Avaya Aura® Communication Manager 6.3

Release Notes

Issue 1
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Contents

Changes delivered to Avaya Aura® Communication Manager 6.3

Communication Manager 6.3 Release Notes

Communication Manager service packs and releases are cumulative, and Communication Manager 6.3 includes the changes delivered to Communication Manager 6.2 SP0, SP1, SP2 and SP2.01, SP3, SP4, and SP5. These changes are grouped as follows:

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 and enter your **Username** and **Password** and click **LOG IN**.
2. Click **DOWNLOADS & DOCUMENTS** at the top of the page.
3. Begin to type **Communication Manager** into the **Enter Your Product Here** box and when **Avaya Aura® Communication Manager** appears as a selection below, select it.
4. Select **6.3.x** from the **Choose Release** pull-down menu to the right. Some PSNs are also found under the **Don't Know** release choice.
5. Check the box for **Product Support Notices** in the content filter to display the available PSN documents.
6. 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 and enter your **Username** and **Password** and click **LOG IN**.
2. Click **DOWNLOADS & DOCUMENTS** at the top of the page.
3. Begin to type **Messaging** in the **Enter Your Product Here** box and when **Avaya Aura® Communication Manager Messaging** appears as a selection below, select it.
4. Select **6.3.x** from the **Choose Release** pull-down menu to the right.
5. Click **View downloads** if necessary.
6. Available downloads for Communication Manager Messaging are displayed. Click the links to see the details.

Communication Manager Software

Communication Manager 6.3 software includes certain third party components including Open Source Software. Open Source Software licenses are included in the Avaya Aura® 6.3 Communication Manager Solution Templates DVD. To view the licenses:

1. Insert the Avaya Aura® 6.3 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 Branch Gateway, Communication Manager, Installation Wizard, Session Manager, and Utility Services 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 and enter your **Username** and **Password** and click **LOG IN**.
2. Click **DOWNLOADS & DOCUMENTS** at the top of the page.
3. Begin to type **Session** in the **Enter Your Product Here** box and when Avaya Aura® Session Manager appears as a selection below, select it.
4. Select **6.3.x** from the **Choose Release** pull-down menu to the right.
5. Click **View downloads** if necessary.
6. Available downloads for Session Manager are displayed. Click the links to see details.

Avaya Video Conferencing Solutions

Communication Manager 6.3 support for Avaya Video Conferencing Solutions including Radvision SCOPIA is documented in the Avaya Aura® Communication Manager SW and Firmware Compatibility Matrix and the Compatibility Matrix tool, both of which are available on http://support.avaya.com. Fixes and known issues for Avaya Video Conferencing Solutions including Radvision SCOPIA are included in the Communication Manager release notes.

System Platform

Communication Manager 6.x Releases and Service Packs are tested with specific versions and updates of System Platform 6.x. For more information, see Communication Manager Software & Firmware Compatibility Matrix at http://support.avaya.com or the appropriate Communication Manager Product Correction Notices.

Enhancements delivered to Communication Manager 6.3

New features and significant enhancements in Communication Manager 6.3 are described in the document titled *What's New in Avaya Aura® Release 6.2 Feature Pack 2* which can be found at http://support.avaya.com. The following changes that are new to Communication Manager are also included in this release.

Enhancements delivered to Communication Manager 6.3 and 6.2 SP1

Table 1: Enhancements delivered to Communication Manager 6.3 and 6.2 SP1 ***1 of 2***

Enhancement	Keywords	Workaround
This change introduces a new Communication Manager log for kernel events. Previously, kernel events were recorded in the /var/log/security log and the /var/ log/messages log. Occasionally, on a non System Platform Communication Manager system, the logging of a large number of kernel events caused the size of the security log to grow very large before the log could be rotated. When that happened, the vi editor and the logv tool might not display security events from these logs.	100788	
A new field **Identify Calling Party Location in INVITE** is added to page 4 of the **Trunk Group** screen. The default value is **n**. When the value of the field is set to **n** Communication Manager sends the INVITE message with a new Via header included. When the value of the field is set to **y**, Communication Manager includes an IP address that identifies the location of the calling party in the bottom-most Via header in the INVITE message. The IP address may be of one of the following: ● H.248 Media Gateway ● MG address - G650 Port Network ● Medpro (TN2302 Circuit Pack) ● C-Lan (TN799 Circuit Pack) ● IPSI (TN2312 Circuit Pack)	112659	

