call dropped when a situation encountered was similar to this:<br>1) MG#1 on Main: MG#1 DCP#1 called MG#1 DCP#2.<br>2) MG#1 failed over to an LSP or ESS.<br>3) Original call between DCP#1 and DCP#2 was terminated.<br>4) New call was made by either DCP#1 or DCP#2.<br>5) MG#1 came back to the main server. | 111018 | |
| When the ping test run against a CLAN board's ethernet port failed, the sockets used were not deallocated properly. This caused several software error logs (proc_errs) to be printed out after the socket audit discovered stale sockets that needed to be freed up. | 111024 | |
| | | ***7 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *8 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager had old data for ports belonging to H.248 media gateways after the media gateways were reset. Occasionally, the first calls made to or by the users owning these ports failed. | 111027 | |
| No Music on Hold Port was observed after a call preserving upgrade. | 111035 | |
| Conditions causing a SIP REFER to fail prevented a subsequent transfer of the call from being reported to IQ/ CMS. | 111043 | |
| The call-appearance on a deskset was locked when a call made from a mobile via the callback feature on the One-X portal was rejected before getting connected to the destination endpoint. Gradually, all the call-appearances got busy and the user could not receive calls. | 111048 | |
| Vectors programmed to allow a call to cover twice did not allow coverage for a second time when the user entered unexpected digits at a prompt. | 111058 | |
| Migration from Communication Manager 4.0 to Communication Manager 6.0.1 contained incomplete IP Interface translations, setting the translation corruption flag. | 111061 | |
| Audio and video SIP calls between two different types of endpoints on Communication Manager dropped when they were placed on hold. | 111084 | |
| High bandwidth and resource usage between Port Networks resulted when an incoming SIP trunk could not be shuffled. | 111103 | |
| Calls involving H.323 endpoints were dropped when the H.323 endpoints received media from a H.248 media gateway and the H.248 media gateway migrated to an ESS or LSP or back to the main server. | 111106 | |
| Ocassionally, terminal update messages were sent to incorrect port locations. These messages caused extreme problems to the receiving entity, such as port shutdown and blank displays or incorrect lamps. The most noticeable of these problems was bad messages sent to maintenance test circuits that caused disruption in the ability of the board to detect authentic packet bus or tdm bus faults. | 111120 | |
| | | ***8 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***9 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| IP phones were unable to register to CLANs, failed ISDN D-channels could not recover, and any other feature and service that used PKTINT LAPD links could not recover. These problems were caused by a CLAN board that failed to answer some LAPD link inquiries. This caused the table to fill up, blocking LAPD link recovery actions. | 111123 | Workaround: Replace the bad CLAN board and then execute a 'reset system 2' to clear the table. |
| SIP calls dropped when outgoing the SIP trunks used H.248 media gateways for VoIP and the far-end SIP trunk responded to an initial INVITE with 180 Ringing and SDP. | 111124 | |
| A rarely executed error log had a fault which caused a system reset. The recovery code required an artifical software load with embedded errors in order to become active. | 111129 | |
| The second line display of CallMaster cleared when **Clear callr-info** was set to **next-call** or **Leave-ACW** and the call was dropped. | 111176 | |
| EC500 calls, routed over Session Manager, did not follow location based routing. | 111183 | |
| Occasionally, Communication Manager performed a reset system 2 during SIP trunk calls. | 111186 | |
| | | ***9 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *10 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| ESS did not register because SID (System ID) reverted to a default value. The problem was triggered by installing Communication Manager 6.x software on an ESS server, without administering the **server role**. Later, the **server role** web page was used to administer the role ESS, setting the appropriate SID and MID. The problem occurred after using a SID that was not the default value, that is **1**. On the web page, the **change** button was selected which caused a pop-up window to appear. Then the **Restart Now** button was selected. The problem could be avoided by:<br>1) Administering the **server role** when the software was first installed.<br>2) Using the default system ID **1**.<br>3) Selecting the **Restart Later** button rather than using **Restart Now** button after administering the "server role".<br>The following steps were taken when the problem had occured previously.<br>1) The /etc/opt/ecs/tmp/fsy_ecs_SystemID file was removed.<br>2) The correct SID via the **server role** web page was re-administered. | 111210 | |
| A small percentage of SIP calls that were not correctly entering the connection preservation state and resulting into a link pull/Session Manager failure was fixed. | 111223 | |
| CMS/IQ did not receive an event when the **Stroke Count 6** and the **Stroke Count 7** button was pushed. Stroke Count buttons 0-5 and 8-9 worked correctly. | 111235 | |
| Communication Manager reset when an Avaya 1040/1050 video Multipoint Control Unit conferenced in a 5th party. | 111245 | |
| Tandberg H323 video endpoint registered to a Tandberg VCS tries to call an Avaya SIP video endpoint registered to Session Manager. The resulting call has audio, but only ONE-WAY video. The Tandberg H323 endpoint receives video from the Avaya endpoint. However, the endpoint has no video from the Tandberg. | 111246 | |
| Occasionally, outbound calls made from One-X Communicator through a session border controller failed because of the extra characters that were inserted into the avext parameter by Communication Manager. | 111247 | |
| No video was observed on Polycom SIP HDX calls to/from Polycom H.323 HDX. | 111250 | |
| | | ***10 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4 *11 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| A system was configured with a SIP signaling group that had Direct IP-IP Early Media enabled, SIP endpoints assigned to a network region (NR1), a port network, H.248 media gateway, media processing resources and other stations assigned to another network region (NR2), and a CAC limit assigned for NR1 to NR2 connections. Although no BW was required on the exhausted NR1-NR2 link, calls between NR1 SIP endpoints were blocked when the CAC limit for NR1-NR2 had been reached. | 111254 | |
| Blind call transfer over a SIP trunk did not trigger display update on pickup group members. | 111255 | |
| The enhanced pickup call alerting notification was delayed after the call was transferred over a SIP trunk. | 111256 | |
| On Communication Manager, users could not originate voice calls from world-class BRI data modules that were connected to H.248 media gateways. | 111257 | |
| A call, answered by a SIP endpoint with SIP Direct Media enabled on bridge appearance, was dropped after 32 seconds. | 111284 | |
| Agents could not log in because the BCMS measured agents limit had been exceeded. In an EAS environment, when the BCMS Measured ACD Members (as shown on the **display capacity administration** screen) exceeded the system limit, the number of BCMS Measured Agents was not maintained correctly when the agents logged in and logged out. | 111309 | |
| List trace station failed to record ASAI REFER for 3PMC. | 111316 | |
| `Error encountered, cannot complete request` message was displayed after executing **`enable sync media-gateway`** when the capability was enabled. | 111373 | |
| Occasionally, a Service Pack could not be removed because the shared library was accessed by a transient process, like fasttop. | 111410 | |
| There were problems with hold/unhold and NCR. | 111421 | |
| The SAT command line help for commands that use the keyword list that is derived from screen field keywords, like the **`list station type`** command, was not displayed properly. | 111431 | |
| Content sharing did not work between two SIP video endpoints. | 111475 | |
| | | ***11 of 12*** |

**Table 15: Fixes delivered to Communication Manager 6.0.1 SP4** ***12 of 12***

| Problem | Keywords | Workaround |
|---|---|---|
| A system configuration consisted of network region 1 and network region 2 containing media processors and shuffled IP stations in cabinet 1 and cabinet 2, respectively. Incoming PRI call used a trunk in PN1. The call routed to an IP station in network region 2 which uses PN1 media processor. The IP station in network region 2 hit transfer and got connected to a media processor from PN2. Music on Hold was played on the trunk. The IP station in network region 2 then dialed a VDN, queued to a skill and routed to an auto-answer IP station in network region 1. VOA was configured for the auto answer IP station in network region 1. The IP station was also assigned to a media processor in PN2. The IP station in network region 2 completed the transfer while the IP station in network region 1 was still listening to VOA. After the VOA completed, talkpath was established between the incoming trunk and the the IP station in network region 1. Echo was observed in the connection. | 111502 | |
| An incorrect greeting was heard at a remote QSIG endpoint that was covering to a SIP integrated voice mail adjunct. | 111505 | |
| The **`save translations all, ess,`** and **`lsp`** commands failed when the Survivable Processor was on a software load that did not have a compatible service pack. | 111521 | |
| Stable SIP calls consumed more memory, causing memory exhaustion on high traffic. | 111601 | |
| Proper unicode names were observed for SIP trunk calls and calls made on SIP endpoints. Memory pool errors and consequent rebooting of the system was also fixed. | 111640 | |
| | | ***12 of 12*** |

# Problems fixed in Communication Manager 6.0.1 SP5

This release includes the following fixes delivered to Communication Manager 6.0.1 SP5.

**Table 16: Fixes delivered to Communication Manager 6.0.1 SP5**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP5. | 102034,<br>110725,<br>110839,<br>111569,<br>111595,<br>111666,<br>111697,<br>111870,<br>111872. | |
| List Trace Vector/VDN did not display skills greater than 2047 when **`goto`** or **`check skill`** was used in the vector step. Skill 2048 was displayed as **0**. Skill 2049 was displayed as **1** and so on. The maximum skill that can be administered is 8000 and was displayed as **1856**. | 101162 | |
| Active user logins on LSP servers configured for JITC requirements were erroneously locked out every 30 days. | 102018 | |
| The warning header of `480 Temporarily Unavailable` SIP message was not tandemed by Communication Manager to the calling station. | 102694 | |
| A call dropped when an unsupported codec was sent by the far-end switch to Communication Manager in a reinvite message. | 102715,<br>103274. | |
| Adhoc video conferencing using Meeting Exchange conferencing server on ADVD phones did not work properly. Video was observed only at the endpoint of the conference host. All the other participants of the conference call observed only audio.<br>Call Scenario: ADVD1 called ADVD2 and ADVD3. The spotlight of the call made by ADVD1 to ADVD2 was merged with the spotlight of the call made by ADVD1 to ADVD3 to hold the conference. The user could not hold a 3 party conference consistently. This problem occurred with all the servers and was not specific to media gateways. | 103520,<br>111152. | |
| | | ***1 of 10*** |