Table 1: Enhancements delivered to Communication Manager 6.3 and 6.2 SP1 ***2 of 2***

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 a new Message Tracer release (6.4.3.9) that includes support for new added Internal Call Process fields.	120502	

Enhancements delivered to Communication Manager 6.3 and 6.2 SP2

Table 2: Enhancements delivered to Communication Manager 6.3 and 6.2 SP2

Enhancement	Keywords	Workaround
The video enabled station limit has increased from 18000 (H.323 station limit) to 41000 (SIP station limit).	112983	
The upper limit of valid user IDs for the Communication Manager SMI was increased from 65535 to 2000000000.	120997	
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 delivered to Communication Manager 6.3 and 6.2 SP4

Table 3: Enhancements delivered to Communication Manager 6.3 and 6.2 SP4 ***1 of 2***

Enhancement	Keywords	Workaround
Initially, for the OneX application, barge-in tone was disabled by design. With this package, the barge-in tone has been enabled by default.	121111	
Increase Timer to Support Voice Quality Test 100 Added the **Timer** field for Terminating Trunk Transmission Test "Test Type 100" on page 2 of the **System-Parameters Maintenance** screen that allows the SAT user to enter the number of seconds between 65 and 999. This is the time the test 100 test call is left with. The terminating trunk transmission test "Test Type 100" times out after 65 seconds, but it is necessary that the call be left up for at least 5 minutes when the test is being run on an analog CO trunk in a Media Gateway.	121412	

Table 3: Enhancements delivered to Communication Manager 6.3 and 6.2 SP4 ***2 of 2***

Enhancement	Keywords	Workaround
Previously, when there was a collect step following an adjunct route step, the collect step killed the adjunct route. This was working as designed. Now the collect step will not kill the adjunct route, as per new FCC mandated rules.	121534	
During Session Manager fail over, a held line appearance on a SIP phone could become stuck. Now an on-hook fnu invite message will be sent to Communication Manager to clear the held line appearance.	121956	
This is a new MTA release 6.4.4.4. This release of MTA includes parsing support for the following: 1. Multithreading Support (mt110216) 2. Parsing of large MST messages(mt120017) 3. New Capro fields, CRD fields and Denial Events (mt120015) The decoding of above changes is not supported by earlier Message Tracer release.	122177	

Enhancements delivered to Communication Manager 6.3 and 6.2 SP5

Table 4: Enhancements delivered to Communication Manager 6.3 and 6.2 SP5

Enhancement	Keywords	Workaround
New inline errors from MM711 and MM716 boards have been fixed to inform Communication Manager that there are over current and over heating problems with individual ports on the board. When the uplink is sent, the board removes power from the port until the detected over-current or over-heating problem is resolved. Communication Manager must know of this so that it can place the port in an out-of-service state and log errors and an alarm.	121963	
Message Tracer Analyzer which includes the parsing support for the new Call Processing, CRD and Denial Event additions.	122801	

Enhancements delivered to Communication Manager 6.3

Table 5: Enhancements delivered to Communication Manager 6.3 ***1 of 4***

Enhancement	Keywords	Workaround
Limits for user sessions and user processes were set for increased security.	111318	
Options for idle login timeouts are added to the **Login Account Policy SMI** page.	111319	
System control settings are altered to increase network security.	111322	

Table 5: Enhancements delivered to Communication Manager 6.3 ***2 of 4***

Enhancement	Keywords	Workaround
The following new capacity fields are added to page 11 of the display capacity screen under the IP PORT USAGE COUNTS section to help determine the IP port usage: ● Total IP Station Ports: Total IP Ports used by stations ● Administered IP Stations and Attendants: Total ports used by administered IP Stations and Attendants with extensions ● Softphone Enabled on Station Form: IP Ports used by softphones ● Unnamed Registrations (TTI ip phones): IP Ports used by unnamed registrations that are in a dissociated state to provide dial-tone	112478	
SHA-512 is now the default password-hashing algorithm.	112837, 120822.	
For security reasons, the customer wanted to hide the post-dialing DTMF digits on the list trace command. The **Hide Post-Dialing DTMF On List Trace** field to hide the post-dialing DTMF digits has been added to page 1 of the **system-parameters security** screen. The valid values of this field are y and n. The field has been defaulted to n causing digits to be displayed in the list trace output. The digits are not displayed when the field is set to y. The user profiles control the access to the **system-parameters security** screen.	120076	
A new denial event **5059 CDR Resource Exhaustion** is logged when the CDR buffer overflows and the value of the **Call Record Handling Option** is set to attendant.	120092	
Additional firewall information is displayed on the Firewall SMI page.	120104	
Communication Manager handles Refer request with the Refer-To header without the user part.	120766	
The password strength checking module is migrated to a more secure module.	120826	
Session Manager could not see `ANI Requested` when doing synchronization of information on the **list ars analysis** and **list aar analysis** forms. So add new data in **ANI Req** column that displays y or n.	121362	