| Problem | Keywords | Workaround |
|---|---|---|
| MIA (Most Idle Agent) across splits did not work for BCMS Agents. | 110211 | |
| An SRTP call made from Communication Manager as Feature Server with Direct Media enabled to Communication Manager as Evolution Server with Direct Media disabled failed. | 110275 | |
| Communication Manager Feature Server did not send an outbound call event to Avaya OneX Server. | 110356 | |
| Content sharing in a People Connect video call did not work. | 110552 | |
| Print statements that were resident in the code base were activated when data was to be collected for a customer problem. Occasionally, the activation of these prints compromised on the performance of the system. | 110618 | |
| Incorrect VDN display was observed when calls that tandemed over SIP trunks and used LAI/NCR did not carry UUI data properly to the station of the agent. | 110656 | |
| QSIG value coverage did not allow the service observer to drop from the call. | 110746 | |
| Incorrect voice mail integration was observed when HistoryInfo header was not inserted in the invite after the QSIG Reroute Request was received. | 110786 | |
| On Communication Manager, a SIP-SIP call in direct media intermittently reverted to RTP and SIP-H.323 direct media calls intermittently had no talk path. | 110885 | |
| A call did not route via second preference when the default proxy on the **locations** screen was configured with a route pattern that had 2 preferences. The call did not term on the first preference. | 110925 | |
| The SMI web interface was updated to have a `200 ok return` code on non-existent pages. | 110957 | |
| An agent was pending for an Aux-work mode with a Manual-In ACD call on hold and had an active call on a different call-appearance. The pending **aux-work** lamp did not light and the **after-call-work** lamp changed to the pending state when the far end dropped the held ACD call. When the agent was idle, the **after-call-work** lamp stayed pending and the **aux-work** lamp was lit. | 110983 | |
| Priority calls, made to a station with OneX mobile integration and SAC enabled, rang softly. | 111005 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| H.264 codec with different payload types was not tandemed by Communication Manager. | 111040 | |
| The last HistoryInfo entry for a call, made from one station to another and covered to MM, was incorrect. | 111063 | |
| A call ended up on a switch because of aar digit-conversion after an ASAI deflection did not drop the principal station. Therefore, the principal station continued to ring. | 111066 | |
| When an ESS was active, IP signaling groups did not go into service properly. The **`status signaling-group`** command showed the service state as disabled. | 111094 | |
| ASAI redirect calls request was not responded to and the call failed to redirect when the ARS or AAR analysis specified a range of allowable digits (min!=max). | 111153 | |
| T.38 faxes with an H.248 media gateway failed when the acknowledgement sent in response to T.38 SIP re-invite received by Communication Manager had a datagram size that was greater than 500. | 111182 | |
| In the case of double reroute, a wrong voice mail greeting was heard. | 111202 | |
| When addressed using AAR or ARS, a vector route to step that successfully terminated to a phone number over an R2MFC trunk retained the trunk port until the calling party dropped. Vector processing remained active until both the parties dropped. When addressed using AAR or ARS, a vector route step that did not successfully terminate hung. | 111237 | |
| The coverage failed when an incoming call, over a tie trunk that had **Incoming tone (DTMF) ANI** field on the **trunk group** screen set to ***ANI*DNS***, tried to cover to SIP Modular Messaging. | 111238 | |
| Display Reinvite was sent to a far end switch. This resulted in talk path failure after unhold in SRTP SDES shuffle off scenario. | 111261 | |
| A call, made to an IP Softphone whose Telecommuter was a SIP trunk with Direct Media enabled, dropped after 32 seconds. | 111271 | |
| In an IGAR environment that has SAC activated on the principal station and a coverage answer group as coverage point, team buttons flashed GREEN on stations after all the calls were dropped. | 111272 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| A fax call over IP dropped when it was kept longer than the session refresh timer. | 111278 | |
| The first codec in an IP codec set was G.729 and the second was G.711A. A call was made from a Communication Manager analog station to a PSTN station via a SIP trunk. The PSTN station sent a re-invite with G.729A to renegotiate the codec after answering the call. This re-invite was ignored by Communication Manager. | 111288 | |
| Handling a number of board network regions was controlled by putting boundary checks on index values to avoid running out of array. | 111296 | |
| One way audio was observed when cell phone users called an agent and got fast busy after brief connect. | 111302 | |
| An incoming call made to an attendant that came through an attendant VDN dropped before it was transferred to a station that had coverage set to a voice mail. | 111317 | |
| SIP stations could not retrieve messages when IDM (Initial Direct Media) was enabled on the SIP signaling group and Telephone event payload type was configured on the trunk group. | 111326 | |
| The called party number for inbound calls marked with TON national/subscriber was not modified and sent over ASAI as an international number. | 111331 | |
| The digits, on a tandem call involving an H.248 media gateway, were outpulsed twice. | 111334 | |
| **Send All Calls Override** for priority calls and **Dialling** did not work when the called party was an analog station. | 111335 | |
| On Communication Manager, held SIP stations and SIP trunks dropped when they used H.248 media gateway VoIP resources. | 111336 | |
| Repeated DS1 board uplink alarms caused a Communication Manager warm restart. | 111342 | |
| The originated event on ASAI client did not have the modified called party number in spite of SA9114 being enabled. | 111355 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| SIP to QSIG interworking did not handle **+** in the digit string. | 111356 | Workaround: Delete the leading **+** from the digit string in the **inc-call-handling-trmt trunk-group** screen for the SIP trunk. |
| Occasionally, the **`status tti`** command displayed invalid results. The results of the command displayed either that the TTI translations were incomplete or a percentage of complete TTI translations that was greater than 100%. | 111375 | |
| The MWI lamp for a user in Siemen's QSIG network did not light when the messaging adjunct was a SIP integrated Modular Messaging. | 111403 | |
| Users could not join a conference call made over a SIP trunk on Avaya Meeting Exchange. | 111407,<br>111408. | |
| The alerting device number and the corresponding TON were reported incorrect for outbound calls with SA9114 enabled. | 111424 | |
| An incoming Release line trunk call dropped after it was transferred over another Release line trunk on a different PN to a VDN. | 111435 | |
| The **Transfer** softkey was not visible when a call transferred over a trunk was cancelled. | 111436 | |
| Only audio was observed on a dial-out video conference from Polycom RMX to Polycom SIP HDX. | 111443 | |
| System restarted when a number of calls were kept in the ringing state. | 111446 | |
| An inbound SIP trunk call dropped after announcement in vector. | 111450 | |
| The system cold reset several times. | 111452 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| Under network outages, Communication Manager could not communicate with the port-networks managed by it. Communication Manager did not have an accurate picture of the VoIP channels in-use on the MedPro boards (TN2302, TN2602) in those port-networks. After the problem was solved, the attempt to re-synchronize the internal data with the MedPro boards caused a spike in CPU occupancy. When the user had many MedPro boards in various port-networks, the re-synchronization operation drove Communication Manager to overload. | 111458 | |
| The Avaya DCP endpoint was dropped from a call when the Cisco endpoint put the call on hold. | 111459 | |
| Communication Manager was configured for **MIA (Most Idle Agent) Across Splits or Skills?** as **Y** and **Temporary Bridged Appearance on Call Pickup?** as **N**. Two stations were in a non-ACD hunt-group and a pickup-group. The second station was logged on to an EAS skill and was in an available work mode. A call made to the non-ACD hunt-group rang at the second station and was answered by the user at the first station. A call made to the EAS skill of the second station queued up and did not terminate at the endpoint of the agent even when the agent was idle and available. | 111477 | |
| The agent could not answer a transferred call with SSC, another party on the call, and **VDN return destination** enabled. | 111492 | |
| Occasionally, outgoing H.323 or SIP calls could not be made on Communication Manager. | 111501 | |
| An incorrect display was observed on the station when the **(SA8851) - Remove Caller Id from Set Display?** field was set to **Y** on the **system-parameters special-applications** screen and the incoming R2MFC trunk call was made. | 111507 | |
| An incorrect voice mail greeting was heard when an incoming QSIG call covered after it was forwarded. | 111508 | |
| Outgoing calls over an IP trunk using the Processor Ethernet interface failed after a server interchange if the standby server had a more recent service pack than the primary server. | 111511 | |
| During heavy call traffic, few calls dropped on Communication Manager with SIP Direct Media enabled. | 111519 | |
| Incorrect display update was observed on the transferred station when the user performed a blind transfer. | 111536 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager reset when it received a SUBSCRIBE message greater than 9216 bytes. | 111541 | |
| Communication Manager could run out of memory, causing a warm reset. | 111545 | |
| A video call was made from a CS1K endpoint to a Communication Manager endpoint over a SIP trunk. Communication Manager dropped the call when it was put on hold by the CS1K endpoint. | 111565 | |
| Phones did not work and the port network was rebooted. | 111566 | |
| The calling party heard music on hold when a trunk call over an H.323 trunk with Data Restriction Set was internally transferred and put on hold. | 111570 | |
| Occasionally, Communication Manager reset, resulting in temporary loss of call service. | 111580 | |
| Communication Manager displayed an incorrect name presentation indication for forwarded and redirected calls when **(SA8967) - Mask CLI** and **Station Name for QSIG/ ISDN Calls?** was enabled and **Per Station - Send Calling Number** and **Name?** was set to restricted. | 111583 | |
| **Send All Calls** and **Call Forward Override** did not work when the called party was an analog station. | 111594 | |
| Stations displayed the system-wide national cpn prefix for all the incoming national calls. | 111596 | |
| Sockets were used up even when no SIP TLS trunk was provisioned on Communication Manager. | 111600 | |
| ISDN trunk calls were torn down from the far end after receiving a number of periodic INFO messages from Communication Manager that did not conform to Bellcore ISDN connection protocol. | 111618 | |
| Ringback was not heard by the user when an incoming ISDN trunk call was tandemed over SIP trunk to MS OCS. | 111620 | |
| During high SIP call traffic, Communication Manager could reset due to heap memory exhaustion. | 111634 | |
| People Content video call made between Polycom HDX and Polycom RMX did not work. | 111677 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager had 2 IQ Call Management System Reporting Adjunct links. Names Download was in progress for the first link and a pump-up request was received for the second link. Communication Manager aborted Names Download for the first link. | 111683 | |
| An incoming SIP call dropped after it was queued on an ACD. | 111719 | |
| Communication Manager restarted when a video call was blind transferred to a Lifesize endpoint. | 111731 | |
| Users did not receive SNMP traps when the server overheated. | 111734 | |
| An H.323 trunk call made between Communication Manager and CS1K dropped after 15 seconds. | 111744 | |
| Poor and delayed video quality was observed on a call made between an Avaya 10x0 endpoint and a Polycom video endpoint when Communication Manager changed dynamic payload number for H.264. | 111749 | |
| The **System Logs SMI** web page and the associated **`logv`** bash command did not display error log entries for the current year when the Communication Manager template was built in the previous year and not installed until the current year. The **`logv`** command added the value for the year to the log entries because this was not done by the Linux Syslog logging service. The **`logv`** command did not update the year because a transition of log entries from December to January was not seen. This problem was fixed by setting the **Access Date** of the oldest error log to the **first boot** date when a new software release was installed or a software patch was activated. | 111750 | |
| Calls made from Communication Manager to an H.323 trunk connected Radvision gatekeeper did not complete. | 111758 | |
| Occasionally, Communication Manager reset because of a segmentation fault. | 111764 | |
| When a SIP station with exclusion answered a bridge call and tried to initiate another call by going off-hook, the call on bridge appearance got stuck. | 111780 | |
| Communication Manager did not send out Transfer Complete FAC when SSC parties were transferred on an active call. | 111786 | |
| Occasionally, H.323 trunk calls made via a CLAN board failed. | 111810 | |

| Problem | Keywords | Workaround |
|---|---|---|
| The buttons on a SIP phone were not downloaded properly at the time of registration when a large number of buttons was assigned. | 111857 | Workaround: Administer fewer buttons. |
| Calls made to an unplugged auto-answer IP station received no ringtone. | 111858 | |
| Avaya One-X Communicator in shared control application mode with Avaya 96xx phone displayed incoming calls incorrectly. | 111873 | |
| Occasionally, a warm reset, that was initiated by the SAT command **`reset system 1`**, escalated to a cold reset. This impacted the call service. | 111906 | |
| Malicious Call Trace feature did not work with location-specific prefixes. | 111950 | |
| SIP to SIP calls did not work when Direct Media was enabled. | 111961 | |
| No content could be shared from a Life Size endpoint. | 111973 | |
| There was no talk path after Communication Manager with Media Gateways completed a SIP request to blind transfer an existing call. | 112001 | |
| In a Direct Media setup, a SIP station heard an extra beep when it originated a call that was covered because of **Send All Call** or **Don't Answer** coverage criteria. | 112045 | |
| The bridge phone could not answer a call made from the bridge button when the **ROIF** (Redirect on IP failure) field was turned on the **system-parameter feature** screen, page 14. The **Switch Hook Query Response Timeout:** was set to a non-zero value. | 112074 | |
| The exclusion button of a SIP station was lit up even when the station was on-hook. | 112095 | |
| A SIP phone, that called into a **meet-me** conference and used the conference button to add in another user to the conference, was locked out after pushing the join button. | 112114 | |
| A user from SRTP SIP A called another user at SRTP SIP-B and SDM was enabled. The **unhold** feature at SIP B did not work after the user at SIP-B put the call on hold for a longer time than the session refresh timer. | 112192 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| On a SIP to SIP station call that was held and unheld, Direct Media talkpath was delayed by an extra 400 ms. This resulted in a delay of talkpath establishment. | 112193 | |
| A bridge phone was configured in **off PBX station mapping** screen as an OPS station and the SAC (Send All Calls) state was enabled. A call made to the primary phone ended in the coverage point of the bridge phone. | 112201 | |
| Occasionally, on Communication Manager, calls involving H.323 stations, SIP stations and trunks, and VoIP resources from legacy PN (port-network) lost talk path when the call started using the PN VoIP resource during a scheduled maintenance audit. This issue could have also affected users of traditional station types like Analog and DCP on legacy PNs who make calls to other users on other legacy PNs or H.248 media gateways. | 112212 | |
| Occasionally, authentication files failed to install. | 112257 | |
| In a conference, Communication Manager sends 2 messages to a registered SIP station about the exclusion status from PUBLISH. The first message is sent when the exclusion feature is set to ON. The second message is sent when the SIP station is excluded from the conference. When the **Exclusion** button was pressed on PUBLISH, it was observed that the order of the messages, sent by Communication Manager, was reversed. This resulted in the user, signed into the SIP station, to believe that the conference was on even after the station was excluded. | 112347 | |
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# Problems fixed in Communication Manager 6.0.1 SP5.01

This release includes the following fixes delivered to Communication Manager 6.0.1 SP5.01.

**Table 17: Fixes delivered to Communication Manager 6.0.1 SP5.01**

| Problem | Keywords | Workaround |
|---|---|---|
| ASAI redirection failed when the call was remotely redirected. | 112523 | |
| | | |

# Problems fixed in Communication Manager 6.0.1 SP6

This release includes the following fixes delivered to Communication Manager 6.0.1 SP6.

**Table 18: Fixes delivered to Communication Manager 6.0.1 SP6**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP6. | 111542, 111999, 112246, 112328, 112720. | |
| The **list measurement ip dsp-resources** screen displayed incorrect data for the **GW Number** and **GW Type** fields. | 102572 | |
| The subnet mask value for the procr ip-interface was inconsistent with the value administered on the server when the **`display ip-interface procr`** command was executed. | 102649 | |
| Calls handled by messaging were dropped. | 102655 | |
| ISDN BRI trunk ports appeared to not have been inserted properly into the system. The demand tests, when run, showed the port in a No Board state even when the board state showed that it was OK. | 103374 | |
| Occasionally, FAX over SIP failed. This happened because Communication Manager did not relay SIP messages. | 110175 | |
| During scheduled maintenance on inactive ESSs, denial events 1367 and 2117 showed up in the logs. | 110877 | |
| An audio call made from a SIP station to another SIP station dropped after 32 seconds when the **IP Video** field was set to **True** on the **signaling group** screen and **SIP direct media** was disabled. | 110966 | |
| An audio call was made from a Tandberg E20, registered to a Tandberg (VCS) Video Conferencing Server, to an Avaya 9640 SIP voice terminal. The audio was not resumed when the Tandberg E20 put the call on hold and then resumed it. | 111071 | |
| The agent endpoint displayed incorrect data for predictive dialed calls after the calls were answered. | 111215 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| An ISDN caller heard silence instead of secondary dial tone because the call could not be transferred to remote access dial tone. | 111282 | |
| Call transfers involving ADVD (Avaya Desktop Video Device) incorrectly resulted in calls with no video. | 111392 | |
| The **list meas announcements board** reports showed inconsistent data. At the top of each hour, numerous proc_errs from meas_m-proc_aus() were listed in the logs. | 111433 | |
| Occasionally, video calls did not get appropriate video resolution because Communication Manager injected H.263 as the first payload number of the video m=line when it should not be the first one. | 111485, 111701. | |
| Incoming international calls made to an X-ported station over ISDN-PRI/IP trunks covered directly to the second coverage point when the first coverage point had a SIP station. | 111518 | |
| On a call that covered to Modular Messaging through a SIP trunk, the calling station partially displayed the called station extension. | 111525 | |
| Communication Manager did not respond to a REFER msg that was received in an invalid state. | 111552 | |
| All system and application links, along with calls, were dropped. Non translation feature data was lost and Circuit Packs were reinitialized. | 111603 | |
| Bouncing of PPP links caused MWI failures and corrupted memory pool tables. | 111610 | |
| A call made from an Avaya 10x0 to a Polycom HDX SIP endpoint failed when SIREN codecs were administered. The call could not be completed. | 111626 | Workaround: Remove the SIREN codecs from the IP-codec-set in Communication Manager. |
| Customers using third party interworking gateways, like the Avistar C3 Connect and Cisco/Tandberg Video Communication Server, queried for SIP capabilities before sending the initial INVITE. When Communication Manager received an out-of-dialog OPTIONS message, it responded with the following limited audio only capabilities: G711Ulaw, G711Alaw, G723, G729, and RFC2833TEL. These capabilities do not include any video or wide-band audio codecs. | 111644 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| Shuffling was enabled on the SIP trunk when the order of codec on Communication Manager in IP-codec-set was:<br>G.711A<br>G.729<br>and the order of codec on the terminating SIP endpoint was:<br>G.729<br>G.711A<br>No talk path was observed when the terminating SIP endpoint answered the call with the above preference in codec. | 111646 | |
| In trunk group measurements, inconsistencies could be displayed between the overflow peg counts and the %ATB. There could be non zero overflow peg counts that still show 0% ATB (All Trunks Busy). | 111668 | |
| A call was made from a SIP integrated Modular Messaging system to an H.323 station over a SIP signaling group with Initial Direct IP-IP audio enabled. The call dropped when it was answered by the H.323 station. | 111670 | |
| The display at the endpoint of the agent was updated when an agent logged in. The display was not updated when the agent did not receive or originate a call. The display was cleared when the agent logged out. | 111733 | |
| The Communication Manager internal URI object size and string buffer size was increased to accommodate large refer to header. | 111748 | |
| Communication Manager denied a hold request made by OneX Communicator when Direct Media was disabled. | 111785 | |
| Occasionally, there was no talk path or cross talk when a Device Media Call Control softphone was conferenced in a Multiconnect system. | 111808 | |
| Problems were encountered when agents logged in through a CTI application. | 111815 | |
| Incoming international calls, made to a SIP station over an ISDN-PRI or an IP trunk, failed. | 111821 | |
| Directory feature intermittently failed. | 111831 | |
| A call was dropped when PRACK (Provisional Response Acknowledgement), with new SDP, terminated on Communication Manager. | 111853 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| DTMF was not detected on OneX Mobile when a call was established using the OneX call back feature. | 111874 | |
| In a two-way call initially involving a non video and an H.323 video endpoint, an attempt to conference in a SIP video endpoint resulted in the SIP endpoint dropping from the call. | 111877 | |
| Integrated announcement on MG was clipped after every 10 or 15 calls when there were multiple MG's in the same Network Region. | 111883 | |
| An ASAI 3PCC (third party call control) request received a NACK (Negative Acknowledgement) from Communication Manager when an agent requested an aux_work mode change. | 111898 | |
| An ASAI transfer request made to a VDN, with the first step as a collect step, failed. | 111901 | |
| On duplicated Communication Manager servers, the request for a Communication Manager restart on the server going active did not complete when a server interchange occurred during a narrow window and the number of port network control connections to the new Standby server was better than the number of port network control connections to the new Active server. The Communication Manager software on both the servers remained in the Standby mode and Communication Manager did not process any calls. Manual intervention was required to run a command and restart Communication Manager or request for a server interchange. | 111907 | |
| A call was made from an H.323 phone on Communication Manager 1 to an MMCS configured on Communication Manager 2. MMCS sent plain REFER message to Communication Manager 1 to transfer H.323 to the MMCS audio bridge. The transfer was unsuccessful when the display name of H.323 was 14 characters. | 111927 | |
| Supervised transfer from the bridge appearance of a phantom station ended up at the wrong mailbox. | 111933 | |
| The system reset for internal and external calls when the MCT (Malicious Call Trace) button was activated and the trunk administered on the system-parameters features screen for MCT Voice Recorder Trunk Group had Suppress # Outpulsing set to Yes. | 111968 | |
| Customer could not enter data into the Port Location screen and submit it. The system displayed Entry is bad. Occasionally, the Communication Manager application rebooted. | 112006 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| A 3PCC request following an answer failed when it was sent before the ACK to the SIP OOD REFER sequence. | 112010 | |
| Occasionally, SIP phones were unable to resume a held call when the Transfer to Voicemail feature was disabled. | 112011 | |
| A ringback tone was heard in addition to the announcement that was played when a Voice Portal blind transferred a call to a Communication Manager extension that was busy and the call forwarded to a coverage path. | 112012 | |
| Communication Manager did not send 255.255.255.255 in c=line when it received INVITE for hold with c=IN 0.0.0.0 in SDP on an ongoing SIP call. Therefore, hold/unhold failed. | 112021 | |
| OPTIM (Off-PBX-telephone Integration and Mobility) FNE (Feature Name Extension) did not work when SIP Direct Media was enabled on the SIP trunk. | 112026 | |
| The called party could not see the complete calling party name on the OneX Communicator when a value, other than blank, was administered in the **Switch Hook Query Response Timeout** field in the **system parameter features** screen. | 112044 | |
| Occasionally, stack over flow occurred and Communication Manager crashed. | 112052 | |
| Communication Manager reset when administrator tried to bulk unregister SIP stations. | 112053 | |
| Heap Corruption occurred due to Link Bounce. | 112070 | |
| A SIP trunk call was made to a VDN that had an announcement in its vector. The caller did not hear the announcement when the far end was shuffled. | 112092 | |
| An IGAR call made to an audio-group with more than one announcement board failed. | 112107 | |
| An active SIP call dropped after 32 seconds when Communication Manager sent a shuffle re-INVITE due to SA8965 and the far end sent a re-INVITE at the same time causing a glare condition. | 112113 | |
| Communication Manager had certain vulnerabilities that are described in *Avaya Security Advisory ASA-2011-148*. To see this document, go to http://support.avaya.com and search for that number. | 112134 | |
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| Problem | Keywords | Workaround |
| --- | --- | --- |
| Communication Manager went into rolling reboot after reading a translation that had more than 70000 bridge buttons. | 112150 | |
| When port network VoIP resources were not available, a shuffled SIP call was completed with one way or no talk path instead of being denied. | 112155 | |
| A call, made using a TSAPI client, was incorrectly cleared upon disconnection when the **Station Tone Forward Disconnect** field on page 9 of **system parameters features** screen was set to Busy. | 112157 | |
| The customer was unable to administer new ports like stations, trunks, announcements for Communication Manager when the Unnamed registration feature was enabled. | 112161 | |
| A call made to a VDN or an EAS logged-in agent failed with a Denial Event `5019: HDE StatLock, FAC denied` when the **Station Lock Active** field was set to **yes**. | 112176 | |
| A SIP phone could not get out of the SAC state when the SAC state was out of sync with Communication Manager's SAC state. | 112177 | |
| After an IP phone did a hard reboot,it did not display button labels in unicode even though it had support for the native language. | 112188 | |
| A memory leak was observed when Communication Manager received an INVITE with duplicate payload type in SDP. | 112197 | |
| Tone detection on a switch with multiple PN (Port Network) connected with fiber failed when the trunk and the tone receiver took resources from different PNs. | 112205 | |
| Video calls completed with only audio. | 112217 | |
| Display in call forward by rerouting scenario did not work correctly | 112236 | |
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| **Problem** | **Keywords** | **Workaround** |
|---|---|---|
| A PSTN call was made to VDN-1 that had **Delay PSTN Connect Message** set to **Y**. Vector processing of VDN-1 routed the call to VDN-2 that happened in one of the following ways:<br>1. A route to VDN-2 step mentioned in the VDN-1 vector.<br>2. A call routed to a hunt/agent via vector 1 and transferred/ conference to VDN-2.<br>3. A call routed to a station/hunt/agent via vector 1 and covered to VDN-2.<br>The VDN-2 **Delay PSTN Connect Message** field did not matter in this case. VDN-2 routed the call to a non-hunt party, like a station. The station answered the call. It was expected that the CONNECT message would be sent from Communication Manager to the PSTN. | 112260 | |
| The display on the bridge appearance of SIP stations had unwanted double quotes when a trunk call was made to the principal station. | 112273 | |
| For a bridged appearence that dropped out of a conference that involved the originator and the principal station, the display was blanked. | 112275 | |
| Occasionally, Communication Manager reset, disrupting the call service. | 112277 | |
| Occasionally, an AES connected to a duplicated Communication Manager system with multiple CLAN connections did not go into the proper preserved session state upon a Communication Manager server interchange. Hence AES did not always recover properly. | 112283 | |
| When EC500 was activated or deactivated on the primary station, it was also activated or deactivated on the bridge station. | 112307 | |
| Application entries were missing from the stations with off-pbx-telephone integration screen. Due to this, applications associated with stations became inoperable. | 112309,<br>112479. | |
| A caller dialed the meet-me extension and conferenced in another phone that dropped the call. The display was not updated at the endpoint of the first caller when another caller joined the meet-me conference. | 112323 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| For SIP stations, Communication Manager sent update display to principal and bridged stations irrespective of the field value of **Keep Bridged Information on Multiline Displays During Calls** on page 19 of the **system-parameters features** screen. | 112329 | |
| A user tried to bridge in a call that was put on hold by another bridge phone. The exclusion feature was on before the user put the call on hold. The call failed irrespective of the exclusion button being on or off. | 112346 | |
| In a Call Center Elite non-EAS configuration with CMS R13.1 or later, UCIDs were occasionally reported during periodic audit of agent states for unstaffed agents. | 112359 | |
| Incoming and missed call logs were not seen on the OneX client. | 112394 | |
| Default behavior was changed when DTMF payload type was not configured on trunk. | 112405 | |
| Software added in the previous Communication Manager release verified that Communication Manager went active after a server interchange. This code introduced a bug in the procedure used to upgrade software on a system with duplicated servers. A demand server interchange, requested from the Standby Server after the Standby Server was updated, was denied. This fix corrected the problem. | 112408 | |
| SIP station A called Station B and the automatic exclusion feature was enabled. Station C, that had the bridge appearance button of B and an exclusion button, answered the call. The exclusion button on C was lit. The call was put on hold at Station C, putting out the lamp of the exclusion button. The Exclusion button was lit again when the call was resumed on Station C. After putting the call on hold again, it was observed that the lamp of the Exclusion button at Station C was always lit, irrespective of holding or unholding the call. | 112428 | |
| A call did not terminate on far end Communication Manager when the coverage remote had pauses between the digits to ESIG over a SIP trunk. | 112466 | |
| The display on the bridged appearance was not updated after the originator transferred the call to another party. | 112467 | |
| The conferee could be dropped out of the conference call in an hour after the call was established. | 112469 | |
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| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager received a "third party clear call" from CTI for an active call to an agent. Right after Communication Manager acknowledged the cleared call, CTI started a new call to the same agent using "third party auto dial." The second call was routed to an announcement, or anywhere else when the agent endpoint sent an on-hook message in response to the request to drop the first call. This caused the second call to drop. This happened when Station Tone Forward Disconnect was set to busy or intercept. | 112495 | |
| ASAI redirection failed when a call was redirected remotely. | 112508 | |
| The SIP trunk got stuck when an ISDN caller ended a call before the transferred-to party over the SIP trunk answered. | 112519 | |
| A call made to the principal station could ring forever on the bridge phones when the SAC on the principal station was enabled. | 112535 | |
| Ignore was changed to call stndseq for CP_TAKE_CTL and CP_REL_CTL. The stndseq return code is SEQ_SUSP. The old ignore code returned SEQ_FREE. This caused call records to hang. | 112589 | |
| A dial-out call made from a Polycom SIP RMX to a Polycom SIP HDX failed. | 112606 | |
| The final bandwidth allocation was random when the MultiMedia bandwidth in the ip-codec-set was set to a value greater than ~3.2 MBps. This resulted in poor quality video calls. | 112724 | |
| Long SIP extension numbers failed to register. | 112791 | |
| A call that originated from a bridged station was not notified to the other bridges of the principal station. | 112833 | |
| Server interchange caused traps in SIP trunked systems. | 112841 | |
| An ASAI 3PCC request to transfer a call to a VDN failed when the first step was a vector step. | 112933 | |
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# Problems fixed in Communication Manager 6.0.1 SP7