Table 5: Enhancements delivered to Communication Manager 6.3 *3 of 4*

Enhancement	Keywords	Workaround
A restore of the backup image that has a hostname with the underscore sign (_) in it issues a message to the user to use the force option to restore the image.	121432	
Two new features have been introduced: The First Held feature: The first call that has been put on hold will be remembered by Communication Manager so that when the call is transferred remotely, the UCID of the call can be sent in UCID2 of the UUI. The Tandem UCID2 feature: When a call is being covered, forwarded or tandemed through Communication Manager, the value of UCID2 is sent with the original inbound call.	122072	
When SA9124 is activated, Communication Manager stores the called-party information from the first alerting or offered event for in-bound ISDN PRI, ISDN BRI, H.323, and SIP trunks calls and uses that as the connected-party information for all the subsequent messages of that party. For out-bound calls, the connected-party information is stored in either the alerting or connected events and this information is used for all the subsequent messages of that party.	122125	
Security enhancements were made to the web server configuration.	122732, 122733	
With this enhancement, users can migrate data from a CM 5.2.1 system to a VMware system: ● System Security ● System Access (Users, Groups, Permissions, Passwords, etc) ● SNMP configuration ● Scheduled jobs (cron jobs)	122751	
A new parameter count has been added to the list off-pbx station-mapping command.	130132	

Table 5: Enhancements delivered to Communication Manager 6.3 ***4 of 4***

Enhancement	Keywords	Workaround
A new field **(SA9125) - Select Standard Transfer Dynamically for SEMT?** has been added to page 10 of the **system-parameters special-applications** screen. With this field, Communication Manager checks if the supported header of the incoming REFER message from SIP User Agent contains the Replaces parameter. If the REFER message does not contain the Replaces parameter, Communication Manager marks this user agent as incapable of performing SIP Endpoint Managed Transfer.	130158	

Enhancements delivered to Communication Manager 6.3 for Avaya Video Conferencing Solutions

Table 6: Enhancements delivered to Communication Manager 6.3 for Avaya Video Conferencing Solutions

Enhancement	Keywords	Workaround
Video support for Radvision SIP endpoints registered to Session Manager. **Note:** This feature includes support for only XT5000, XT4200, and XTE240 and not for XT1200.		
Support for Radvision XT endpoints with embedded MCU (SIP registered to Session Manager). **Note:** This feature includes XT5000 and does not include interop with ADVD and Avaya one-X® Communicator.		
Multiple Communication Manager and Session Manager support for Video (using SIP trunks).		
Multiple Communication Manager support for Video (H.323 endpoints over H.323 trunks).		

Problems fixed in Communication Manager 6.3

Problems fixed in Communication Manager 6.3 and 6.2 SP0

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***1 of 7***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP0.	112760, 112836, 113004, 113052, 113118, 113268, 120058, 120154, 120221, 120255, 120295.	
DTMFs were not sent when a call was made to a SIP station and initial IP-IP Direct Media was enabled.	112129	
No video was observed on a call between video-capable endpoints when the conference and transfer features were used.	112255, 112734, 113017.	
SIP signaling groups that came into service without VoIP resources sent resubscribes (resubscribe NOTIFY) once to the far-end. However, since the SIP trunks did not go into service until the VoIP resources were present, the far-end Session Manager and Communication Manager got out of synch. Session Manager then, did not send polling subscribes back to Communication Manager, and SIP calls from Session Manager did not behave correctly.	112418	
Call transfer, from a SIP station on Communication Manager that had media encryption disabled to a SIP station on Communication Manager that had media encryption enabled, failed.	112513	
After the daily maintenance activity was performed, all SIP calls failed.	112611	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***2 of 7***