This release includes the following fixes delivered to Communication Manager 6.0.1 SP7.

**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP7. | 112402, 112403, 112629, 112909, 112921, 120154, 120243. | |
| An H.323 softphone put a call on hold. The softphone could not reconnect to the call by running the **`3PCC reconnect`** command when the H.323 SIP override feature was activated. | 100875 | |
| Customers could not see the ARS digit conversion tables and the system displayed the EECCR message. | 110940 | |
| Customers were unable to use MM714 Analog Media Module to administer loudspeaker paging. | 102627 | |
| A board removed alarm was active on a board. The alarm stopped working for 10 minutes when the board was installed. | 102772 | |
| **Communication Manager warning alarm** filter was set to All CM alarms in the **CM Filters** Web page and by executing the **`custalmopt`** command. The Cleared Alarm Notification feature did not work properly when Communication Manager had active warning alarms. Also, the Cleared Alarm Notification feature did not work after toggling between **Send only Minor** (and Minor) **Alarms** and **Send All CM alarms**. | 110010 | |
| On Communication Manager running on System Platform, the command **`setnic`** returned an error message displaying that setnic is not supported on the configuration. | 110462 | |
| Transient conditions such as short network outages caused the system to raise Filesync alarms. The file synchronization succeeded, but the alarms had to be cleared manually. | 111092 | |
| | | ***1 of 10*** |

**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| An incoming QSIG trunk diverted a call to VDN. The VDN routed the call to a station. The call did not cover to the coverage path of the station when the **QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path** field was set to **y** on page 1 of the **system-parameters coverage-forwarding** screen. | 111191,<br>111528. | |
| SIP signaling groups were in the bypass state because the far-end took very long to start functioning. This happened when the far-end tried to start functioning again. Signaling groups on the near-end went into the far-end bypass state when the layer 3 test (test number 1675) failed because the far-end did not respond. This happened when the listen port of the far-end signaling link had shut down. When the far-end re-established the listen socket, it caused the near-end to re-establish. When the near-end received the establish request, it should have been able to analyze whether the associated signaling groups could start functioning quickly. | 111909 | |
| Communication Manager reset when H.323 IP trunks were used in the system. | 112025 | |
| This is a new feature from an ASAI perspective. Earlier the system took any digit sent by call processing and passed it on. However, the feature now removes b* before passing it on. | 112159 | |
| CDR was not generated for calls that were made from the OneX server. | 112163 | |
| Call pickup alert was sent to the station when the **Call Pickup Alerting?** field was disabled on the **system parameters** screen. | 112171 | |
| The transferred device in a Telephony Services API Transferred event was not set when a call was answered by a member of the coverage answer group. | 112244 | |
| Avaya Security Advisory ASA-2011-281 describes certain vulnerabilities of Communication Manager. You can view this document on the Avaya support site. | 112278,<br>112817. | |
| The Dial Plan Transparency call failed on 96xx phones that had Special Application firmware installed. | 112279 | |
| Communication Manager did not prefix location specific international access code correctly for inter-location calls. | 112295 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| An agent logged into a split using the Add Agent Skill FAC. There was no logged in event when the split was monitored. Consequently, there was no log out message when the agent removed a skill. | 112384 | |
| A call was made from EC500 to a station. No Denial Event was observed when both the stations were in different COR with no call permission to each other and had the same **Station Lock COR** for each COR. | 112385 | |
| In the case of blind transfer, an incorrect UCID value was sent in a DOMAIN control message to an ASAI adjunct when the user-to user value in REFER contained a non-UCID value. | 112387 | |
| An agent using a personal computer for voice was on a call. At the same time, the agent received a second call in the ringing state. Under the following conditions, there was no voice path for the first call:<br>● The calling agent or the IP agent was administered with IP-IP Audio Connections? n.<br>● The voice path was bridged through an H.248 controlled media gateway.<br>● The VoIP resource was not cached. | 112400 | |
| A station had EC500 enabled and had logged off. The secondary number assigned for EC500 was busy on another call and the PSTN sent DISC with in-band busy indicator. When a call was made to this station, the caller heard ringback instead of the busy tone. | 112415 | |
| A delay was observed when SIP signaling groups started functioning after a system restart. | 112419 | |
| The Layer 3 test for SIP signaling groups (test number 1675) could get into a state where several signaling groups with the same near-end/far-end IP address and the same near-end/far-end list port were denied from running the test. When this happened, signaling groups might not have gone into or out of service properly because signaling groups were "locked out" from running test 1675. Only one signaling group with a specific near-end/far-end attributes is allowed to run test 1675 at a time, all others are denied when the test is attempted. There seemed to be a condition, perhaps a COLD 2 restart, that could leave one signaling group with the "lock" condition so that none of the signaling groups were able to run the test. | 112439 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| A SIP invite message was sent when an agent tried to log in to a station. The system encountered a segmentation fault when the agent ID was missing from the invite message. | 112453 | |
| A system with no Port Networks had only media gateways and an H.323 station in a Network Region that did not contain any VoIP resource. After the station originated a call, the call failed when the **Extend Call** button was pressed. | 112455 | |
| Customers were unable to use the table numbers 256 through 512 in **if holiday** and **if service-hours** vector steps. | 112458 | |
| The **Active Channels:** field on page 1 of the **status link procr** screen always displayed 0. Page 2 of the screen was always blank when communication-interface processor channel was administered. | 112464 | |
| While upgrading the system from earlier releases, the value of the **Maximum Agent Occupancy Aux Work Reason Code** on the **system-parameters feature-related** screen was incorrect. This caused errors while submitting the **system-parameters feature-related** screen. | 112476 | |
| Communication Manager reset when it tried to send digits to the third party PBX over an IP trunk. | 112496 | |
| Communication Manager incorrectly started the No Answer timeout timer when the **QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path** field on the **list trace station** screen was set to **n**. After the timer timed out, Communication Manager tried to cover the call to coverage path 0 which was an invalid coverage path. | 112503 | |
| Tone Commander BRI terminals aliased as 8520T type terminals received a premature REL_COMP message during incoming calls. Due to this, calls dropped. | 112506 | |
| Occasionally, the system restarted when connected to SIP video endpoints. | 112516 | |
| Prior to this fix, an IP softphone could be administered on the SAT station screen when the internal system limit for station LAN ports was reached. The SAT station screen displayed IP softphone enabled for the station. However, the IP softphone could not register to the station and the shared control of the station failed for AES monitoring applications. With this fix, additional administration validation is implemented. This will protect against administering and enabling IP softphones that exceed the internal system limit for station LAN ports. | 112524 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| There was an increase in the internal Communication Manager capacities to get a station LAN port limit that supports the advertised capacity of station limit and IP softphone limit. Each administered IP station requires 1 station LAN port and any station with IP softphone=y requires 2 additional LAN ports. This required the total number of LAN ports to equal the station limit plus twice the IP softphone limit. With this increase, a problem that triggered resets from the usage of H.323 trunk groups using non-shared signaling in the trunk member range above 14,000 was also fixed. | 112525 | |
| The calling party name was displayed as Unknown when OneX Communicator was in the shared control mode and had Unicode support enabled. | 112548 | |
| The inter-network region connectivity test did not use different media resources to run tests between regions. The test kept using the same resources and generated many software errors when there were 10 or more resources in a region. | 112577 | |
| The inter PBX attendant feature did not route the call to an attendant when an IAS code of more than 7 digits was used in the console-parameters screen. The call could have been routed to the attendant via a vector by using a vector directory number and disabling the IAS code. | 112580 | |
| A station had EC500 enabled and DM enabled on the signaling group. Calls on the station dropped when an internal audit was run. | 112605 | |
| Memory pool errors were logged after Communication Manager rebooted (reset system 4) or restarted (restart system 2). The memory pool errors indicated heap memory problems. | 112613 | |
| An IP agent had a telecommuter over a SIP trunk. The agent did not hear VOA when an incoming call was on a SIP trunk. | 112623 | |
| Communication Manager could not respond to Avaya one-X® Communicator requests to negotiate a lower profile level of H.264 codec in order to reduce CPU utilization during a call. | 112625 | |
| Dial Plan Transparency (DPT) did not work in the Local Survivable Processor (LSP) mode when the **idle appearance select feature name** extension was used on a logged off station. | 112626 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| The **`list trace station/tac`** command did not output short inter-digit timer expiration events. | 112657 | |
| The system displayed the error message `Error encountered, cannot complete request` after executing the **`status port network`** command when the Sync Over IP feature was enabled. | 112661 | |
| SIP subscription refresh requests passed even when there were no channels available on the requested signaling group. | 112667 | |
| VDN return destinations and vectors that loop between each other without limit were defined. This caused the system to deny all ISDN calls when the callers were connected for a long periods of time. | 112671 | |
| A customer could not add an IP media processor (TN2602) IP interface by running the **`add ip-interface`** command on the SAT screen while administering the 129th IP interface. The SAT interface hung due to issues with validation. | 112684 | |
| A call was made over a Register Signaling 2 Multi Frequency Compelled (R2MFC) trunk to a VDN. The calling party number was displayed incorrectly at the SIP station when the VDN routed the call. | 112689 | |
| The analog test call for trunks aborted with a 1901 abort code when the test was run through the SAT interface (**`test analog-testcall`**). | 112692 | |
| Occasionally, a SIP video endpoint dropped from conference calls that were made between an audio-only SIP endpoint, an H.323 video endpoint, and the SIP video endpoint. | 112721 | |
| Two CMS adjuncts (processor channels) shared the same local interface. The failure of 1 CMS connection caused the system to restart after the connection was re-established. When 2 CMS processor channels were administered to use the same interface link number and port number, they shared a single listen socket. At one point, both the channels went down, maybe due to an underlying socket failure caused by a network outage. After some time, one channel recovered when one socket came up, maybe due to partial network restoral. However, the other channel remained idle. The recovered channel stopped functioning again, again due to network outage. When it started functioning again, the system restarted. | 112728 | Use different port numbers for the 2 CMS processor channels. This way, each CMS will have its own listen socket. |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| No video was observed on an audio call made from an audio endpoint to a video endpoint and subsequently conferenced or transferred to other video endpoints. Occasionally, endpoints even dropped from the call. | 112747 | |
| The redirection notification feature, when configured for 5 second updates, could prevent a disconnected H.323 endpoint from being placed out of service by the aging process. | 112749 | |
| CDRs were generated with service observer as the originating party. | 112769 | |
| The text in the warning header of 480/483 messages could not be parsed by an endpoint causing the call to fail. | 112778 | |
| An IPv6 address was administered on a survivable server which was also the network region backup server. This caused the system to reset when the server became active. | 112779 | |
| IP endpoints failed to register. Denial event 1926 IP RRJ-Authenticatn failed. Also, proc error 7171 63947 was generated randomly for the same extension. | 112787 | |
| Users were unable to make calls when the internal call records for some SIP calls were exhausted. | 112797 | |
| Occasionally, Avaya Performance Center (APC) reported UNKNOWN for the state of ACD calls. | 112804 | |
| Calls made to the service link were not placed when eConsole and OneXAttd were registered in the telecommuter mode. Instead, the media stream went to the soft client. | 112808 | |
| The system reset when a SIP endpoint administered the maximum number of buttons. | 112811 | |
| Changing the case of a letter from lower to upper or vice-versa in a node-name on the Node Names screen caused question marks to appear on screens that referenced the node-name that was changed. | 112827 | |
| The system restarted when a large number of PROCR sockets for H.323 stations closed simultaneously. | 112855 | |
| In a 3 party conference call involving the principal station, the bridging station and the originator, the display on the principal station went blank after it was dropped out. | 112872 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| SIP calls, made to an agent with a SIP service link, lost talk path when the agent activated one step recording by pressing the audix-rec button. | 112873 | |
| Some `ossi` terminal type combinations of the SAT interface did not work correctly. | 112879 | |
| Any time a DS1C board was translated in the Center Stage fiber arrangement, a system Warm start would occur as soon as an attempt was made to insert the DS1C board. Multiple Warm starts could lead to a CM reboot and multiple reboots could cause the server to reboot. | 112910 | |
| On a Direct Agent call, the Call Center workmode button lamps flickered and stopped glowing when the agent answered the call on a station that had no auto-in or manual-in buttons. | 112919 | |
| There was no coverage for incoming QSIG and SIP diverted calls to vectors that had a route to step with coverage to an extension. | 112934 | |
| Customers using Call Center releases prior to 6.0 and CMS releases prior to R16.1 were unable to administer new or existing policy routing tables. | 112938 | |
| Occasionally, Communication Manager reset when video endpoints changed capabilities midway in a call. | 112950 | |
| Logs were flooded with error messages while making video calls. Excessive log entries reduced performance and obscured the essential information in the logs. | 112953 | |
| An ASAI 3PCC request made to transfer a call to a VDN failed when the first step was a collect step. | 112956 | |
| A SIP-RMX dial-out call made to SIP-HDX failed. | 112960 | |
| On Communication Manager, Multiple Level Precedence & Preemption (MLPP) was enabled and shuffling was disabled. An H.323 phone called a SIP phone. After the SIP phone answered, two way talk path was observed. However, ring back did not stop at the H.323 phone. | 113010 | |
| An auto-answer agent logged out of the system and logged back in with the headset and speakerphone turned off, and the handset in the cradle. If the Block Hangup feature was activated the agent could receive auto-delivered calls. In this case the agent would not be alerted to the call and the caller would hear dead air. | 113030 | |
| | | ***8 of 10*** |

**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| Rebuild processes froze and were not completed after IP synchronization was enabled. | 113049 | |
| A SIP station could not deactivate the call-fwd or cfwd-bsyda button after Communication Manager was restarted. | 113081 | |
| No video was observed on ADVD transfer calls and OneX Communicator displayed a blank video window on an audio call to a SIP OneX Communicator. | 113160 | |
| Coverage on don't answer was set on the principal station. A call that was transferred to this station traversed its coverage path even when it was answered on its bridge appearance. | 113175 | |
| No video was observed when an H.323 HDX that was registered to Polycom CMA called an Avaya 1000 series video endpoint. | 113191 | |
| On calls made between H.323 and SIP endpoints, H.323 endpoint received no audio when a Siren audio codec was chosen. | 113194 | |
| A call made from a SIP endpoint to a H.323 OneX Communicator dropped when the One-X Communicator closed the video window. | 113222 | |
| A conference call was made between an audio phone, an HDX and another video endpoint. The call dropped when the video endpoint hung up leaving the audio-only endpoint and the HDX. | 113231 | |
| A conference call was made between an audio-only SIP endpoint and 2 video endpoints, ADVD and SIP OneX Communicator. There was no audio when one of the two video endpoints hung up. | 113232 | |
| Communication Manager could reset due to a software segmentation fault. | 120016 | |
| A call was made from a SIP video endpoint to an ADVD. Only audio was observed on the call when the ADVD performed a blind transfer. | 120025 | |
| A call was made from a 10x0 video endpoint to an ADVD. One way video was observed on the call when the ADVD blind transfered the call to an HDX. | 120050 | |
| Communication Manager reset in systems that use H.323 video when **video debug prints** was enabled. | 120080 | |
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**Table 19: Fixes delivered to Communication Manager 6.0.1 SP7**

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, Communication Manager with active video calls reset. | 120087 | |
| Incoming PSTN call made to an x-ported station could not be answered on its bridged appearence. | 120204 | |
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# Problems fixed in Communication Manager 6.0.1 SP8

This release includes the following fixes delivered to Communication Manager 6.0.1 SP8.

**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP8. | 120743,<br>111234,<br>120760. | |
| A race condition in the SAT process caused problems for programs that used the OSSI interface to Communication Manager, such as LoadAgent. | 102799 | |
| Occasionally, Communication Manager was unable to route an incoming call over an R2MFC trunk to an outgoing ISDN PRI trunk. | 110369 | |
| Occasionally, a segmentation fault occurred on Communication Manager when a SIP INVITE message was sent causing the system to reset. | 110460 | |
| An incoming SIP call dropped when the **Direct Media** field was set to ON, and the INVITE message contained an SDP with c=0.0.0.0 and a valid port. | 110837 | |
| The inter-gateway connection that was established to provide synchronization between media gateways was torn down after a link bounced. | 111922 | |
| An outbound call made by a SIP station to Modular Messaging via Session Manager failed when the incoming and outgoing SIP trunks had different transport types in Communication Manager. | 112020 | |
| Occasionally, a SIP Direct Media call caused a system restart. | 112067 | |
| DTMFs were not sent when a call was made to a SIP station and initial IP-IP Direct Media was enabled. | 112129 | |
| Calls generated by ASAI and transfered to an ASAI-generated call that was waiting in a queue and was on HOLD were reported to CMS as abandoned while on HOLD. These calls were not counted as connected when the queued call was delivered. | 112220 | |
| | | ***1 of 12*** |

**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| An agent had EC500 enabled. When the agent received an ACD call, reporting recorded the call as interflowed. | 112256 | |
| On an incoming SIP trunk call that was tandemed over an ISDN or QSIG trunk, Communication Manager replaced the prefix + in the calling party number in PAI header with B* in the outgoing setup message. The calling party numbering format was also incorrect. | 112330 | |
| Occasionally, QSIG Path Replacement did not work after an interchange of duplicated Communication Manager servers. | 112343 | |
| Previously, a video call would drop when one of the EPTs sent H263++ codec in the SDP (Offer/Answer) with no profile information in the codec attributes. Processing this codec information resulted in segmentation fault and call failure. | 112430 | |
| Occasionally, Communication Manager reset. | 112545 | |
| Certain calls between Communication Manager and Communication Server 1000 failed if the H.323 trunks between the two used different codecs but the same encryption. | 112556 | |
| The first coverage point was a station with EC500 and the second coverage point was cov-ans grp which had the above station. Calls terminated on the first coverage point and were denied. | 112596 | |
| After daily maintenance was performed, all SIP calls could fail on a system running moderate call traffic over SIP trunks. | 112611 | |
| Occasionally, calls dropped when they were delivered to the OneX Attendant in the telecommuter mode with a permanent service link. | 112645 | |
| Audit dropped an active call on a DECT station when one of its bridges answered the call. | 112655 | |
| A SIP phone was used to make a call to Voice Portal. Voice Portal transferred the call to a VDN. At the VDN, the vector began with an announcement step, and the announcement was not heard. | 112685 | |
| A call redirected to voicemail over a SIP trunk was reported as abandoned by CMS when the caller pressed zero to speak to an operator. | 112723 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| A call was made over an H.323 trunk. The caller did not hear music on hold when the trunk was on a network region that was not connected to the network region of the port network on which the audio source was administered. | 112752 | |
| A caller heard truncated announcement when an unanswered call was forwarded to voice mail. | 112761 | |
| When a call was made to an agent with skill level 5 and DAC (Direct Agent Calling) enabled in the COR screen, ringback was not heard at the calling station. | 112820 | |
| Occasionally, a data record was orphaned in the BCMS/ VuStats tables. This record showed up as a call in queue on monitor or list BCMS reports when the call was not in queue for any hunt group. | 112823 | |
| Occasionally, customized labels of buttons on the button module were deleted with the change of station type. | 112839 | |
| Board translations could get removed on an H.248 Media Gateway link bounce. This would cause corruption if there were port translations associated with the board. This occurred if the MG uplinked an indication that a board was having trouble coming out of reset initialization, so the board would get partially inserted as an Unknown (WAN) type of board. | 112935 | |
| When a large number of long duration SIP calls were made, the system ran out of memory and crashed due to memory leaks. | 112951 | |
| An ASAI 3PCC transfer left the transferring SIP endpoint stuck in alerting or on a dead call. | 112954 | |
| Intermittently, calls that routed to agents had music added to the call when they zeroed out of voicemail. | 112973 | |
| Misadministration of UDP AAR tables resulted in routing loop between Communication Manager and Session Manager. This consumed all the administered trunks between them. | 112978 | |
| The bridged party was alerted with an audible ring even when **Bridged Call Alerting** was disabled. | 112979 | |
| Occasionally, Communication Manager reset. | 113009 | |
| When UUI (User to User Information) was not sent in the format of Special Application 8481 (SA8481) during a third party call, a segmentation fault was observed. | 113025 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| Occasionally, Communication Manager could reset during a call preserving upgrade. | 113033 | |
| The synchronization timing of a media gateway could be set to VOIP when the Synchronization Over IP feature was off. Also, the CLI synchronization administration commands could not be executed because the administration control was in Communication Manager. | 113050 | |
| Occasionally, Communication Manager logs filled up with unnecessary POTENTIAL FOR CROSSTALK DETECTED messages. | 113057 | |
| Occasionally, vector processing could stop causing calls not to complete to agents or attendants. | 113059 | |
| Signaling made to an IP endpoint was momentarily lost when the endpoint was active on a call. It was possible that the signaling channel would not recover. | 113079 | |
| A call made to an agent was redirected to the Audix voice mail through VDN when the agent did not answer. A generic greeting was heard instead of the agent’s greeting. | 113132 | |
| Communication Manager was unable to handle the SIP 302 Moved message on the second route pattern preference. This prevented direct calls and coverage calls to a third party voice mail system from completing if the primary Session Manager link was down. | 113135 | |
| Occasionally, there was a Communication Manager reset during call clearing when an audit was run. | 113172 | |
| Communication Manager reset when a SIP trunk call got forked downstream. | 113178 | |
| A call, that traversed over a QSIG trunk and a SIP trunk, and then transferred to the display on the destination station, displayed the trunk name and the access code instead of the calling party information. | 113192 | |
| On Communication Manager, an error issued by an H.248 media gateway for a particular port on a call caused the call to drop. | 113193 | |
| Call Centers, using Business Advocate with agents who have a mix of skills with and without Dynamic Queue Position, experienced large delays in handling calls queued to skills with Dynamic Queue Position. | 113220 | |