Problem	Keywords	Workaround
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	
IQ, Proactive Contact and CMS did not have accurate reports of abandoned calls when an ICR 2.0 on Avaya Experience Portal pulled back a call after delivery to a Call Center Elite auto-answer agent and before transferring the call to media.	112794	
Due to ill-formed SDP, a high-definition SIP video call using H.264 codec dropped at session refresh time.	112815	
There was no talk path when the far-end changed key after hold on a call that was on TDM.	112821	
There was no talk path on a conference call with Avaya Conference Server.	112862	
A call made from an Avaya 1050 endpoint to an Avaya 1010 dropped after sometime.	112899	
Calls made to MMCS as moderator caused Communication Manager to reset.	112901	
Occasionally, Communication Manager reset when video endpoints changed capabilities midway on a call.	112950	
When a large number of long duration SIP calls were made, the system ran out of memory and crashed due to memory leaks.	112951	
Logs were flooded with error messages while placing video calls. Excessive log entries reduce performance and obscures important information in the logs.	112953	
Intermittently, calls that routed to agents had music added to the call when they zeroed out of voicemail.	112973	
On Communication Manager, MLPP (Multiple Level Precedence & Preemption) was enabled and shuffling was disabled. An H.323 phone called a SIP phone. After the SIP phone user answered the call, two way talk path was observed but ringback did not stop at the H.323 phone.	113010	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***3 of 7***

Problem	Keywords	Workaround
On Communication Manager, an H.323 station called a CS1K SIP station over a SIP trunk. The CS1K SIP station user put the call on hold and then put it on unhold. No talk path was observed after the call was unheld. The call dropped after 32 seconds.	113011	
A call between 2 H.323 video endpoints, such as HDX H.323, had only video and no audio when audio shuffling was turned on.	113016	
An H.323 phone was operating in the auto-answer mode and was power cycled. When the phone subsequently re-registered with Communication Manager and a call was made, the call was auto-answered by Communication Manager and cut through to the phone. The call had no audio until the user went off hook.	113036	
There was no talkpath on a SIP station that had auto answer mode enabled after the station put a call on hold and then resumed it.	113045	
On Communication Manager (CM1), shuffling, NCR (Network Call Redirection), and MOH (Music On Hold) was enabled. On another Communication Manager (CM2), NCR and MOH were enabled. A SIP phone on CM1 (SIP1) called another SIP phone on CM2 (SIP2). SIP1 put the call on hold and SIP2 also put the call on hold. Then, SIP1 put the call on unhold and SIP2 also put the call on unhold. There was no talk path and the call dropped after 32 seconds.	113048	
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	
When the **Override ip-codec-set for SIP direct-media connections?** field was set to **NO**, SIP to SIP calls that reconfigured from TDM connected to direct-ip used an audio codec based on the preference of the SIP endpoint, rather than the audio codec based on the system administrator preference as described on the **ip-codec-set** screen.	113070	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***4 of 7***

Problem	Keywords	Workaround
On Communication Manager, Shuffling was enabled and Music on Hold was disabled. Attended transfer between 3 SIP stations failed and there was no talk path. The call dropped after 32 seconds.	113071	
A SIP station user was unable to deactivate the call-fwd and cfwd-bsyda buttons after Communication Manager restarted.	113081	
1) An agent had at least 2 skills on page 2 of the **agent-loginID** screen. 2) The first skill was administered without timed After Call Work (ACW). 3) The second skill was administered with timed ACW. 4) A call was made to the second skill and the agent answered. 5) The agent finished the call and went into timed ACW. 6) Another call was made to the first or second skill while the agent was still in timed ACW. When the second call was made while the agent was still in timed ACW, timed ACW was preempted and the second call was delivered to the agent.	113082	
On Communication Manager, Shuffling was enabled. A CS1K phone user made a call to an H.323 station over a SIP trunk. The CS1K phone user put the call on hold and then unheld it. The call dropped after the user unheld the call.	113095	
A SIP one-X Communicator user was unable to make calls after sending ISAC/16000 and ISAC/32000 wideband audio codecs.	113103	
The Communication Manager virtual machine restarted on an S8300D server each Sunday morning at 4:30 AM.	113111	
There was no video when an Avaya A175 Desktop Video Device called a non-video SIP endpoint which conferenced or transferred the call to another video capable Avaya A175 Desktop Video Device.	113121, 113144.	
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	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***5 of 7***

Problem	Keywords	Workaround
Occasionally, there was a Communication Manager reset during call clearing when an audit was run.	113172	
Team button notifications were not sent to the new monitoring station when the monitored station was again monitored.	113173	
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	
Communication Manager reset when a SIP trunk call got forked downstream.	113178	
Communication Manager did not allow SIP INVITE messages without media. This put a negative impact on features such as Callback Assist.	113208	
On Communication Manager with call preservation administered, a far-end domain failover to another backup server caused a call to drop when the call used H.248 media gateway resources.	113212	
There was no talkpath when a SIP station called an H.323 station when Direct Media was disabled.	113239	
A call made to an IPv6 H.323 station over an IPv6 SIP trunk failed when SIP Direct Media was enabled on the SIP trunk.	113245	
Normal Service Observation functionality on a VDN did not change when Service Observation by Location was activated on the VDN.	113248	
There was no talkpath on a SIP call made from an IPv4 endpoint to an IPv6 endpoint when SIP Direct Media was disabled.	113272	
During heavy SIP traffic, the system restarted.	113285	
A call that was covered to a station that had enhanced call forward set dropped without covering to the subsequent coverage points.	120056	
A Life Size endpoint tried to dial in an Avaya A175 Desktop Video Device when the Avaya A175 Desktop Video Device was on another video call. The call dropped when the Avaya A175 Desktop Video Device answered and transferred it.	120066	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***6 of 7***

Problem	Keywords	Workaround
On Communication Manager, attended transfer between a Capneg SIP phone and an RTP non-SIP phone failed.	120069	
A call made from an RTP SIP phone on Communication Manager to a Capneg SIP phone on another Communication Manager resulted in a system reset.	120088	
A call that was made from an RTP SIP phone on Communication Manager with media encryption enabled to a Capneg SIP phone on Communication Manager with media encryption enabled dropped when it was put on hold.	120102	
Communication Manager reset when a 422 Response was received to the initial call establishment INVITE for a SIP to SIP video call using the H264_SVC codec.	120133	
Bridge notification was not cleared when a call was dropped on the principle station.	120131	
The system reset when a user tried to upgrade an audio call to a video call.	120140	
When a SIP station to SIP station call covered to Communication Manager Messaging (CMM), Communication Manager could outpulse a string of digits to the CMM which caused CMM to play announcements very quickly.	120147	
A SIP station could not make a call to another SIP station over a QSIP trunk when direct media was enabled on the originating Communication Manager.	120175	
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	
An incoming PSTN call made to an x-ported station could not be answered on its bridged appearance.	120204	
A call that was routed to an EC500 station over a SIP trunk dropped after 32 seconds.	120205	
Calls made between Capneg SIP phones on different Communication Manager systems became RTP when Direct Media was not enabled	120219	

Table 7: Fixes delivered to Communication Manager 6.3 and 6.2 SP0 ***7 of 7***

Problem	Keywords	Workaround
Communication Manager reset when a 422 Response was received to the initial call establishment INVITE for a SIP to SIP call over a QSIG trunk.	120232	
A SIP call dropped when an EC500 station bridged in.	120235	
A call, made to an IP Softphone whose telecommuter was a SIP trunk and had direct media enabled, dropped after 32 seconds.	120289	
Communication Manager reset during a Direct Media call when SIP debugs were enabled.	120392	
A video-enabled call made from an Avaya A175 Desktop Video Device to another Avaya A175 Desktop Video Device dropped after 32 seconds after it was answered on the EC500 endpoint.	120399	
On Communication Manager, when a service-link call was made using a SIP trunk, the user could not connect to another incoming call. The user continued hearing the original connected call even after placing the call on hold.	120496	Drop the initial service-link call.
When the caller and called parties were SIP stations, the send all calls feature failed for remote coverage paths.	120567	

Problems fixed in Communication Manager 6.3 and 6.2 SP1

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***1 of 16***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP1.	112556, 113053, 113075, 120062, 120084, 120111, 120113, 120286, 120325, 120408, 120471, 120475, 120512, 120561, 120630.	
A race condition in the SAT process caused problems for programs that used the OSSI interface to Communication Manager, such as LoadAgent.	102799	
The number of simultaneous video calls that can be made on Communication Manager was limited to one-half of the design intent. This limit was incorrect.	103102	
A call that was parked by a SIP endpoint was not unparked from the parking station after the Call-Park Timeout Interval expired.	111572, 112438.	
Occasionally, customized labels of buttons on a button module were deleted when the station type was changed.	111642	
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	
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 numbering format was also incorrect.	112128, 112330.	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 *2 of 16*