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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| DTMF tone was not played on a G700 media gateway even when the VoIP and media gateway firmware supported in-band DTMF. | 113259 | |
| All active calls were dropped when the Voice and Network Statistics feature was enabled and there was a server interchange or a system restart. | 113260 | |
| Talk path was lost between stations after two successive interchanges of media resources in a duplex media processor configuration. | 113270 | |
| Memory corruption occurred in the connection manager call processing process. Three complimentary data relation audits discovered the corruption and attempted the necessary recovery actions. Only two of the three audits successfully completed the necessary actions. The third audit aborted without providing the necessary recovery. The problem was visible on the status audits cumulative screen, where the INST-LNK audit abort count increased with each audit cycle and the PLIP-LNK audit and UPUSR-LNK audit showed one cycle where data was fixed. The recovery actions of the PLIP-LNK audit and the UPUSR-LNK audit left a port-network in the non-functional state, causing phones to unregister. The system required at least a reset system 2 to recover. | 120022 | |
| When the Terminal Translation Initialization (TTI) feature had associated a phone with a display, the display would not clear. | 120048 | |
| Calls to an unregistered SIP phone went to coverage before they could be answered by the associated One-X Mobile phone. | 120059 | |
| Announcements configured on AUX trunk boards stopped playing after an internal announcement audit was run. | 120064 | |
| SIP calls were dropped when the far end sent comma-separated diversion headers. | 120081 | |
| An incoming SIP trunk call to Communication Manager that originally covered from Microsoft UM voicemail through the find-me feature was not transferred over ISDN to a cell phone when the ISDN trunk did not send the called party number. | 120120 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| A telephone remained in the Discovering mode when an incorrect extension was typed in login field and (SA8904) - Location Based Call Type Analysis feature was enabled. | 120122 | |
| A user heard busy tone and had talk path simultaneously when a call covered to a coverage answer group that had an unregistered SIP endpoint. | 120161 | |
| On a SIP station, the outgoing call that required the authorization code was dropped when another incoming call came in at the second line appearance. | 120167 | |
| One-way talkpath was observed on an H.323 trunk call when the calling IP station used non-G.726 codec and the H.323 trunk side used G.726 codec. | 120187 | |
| An incoming SIP trunk call made to a VDN with the corresponding vector that had an announcement step followed by a collect step failed when shuffling was enabled. | 120190 | |
| On Communication Manager with H.248 media gateways and ephemeral caching enabled, traffic conditions caused Communication Manager to attempt to allocate more VoIP resources from H.248 media gateways than could be supported. Once a H.248 media gateway reported that it no longer has VoIP capacity, Communication Manager stopped attempting to use the media gateway for VoIP. Communication Manager waited three minutes before retrying VoIP allocation from the media gateway. Now Communication Manager will retry VoIP allocation as soon as an ephemeral has been cached or VoIP is released from an active call. | 120201 | |
| Special Application SA8891 caused a memory leak. | 120203 | |
| Multiple transfers of an Avaya 1000 Series video endpoint could result in lost video. | 120209 | |
| Firewall OK alarms were needlessly sent every hour. Now, the Firewall OK alarm is only sent once after a firewall alarm is resolved. | 120212 | |
| Occasionally, the SAT **`list ip-interface`** commands got into an endless loop. This resulted in a high occupancy condition. | 120238 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| Customers could not add the IP Interfaces screen when the Critical Reliable Bearer field was set to y. This happened due to an issue with port network validation that was incorrectly displaying the following error message: `Boards must reside in the same port network` | 120242 | |
| Occasionally, calls could not be made on SIP and H.323 trunks when the trunk audit could not recover stuck trunks. | 120251 | |
| A SIP trunk call made to a VDN that had music, announcements, and collect digits steps failed when the **Prefer use of G.711 by Music Sources?** field was set to **y** and the **Prefer use of G.711 by IP Endpoints Listening to Music?** field was set to **y** on page 3 of the **system-parameters ip-options** screen, and the announcement and the music source were on different media gateways. | 120260 | |
| A patch could not be removed. | 120273 | |
| Corrupted hunt group data prevented saving translations. | 120320 | |
| Music On Hold was played on a call when MOH Class Of Restriction was disabled. | 120323 | |
| There was no audio when audio dynamic payload types were used. | 120377 | |
| A memory leak was observed when Communication Manager received a SIP INFO message. | 120381 | |
| An IP telephone was registered to the wrong extension when it was changed from an unnamed registration to a named registration. | 120382 | |
| Occasionally, there was no video in video transfers on 10x0 endpoints. | 120383 | |
| H.323 endpoints in RMX conference calls did not transmit audio when Siren or G.722.1 Annex C codecs were chosen. | 120386 | |
| System accounts could be removed by Administrator Accounts SMI Pages. Now, these users are protected. | 120415 | |
| On a SIP station, an outgoing call that required an authorization code was dropped when a call came in at the bridge appearance. | 120424 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| A call dropped when PRACK with a new SDP terminated on Communication Manager. | 120439 | |
| Music on Hold did not play when a call shuffled across port networks. | 120440 | |
| A SIP trunk call was made to a VDN that was routed to an agent. The CDR recorded the agent extension instead of the VDN number even when the **Record VDN** field on **system-parameter cdr** screen was set to **y**. | 120445 | |
| Occasionally, Communication Manager did not allocate memory for IP endpoints. This resulted in call failures and loss of talk-path. | 120451 | |
| Occasionally, a SIP call could cause Communication Manager to restart. | 120453 | |
| An ISDN call was answered by a station. When the call was transferred to another station whose coverage path was set to **all**, a generic greeting was played. | 120455 | |
| A call was made to an IP softphone whose Telecommuter is a SIP trunk. The call did not complete and went to coverage. | 120456 | |
| On Communication Manager, there was no talk-path on a call made to a user with 30 or more bridged appearances. This happened when the user with the bridged-appearance links was connected to a H.248 media gateway, and the bridged-appearance users fanned out to many other H.248 media gateways or port-networks. | 120463 | |
| On Communication Manager configured as a feature server, a blind call transfer among three SIP phones caused the call to drop after the transferred-to party answered the call. | 120464 | |
| The P-Intrinsics and user-to-user headers in the SIP Refer-To header URI was not parsed by Communication Manager. As a result, the Invite message sent out from the Refer message did not include the P-Intrinsics and the user-to-user headers. | 120479 | |
| Occasionally, SIP calls either dropped or one-way audio and video was observed on them. | 120485 | |
| On Communication Manager, a 64-party group-page call that used one H.248 media gateway for all parties caused the link to the H.248 media gateway to stop working. | 120492 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| IQ reports did not have correct data on incoming and outgoing non-ACD calls when agents were defined with their first measured skill that was not externally measured. | 120493 | |
| Calls dropped when the party involved in the call was a SIP station or a SIP trunk. | 120494 | |
| The Russian SOSM application (SA8475) was monitoring a station that was part of a forwarding chain. No term event was sent when call processing attempts to terminate to that station unless the station was the principle terminating point. | 120515 | |
| Occasionally, Communication Manager reset. | 120521 | |
| Communication Manager has certain vulnerabilities described in Avaya Security Advisory ASA-2012-127. To see this document, go to http://support.avaya.com and search for that number. | 120550 | |
| Occasionally, the system crashed due to a memory leak that occurred after the equivalent of 10,000 Busy Hour Call Rate of SIP audio calls steady for 3 days or 10,000 Busy Hour Call Rate of SIP video calls steady for 1.5 days. | 120556 | |
| Service observed calls that were made over R2MFC trunks dropped when they were put on hold. | 120578 | |
| On a video conference on an RMX, a video endpoint put the call on hold and then unheld it. The endpoint did not establish video and was dropped from the call. | 120588 | |
| Communication Manager reset when the signaling protocol for a SIP trunk call involved provisional reliable responses. | 120591 | |
| Occasionally, there was no audio path on endpoints over VPN using SIP service links. | 120610 | |
| Call-log information was displayed incorrectly on the principal station for calls that were answered by another station using a call pickup or team button. | 120618 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| On Communication Manager (main server or ESS or LSP), VoIP resources were reserved for longer than the standard period when H.248 media gateway registered with a Communication Manager server after loss of communication. The loss of communication for the media gateway and the Communication Manager server was long enough to force reconstruction of existing calls, that is the ESS and LSP was reconstructing calls for the first time (failover from main), and the main server regained communication with the media gateway after the administered Link-Loss Delay Timer (fallback to main). After reconstruction of calls, the media gateway was unable to report the loss of incoming RTP from a far-end entity (such as an IP trunk or IP station), which tells the Communication Manager server to drop the reconstructed call. This caused Communication Manager and the media gateway to hold onto VoIP resources when they were not needed, thus reducing the capacity to make new calls. | 120639 | |
| The **`list measurements ip voice-stats`** commands stopped running after a cold reboot. | 120674 | |
| A SIP trunk was transferred by a CTI/ASAI application to a VDN, and the VDN waited several seconds before routing the call to an agent. The transferred call produced a significant amount of echo when the system used multiple network regions with multiple media gateways and port networks. | 120675 | |
| Calls that were made to a service-observed VDN with an SSC party connected dropped when the SSC party dropped from the call. | 120685 | |
| When a call was made from a cellphone using the EC500 feature name extension over a QSIG trunk to a station on Communication Manager, the called station did not display the caller name. | 120715 | |
| When calls were made over H.323 trunks, Communication Manager reset. | 120750 | |
| IP signaling groups stopped functioning when a socket closed. The signaling groups did not start functioning even when there were no problems detected. | 120777 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| On Communication Manager, an H.248 media gateway with full VoIP utilization caused trunks assigned to the media gateway region to stop functioning, thus dropping calls in the process. The following conditions apply:<br>● One or more H.248 media gateways in a network region at full VoIP usage<br>● No other VoIP resources used in the H.248 media gateway region, that is no Crossfire boards (TN2602s) or Cruisers (TN2302s)<br>● No other connected region exists in the H.248 media gateway region<br>● Trunks are assigned to use H.248 media gateway region | 120788 | |
| An incoming SIP trunk call failed to detect inbound digits when the **Direct IP-IP Audio Connections** field on the **SIP signaling group** screen was set to **y**. | 120809 | |
| Station A, which was a 96xx SIP station on Communication Manager, called Station B, which was a 96xxSIP station on Communication Manager. Station B had an EC500 number, and the call was answered on the EC500 number. When Station A conferenced in another number, the conferenced party could not be heard on the EC500 endpoint of Station B. | 120813 | |
| Team button updates for the monitoring station were not sent to OneX Communicator when the monitoring station was registered in the shared control mode and the team button was configured on button number 16 or greater. | 120836 | |
| Occasionally, Communication Manager reset. | 120844 | |
| The system displayed the following error message, when a maintenance command was executed from the SAT interface:<br>`All maintenance resources busy; try again later` | 120861 | |
| The Busy Indicator for a phantom extension on a SIP station did not work. | 120885 | |
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**Table 20: Fixes delivered to Communication Manager 6.0.1 SP8**

| Problem | Keywords | Workaround |
|---|---|---|
| When tcm debug logs were turned on, calls were dropped due to system reset. | 121063 | |
| IP phones could not originate calls on a system that only had a single duplicated pair of TN2602 circuit packs (critical reliability) added in a network region. However, the TN2602 pair could be used for inter-gateway communication and call termination. The problems observed were varied and unpredictable and could be masked by the presence of other media processing resources. For example:<br>● The problem was not seen with simplex TN2602 circuit packs in a network region but disabling reliability on the duplicated TN2602 pair did not alleviate the problem.<br>● The presence of additional TN2302 and/or TN2602 circuit packs in the same network region as the duplicated TN2602s may or may not have alleviated the problem.<br>● The presence of an H.248 gateway in the same network region would alleviate the problem.<br>● The presence of TN2302s and TN2602s, and H.248 gateways in other network regions also alleviated the problem. | 121094 | |
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# Problems fixed in Communication Manager 6.0.1 SP8.01

This release includes the following fixes delivered to Communication Manager 6.0.1 SP8.01.

**Table 21: Fixes delivered to Communication Manager 6.0.1 SP8.01**

| Problem | Keywords | Workaround |
|---|---|---|
| Due to an infinite loop, Communication Manager was overloaded. This was followed by a system reset. | 121732 | |
| A SIP trunk call was transferred by a CTI/ASAI application to a VDN, and the VDN waited several seconds before routing the call to an agent. The transferred call produced a significant amount of echo when the system used multiple network regions with multiple media gateways and port networks. | 121745 | |
| Occasionally, Communication Manager reset, causing service disruption. | 121785 | |
| | | |

# Problems fixed in Communication Manager 6.0.1 SP9

This release includes the following fixes delivered to Communication Manager 6.0.1 SP9.

**Table 22: Fixes delivered to Communication Manager 6.0.1 SP9**

| Problem | Keywords | Workaround |
|---|---|---|
| Issues associated with the following keywords were also corrected in Communication Manager 6.0.1 SP9. | 120884,<br>121384,<br>121860,<br>121908. | |
| Occasionally, there was a segmentation fault in the SIP stack. | 110562 | |
| When customers attempted to view MTA data from the **System Logs SMI** page, there were underlying resource issues blocking the request. The SMI page reported success even when it was not successful. Also, the system did not display any data. | 110979 | |
| Occasionally, there was instability in the sychronization of the media gateways when the media gateways were disabled and enabled for the Sync Over IP feature. | 111409 | |
| Resolution of a problem with synchronization over IP for a media gateway caused a segmentation fault, and Communication Manager restarted. | 112982 | |
| When call-appr or brdg-appr button was used on an expansion module, an incoming call to the call-appr/ brdg-appr had the avaya-cm-line field set wrong in the Accept-Contact Header in the INVITE message. | 112986 | |
| Workmode change from ASAI was performed immediately even when the agent had put a call on hold. | 113100 | |
| ASAI redirection to the EC500 station over ISDN trunks failed. | 113104 | |
| An EC500-initiated call failed to route over a trunk when the overlap trunk setting was used. | 113106 | |
| An ASAI application could not drop an announcement party from a call by using a selective drop request. | 113203 | |
| Due to delays in the receipt of STFTPHN_OFFHK messages, a TONE_ON message was not sent to the station. This caused problems with logging in an IP agent. | 120052 | |
| | | ***1 of 6*** |

| Problem | Keywords | Workaround |
|---|---|---|
| A SIP trunk was configured to use special application SA8965. An outbound call over the trunk to a PSTN endpoint that covered to voicemail resulted in one way talk path. The caller could not hear the voice mail announcements but was able to leave a message. This happened due to a SIP INVITE glare condition between Communication Manager and the SIP service provider. | 120136 | |
| Occasionally, outgoing calls were denied over an H.323 trunk when the originator pressed a digit before the call was answered by the far end. | 120361 | |
| Occasionally, Communication Manager incorrectly displayed errors for Port Network and media gateway media processors during an audit. | 120546,<br>120761,<br>120950. | |
| Under the Synchronization Over IP feature, administration of a reference board for a tandem clock left some media gateways unsynchronized. | 120558 | |
| Call transfer failed when an attendant on the CAS-Main transfered an on-going call between CAS-Branch and CAS-Main over an RLT trunk. | 120586 | |
| A call made from an EC500 endpoint failed to route over a trunk when the enbloc trunk setting was used. | 120625 | |
| Agents heard the VDN of Origin announcements delayed by up to two seconds when the resources required to play such an announcement were across port networks and media gateways. | 120627 | |
| When a call covered to messaging and returned over a SIP trunk, the messages that were going to reporting showed the call as abandoned. | 120681 | |
| An IMS user called an xport station that had EC500 Mapping and Terminal Translation Initialization enabled. Communication Manager did not send the call to the cellular phone. | 120768 | |
| An upgrade to Communication Manager 6.0.1SP5, or 6.2 caused degraded software duplication performance and higher processor occupancy. | 120795 | |
| Occasionally, Communication Manager reset while performing an operation related to the EC500 feature. | 120883 | |
| Communication Manager delayed updating the display of a SIP station for an ISDN trunk call from a PSTN. | 120899 | |
| | | ***2 of 6*** |

| Problem | Keywords | Workaround |
|---|---|---|
| When an SMI page attempted to process a dynamic page where the returned data output was large, the amount of memory allocated was exhausted. This caused the page to have the appearance of not responding. | 120924 | |
| Occasionally, Communication Manager reset. | 120927 | |
| When an agent on a Genesys softphone unheld a call the caller heard DTMF tones. | 120958 | |
| There was one-way talk path on a conference call over a SIP trunk when network-call-redirection was activated on the trunk group. | 120965 | |
| Under heavy load, a system failure resulted in a RELOAD of Communication Manager being delayed for seconds when the port networks were not functioning. | 120972 | |
| A generic greeting was heard instead of the greeting of the subscriber when an outgoing SIP call re-routed back to Communication Manager and Communication Manager redirected the call to a Modular Messaging voice-mail server. | 120999 | |
| Occasionally, calls could not be made from SIP phones. | 121020 | |
| Occasionally, a SIP trunk call dropped after a glare condition. | 121045 | |
| During SIP downstream forking, Communication Manager did not send PRACK for a reliable response which could lead to call failures. | 121062 | |
| When there was a call on TDM, there was no talk path when the far end changed key after hold. | 121105 | |
| There were multiple system restarts and a flood of process errors logged against the LIP process due to memory corruption. | 121177 | |
| Vector route-to step with ~ was not processed correctly. This resulted in losing the digits after the ~. | 121192 | |
| The **`status media-processor`** command caused a segmentation fault when there was an error in retrieving the DSP information. | 121193 | |
| Occasionally, a network outage caused the system to reset. | 121273 | |
| Calls that were hairpinned on a TN2602 media processor did not have talkpath due to a race condition internal to TN2602. The timing in Communication Manager has been changed to prevent this race condition. | 121277 | |
| | | ***3 of 6*** |

| Problem | Keywords | Workaround |
|---|---|---|
| Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2012-233. To see this document, go to http://support.avaya.com and search for that number. | 121294 | |
| On Communication Manager, calls made by using IGAR to communicate between legacy port networks and H.248 media gateways did not complete when the trunks used for IGAR had the **Apply Local Ringback?** field set to y. | 121306 | |
| When a system had only IPv6 media resources the system would restart. | 121314 | |
| After multiple transfers, an originating station on an Integral 55 System continued to hear ring back even after the call was answered by a Communication Manager station. | 121324 | |
| A segmentation fault occurred on Communication Manager when there was an ongoing activity on an Enterprise Mobility feature enabled station having bridge appearance on its expansion module. | 121327 | |
| A SIP trunk call made to Communication Manager was routed to Avaya Voice Portal (AVP). AVP answered the call and initiated transfer to H.323 station on Communication Manager. AVP was connected using a SIP trunk. The call dropped immediately after AVP completed the transfer. NCR was enabled for SIP trunk towards AVP. | 121376 | |
| Customer created SMI access profiles were not correctly restored during a Communication Manager template upgrade. | 121387 | |
| CPU occupancy issues were observed while running very large OSSI scripts. In the one known case, the OSSI script was trying to remove 41,000 SIP stations. This caused a server interchange on a Communication Manager Duplex system. | 121415 | |
| Running software that modified large amounts of translations caused high occupancy. | 121427 | |
| The states of the line appearance of a SIP phone and the line appearance of Communication Manager were out of sync after the SIP station failed over from Session Manager 1 to Session Manager 2. | 121435 | |
| Calls made through Voice Portal did not cover to voice mail when the SA8874 CCMS Call Status Messages to 7434ND station was set to ON. | 121440 | |
| | | ***4 of 6*** |

| Problem | Keywords | Workaround |
|---|---|---|
| A translation corruption warning message was displayed while logging into the SAT on an ESS server. | 121450 | |
| Occasionally, dialed digits were out pulsed twice for trunk calls. | 121451 | |
| The tie trunk group failed with denial event 5034 `Invalid MCT trunk group` when it was used for Malicious Call Trace. | 121472 | |
| SIP calls dropped for agents working remotely in the telecommuter mode when the service provider refreshed the SIP call using a reINVITE message. | 121474 | |
| An internal Communication Manager software error caused reset 1 & 2. | 121496 | |
| There was no talkpath on a switched-classified call over ISDN PRI with the **Trunk Hunt** field set to ascend/descend. | 121503 | |
| It was not possible to make a video call when the called party sent SDP/Answer in a reliable provisional response. Only an audio call was established. | 121562 | |
| SIP trunks became inactive after a traffic burst. | 121591 | |
| There was only one-way talk path when an incoming SIP trunk call was put on hold and the unhold operation from the bridge station after sesson refresh INVITE (having a=recvonly and sdp verion changed) was processed. | 121607 | |
| When SAC was enabled on the principle terminating station in a pickup group, all the endpoints of the pick-up group were in the alerting state, that is, the pkup buttons continued to alert, even after the call was covered out of the pick-up group. | 121613 | |
| There was no host name on the outgoing invite message request URI and the To header when the incoming invite message request URI contained escaped characters. | 121626 | |
| When an agent was on a trunk call and the trunk dropped, reporting recorded the call as if the agent hung up the call. | 121636 | |
| Occasionally, Communication Manager reset. | 121689 | |
| Calls made to a non-ACD hunt group terminated to and rang members whose stations were logged out. | 121716 | |
| | | ***5 of 6*** |

| Problem | Keywords | Workaround |
|---|---|---|
| IP signaling groups on an ESS went into the disabled state when the ESS was active controlling port networks and media gateways, and then all the media gateways returned to the main server. | 121718 | |
| Occasionally, CLAN did not accept new registration requests from IP stations. | 121762 | |
| Agents using One X Communicator could not log in to the system. | 121803 | |
| Occasionally, while activating or deactivating a service pack, the **`ldconfig`** command caused a segmentation fault when the server setup steps were running. | 121806 | |
| Occasionally, customers using Service Level Objectives in skills did not receive the Interruptible Aux notifications while using Calls Warning or Time Warning Thresholds. | 121859 | |
| Occasionally, using attendant number 414 caused translation corruption. | 121901 | |
| | | ***6 of 6*** |

# SIP Trunk Capacity Guidelines

The following maximum SIP trunk capacities apply when Communication Manager 6.0 Service Pack #1 or greater is installed on the system:

**Communication Manager Evolution Server Environment.**

- For Call Centers with high levels of call traffic (particularly 24x7 Call Centers): 7,000 SIP trunks.
- For general business use or for non-24x7 Call Centers with moderate call traffic: 12,000 SIP trunks.
- These maximums apply to the Main server in a Communication Manager configuration. Strictly to configure redundant trunks in support of fail-over, an additional 7,000 or 12,000 trunks can be administered on a Survivable Server. These additional trunks *cannot* be used for sunny-day traffic.
- Important: *Any* Communication Manager Evolution Server design (Call Center or general business) with more than 10,000 trunks is *required* to go through Sales Factory review. This trunk capacity is the sum of all trunk types, not just SIP.

**Communication Manager Feature Server Environment.**

- 24,000 SIP trunks for general business use with moderate call traffic. (Call Center Elite configurations are not approved for the Communication Manager Feature Server configuration.)
- Important: *Any* Communication Manager Feature Server design with more than 15,000 trunks (sum of all trunk types - not just SIP) is *required* to go through Sales Factory review.

**** Important **** All Call Center designs should be reviewed by the Sales Factory Design Center. Call Center designs that involve SIP trunking *must* go through the Sales Factory. See the document titled Avaya Aura® Communication Manager and Call Center Release 6.0

SYSTEM SOFTWARE BASED CAPACITES, available at http://support.avaya.com, for further information.

**Note:**

The capacities specified in that document pertain to general business configurations and may not be valid or recommended for Call Center (CC) solutions. Simultaneously achieving the upper bounds for multiple capacities, including SIP trunks, may not be possible for real-world systems. Call rates and the combined effect of other operational aspects of customer implementations are likely to preclude realizing the maximum limits for particular parameters.

# Known problems

This release includes the following known issues in Communication Manager 6.0.1 SP9.

**Table 23: Known problems in Communication Manager 6.0.1 SP9**

| Problem | Keywords | Workaround |
|---|---|---|
| For rotary analog stations, the inter-digit collection timer may expire too fast which prevents dialed calls from completing successfully. The workaround is the only solution to this issue since no Communication Manager Software change is planned. | 101096 | On the **`system-parameters features`** form, page 6, there is a field called, **Short Interdigit Timer** (seconds), which is defaulted to 3 seconds. Increasing this timer can fix this problem. |
| Communication Manager 6.x LSP servers cannot register with pre-Communication Manager 5.2 main servers.<br>If this LSP registers with a Communication Manager 5.1.2 or earlier main server, you may need to enter the serial number of a media gateway in order to allow this LSP to register with the main server. To obtain a media gateway serial number, execute the list media-gateway SAT command on the main server and select one of the media gateway serial numbers displayed. Then configure the LSP with this serial number via the LSP SMI Server Role web page. Note that this works as designed and no fix will be made in Communication Manager Software. | 101016 | |
| Remote access and Telecommuter calls using QSIG over SIP trunks (QSIP) will not complete. These calls are unable to break dial tone to enter the barrier code (remote access call) or the feature access code (telecommuter call). | 100896,<br>112182. | |
| | | ***1 of 3*** |

**Table 23: Known problems in Communication Manager 6.0.1 SP9**

| Problem | Keywords | Workaround |
|---|---|---|
| The active server of a server pair running the Duplex Communication Manager Main/ Survivable Core Template can experience a service outage when System Platform is upgraded or updated on the standby server. | NA | Perform the pre-upgrade step on the active server. Busy out the standby server and upgrade/ update the System Platform. Release the standby server and verify the duplication state. Activate the Communication Manager Software update (service pack) on the standby server and again verify the duplication state. Perform a non-forced interchange of the Communication Manager servers. Busy out the previously active server which is now the standby and upgrade/update the System Platform. Release the standby server and verify the duplication state. Activate the Communication Manager Software update (service pack) on the standby server and again verify the duplication state. |
| | | ***2 of 3*** |

**Table 23: Known problems in Communication Manager 6.0.1 SP9**

| Problem | Keywords | Workaround |
|---|---|---|
| New features or feature options included in Communication Manager service packs are noted in the Enhancements section of the release notes. Often these new features or feature options have new administrative fields. Any changes added to the new administrative fields will be lost if the system is subsequently backed down to an earlier service pack that does not include the new administrative fields. This is the case even if translations that include the changes to the new fields are restored to the system following the activation of the earlier service pack that does not include the new administrative fields. Customers are required to back-up their systems before applying a new service pack so that translations that match the previous administrative fields are available, should the new service pack be removed and the system software restored to its previous state. | NA | |
| To avoid losing service, IP Softphone users should logoff, thereby, restoring their base phone to service prior to deactivating a Communication Manager service pack. | NA | |
| | | ***3 of 3*** |

# Appendix A: Acronyms

| | |
|---|---|
| **3PCC** | Third Party Call Control |
| **AAC** | Avaya Aura Conferencing |
| **AAR** | Automatic Alternate Routing |
| **ACD** | Automatic Call Distribution |
| **ADVD** | Avaya Desktop Video Device |
| **AES** | Application Enablement Services |
| **APC** | Avaya Performance Center |
| **ARS** | Automatic Route Selection |
| **ASA** | Avaya Site Administration |
| **ASAI** | Adjunct Switch Applications Interface |
| **ATB** | All Trunks Busy |
| **ATM** | Asynchronous Transfer Mode |
| **AVP** | Avaya Voice Portal |
| **AWOH** | Administered WithOut Hardware |
| **BA** | Bridge Appearance |
| **BCMS** | Basic Call Management System |
| **BFCP** | Binary Floor Control Protocol |
| **BSR** | Best Service Routing |
| **BRI** | Basic Rate Interface |
| **BTD** | Busy Tone Disconnect |
| **CDR** | Call Detail Record |
| **CIE** | Customer Interaction Express |
| **CLI** | Command Line Interface |
| **CLAN** | TN799 Control LAN circuit pack that controls TCP/IP signalling and firmware downloads |
| **CMA** | Call Management System |
| **CMM** | Communication Manager Messaging |
| **CMS** | Call Management System |
| **CNC** | Control Network C |
| **COR** | Class of Restriction |
| **CPU** | Central Processing Unit |

| | |
|---|---|
| **CPN** | Calling Party Number |
| **CSS** | Center Stage Switch |
| **CTI** | Computer Telephony Integration |
| **DC** | Direct Current |
| **DCP** | Digital Communications Protocol |
| **DCS** | Distributed Communication System |
| **DECT** | Digitally Enhanced Cordless Telecommunications |
| **DMCC** | Device Media and Call Control |
| **DPT** | Dial Plan Transparency |
| **DSP** | Digital Signal Processor |
| **DSCP** | Differentiated Services Code Point |
| **DTMF** | Dual Tone Multi-Frequency |
| **EAS** | Expert Agent Selection |
| **EMU** | Enterprise Mobility Users |
| **ES** | Evolution Server |
| **ESS** | Enterprise Survivable Server |
| **ETSI** | European Telecommunication Standards Institute |
| **FAC** | Feature Access Code |
| **FNE** | Feature Name Extension |
| **FS** | Feature Server |
| **HDX** | A Polycom high definition video room system |
| **HEMU** | Home Enterprise Mobility User |
| **IAC** | International Access Code |
| **IDM** | Initial Direct Media |
| **IGAR** | Inter-Gateway Alternate Routing |
| **IP** | Internet Protocol |
| **IPSI** | Internet Protocol Server Interface |
| **ISDN** | Integrated Services Digital Network |
| **ISG** | Integrated Services Gateway |
| **J24** | Avaya Digital Terminal for Japan |
| **LAN** | Local Area Network |
| **LAI** | Look Ahead Interflow |
| **LAR** | Look Ahead Routing |
| **LED** | Light Emitting Diode |

| | |
|---|---|
| **LSP** | Local Survivable Processor |
| **OPTIM** | Off-Premise Telephony Integration with MultiVantage |
| **MCSNIC** | Mask Calling Number/Station Name for Internal Calls |
| **MCU** | Multipoint Control Unit |
| **MG** | Media Gateway |
| **MGC** | Media Gateway Controller |
| **MIA** | Most Idle Agent |
| **MIB** | Management Information Base |
| **MLDP** | Multi-Location Dial Plan |
| **MLPP** | Multiple Level Precedence Preemption |
| **MOH** | Music on Hold |
| **MPC** | Maintenance Processor Complex |
| **MST** | Message Sequence Trace |
| **MTA** | Message Trace Analysis |
| **MWI** | Message Waiting Indication |
| **NCR** | Network Call Redirection |
| **NIC** | Network Interface Card |
| **NR** | Network Region |
| **OEM** | Original Equipment Manufacturer |
| **OPTIM** | Off-PBX-telephone Integration and Mobility |
| **PAM** | Pluggable Authentication Modules |
| **PBX** | Private Branch eXchange |
| **PE** | Processor Ethernet |
| **PRACK** | Provisional Response Acknowledgement |
| **PROCR** | Processor Ethernet |
| **PSA** | Personal Station Access |
| **PSTN** | Public Switched Telephone Network |
| **PCD** | Packet Control Driver |
| **PCOL** | Personal Central Office Line |
| **PN** | Port Network |
| **PNC** | Port Network Connectivity |
| **QSIG** | International Standard for inter-PBX feature transparency at the Q reference point |
| **R2MFC** | Register Signaling 2 Multi Frequency Compelled |
| **RDTT** | Reliable Data Transport Tool |

| | |
|---|---|
| **RFC** | Request for Comments |
| **RMB** | Remote Maintenance Board |
| **RMX** | A Polycom media conferencing platform, used by CM as a video and audio bridge |
| **ROIF** | Redirect on IP Failure |
| **RTP** | Real-Time Protocol |
| **SAC** | Send All Calls |
| **SAT** | System Access Terminal |
| **SAL** | Secure Access Link |
| **SAMP** | Server Access and Maintenance Processor |
| **SBA** | Simulated Bridge Appearance |
| **SBC** | Separation of Bearer and Signaling |
| **SBS** | Separation of Bearer and Signaling |
| **SES** | SIP Enablement Services |
| **SIP** | Session Initiation Protocol |
| **SDP** | Session Description Protocol |
| **SO** | Service observer |
| **SMI** | System Management Interface |
| **SSC** | Single Step Conference |
| **SSH** | Secure Shell |
| **SSHD** | Secure Shell Daemon |
| **SVNS** | Simple Voice Network Statistics |
| **TAC** | Trunk Access Code |
| **TCP** | Transmission Control Protocol |
| **TDM** | Time Division Multiplex |
| **TSC** | Temporary Signaling Connection |
| **TSP** | Toshiba SIP Phone |
| **TSRA** | Time Slot Record Audit |
| **TTI** | Terminal Translation Initialization |
| **TTS** | Time To Service |
| **UCID** | Universal Call ID |
| **URI** | Uniform Resource Identifier |
| **USNI** | United States Network Interface |
| **USB** | Universal Serial Bus |
| **VALU** | Value-Added |

| | |
|---|---|
| **VCS** | Video Conferencing Server |
| **VDN** | Vector Directory Number |
| **VOA** | VDN of origin Announcement |
| **VoIP** | Voice over Internet Protocol |
| **VEMU** | Visitor Enterprise Mobility User |
| **VLAN** | Virtual Local Area Network |
| **VSX** | A Polycom standard definition video room system |

# Technical Support

Support for Communication Manager is available through Avaya Technical Support.

If you encounter trouble with Communication Manager:

1. Retry the action. Follow the instructions in written or online documentation carefully.
2. Check the documentation that came with your hardware for maintenance or hardware-related problems.
3. Note the sequence of events that led to the problem and the exact messages displayed. Have the Avaya documentation available.
4. If you continue to have a problem, contact Avaya Technical Support by:
   - Logging on to the Avaya Technical Support Web site http://www.avaya.com/support
   - Calling or faxing Avaya Technical Support at one of the telephone numbers in the Support Directory listings on the Avaya support Web site.

   You may be asked to email one or more files to Technical Support for analysis of your application and its environment.

**Note:**
If you have difficulty reaching Avaya Technical Support through the above URL or email address, please go to http://www.avaya.com for further information.

When you request technical support, provide the following information:

- Configuration settings, including Communication Manager configuration and browser settings.
- Usage scenario, including all steps required to reproduce the issue.
- Screenshots, if the issue occurs in the Administration Application, one-X Portal, or one-X Portal Extensions.
- Copies of all logs related to the issue.
- All other information that you gat
- hered when you attempted to resolve the issue.

**Tip:**
Avaya Global Services Escalation Management provides the means to escalate urgent service issues. For more information, see the Escalation Contacts listings on the Avaya Web site.

For information about patches and product updates, see the Avaya Technical Support Web site http://www.avaya.com/support.