Problem	Keywords	Workaround
The system had a conference tone but did not have a service observing warning tone and a service observing conference tone. When an agent with a service observer transferred a call after the third party had answered it, the caller heard the conference tone.	112147	
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	
An agent had EC500 enabled. When the agent received an ACD call, reporting recorded the call as interflowed.	112256	
Calling Party Number was not sent in the SETUP message in a SIP-ISDN interworking call when the incoming SIP trunk call had Privacy:ID.	112271	
The Dial Plan Transparency call failed on 96xx phones that had Special Application firmware installed.	112279	
An inactive Enterprise Survivable Server (ESS) responded to a Location request (LRQ) that was sent from a CISCO gatekeeper with a Location Confirmation (LCF) message.	112324	
Occasionally, QSIG Path Replacement did not work after an interchange of duplicated Communication Manager servers.	112343	
There was no logged-in event when an agent logged into a split using the Add Agent Skill FAC and the split was monitored. Similarly, there was no log-out message when the agent removed a skill.	112384	
A denial event was not observed when calling from EC500 to a station, when both the stations are in different COR(with no call permission to each other) and have the same 'Station Lock COR' for each of the COR.	112385	
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	
Communication Manager did not report the bad extensions on a video call with Tandberg.	112420	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***3 of 16***

Problem	Keywords	Workaround
A conference call was made using 2 SIP phones. The call dropped when the SIP trunk was configured with a unicode name as auto and the SIP phones were administered with name2 values.	112501	
Extra digits were inserted in the called digits received in the SETUP message over an H.323 trunk. This caused incorrect call routing.	112614	
IP agent did not hear VOA when incoming call was over SIP trunk and IP Agent had telecommuter over SIP trunk.	112623	
DPT did not work in the LSP mode when the idle appearance select feature name extension was used on a logged-off station.	112626	
A service observer could not join an active call on the observee which involved unattended conference. Only one service observer was allowed on the call.	112642	
The display screen was not updated on an IP station after attended transfer was made by a SIP station to the IP station.	112658	
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	
An agent at Station A called another agent at Station B. Station B and Station C were part of the pickup group. When the call was answered by an agent at Station C, the call log on Station B showed the call as a missed call displayed as a handset icon with the X symbol instead of the handset icon with two arrows to indicate call redirection.	112690	
Customers could not make a video call from ADVD to an HDX-SIP endpoint with H.264 video codec. ADVD displayed a black screen.	112713	
A SIP video endpoint was dropped from a conference call that was made between an audio-only SIP endpoint, an H.323 video endpoint, and the SIP video endpoint.	112721	
Audio calls that were made from an audio endpoint to a video endpoint and were subsequently conferenced or transferred to another video endpoint did not display any video. Occasionally, some parties dropped from the call.	112747	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***4 of 16***

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	
CDRs were generated with service observer as the originating party.	112769	
SIP calls failed when the SIP messages had a Call-Info header with URN (Universal Resource Name).	112782	
Unplugged IP phones did not unregister.	112783	
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, customized labels of buttons on the button module were deleted with a change of station type.	112839	
On systems with CDR links, a warning alarm was raised every time the periodic maintenance was run.	112853	
An additional pound sign (#) was added to the dialed number when ASAI was used to make calls and the minimum and maximum number of digits in the AAR/ARS table were not equal.	112903	
A station had call forward enabled. Enabling call forward again on that station after fail over and fall back caused a change in the performance of the station.	112911	
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	
Misadministration of UDP AAR tables resulted in routing loop between Communication Manager and Session Manager. This consumed all the administered trunks between them.	112978	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 *5 of 16*

Problem	Keywords	Workaround
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	
There was no talk path when a desk phone answered a long held recall call and the Optim Shared Voice Connection feature was in progress.	112998	
Occasionally, Communication Manager reset.	113009	
Occasionally, the Trunk-ID button omitted the trunk member number.	113021	
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	
Occasionally, Communication Manager could reset during a call preserving upgrade.	113033	
Occasionally, when IP synchronization was enabled, the rebuild process froze and did not finish.	113049	
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	
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	
User saw a VDN that did not exist while executing the **`list usage extension`** command on the SAT screen.	113088	
A user dialed a trunk group TAC and the call was recorded, bridged on to and service observed while dialing. Duplicate digits were signaled out the trunk which resulted in misdialed calls.	113094	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***6 of 16***

Problem	Keywords	Workaround
The translation audit that runs as part of daily maintenance caused processor occupancy spikes that increased as the number of translated extensions increased. The translation audit executed the list station command in the background mode, which consumed a lot of processor time. Timeouts occurred on external devices, such as System Manager, when the occupancy spikes lasted for several seconds and delayed the response to INVITE messages. This could cause calls to fail during this part of the translation audit.	113097	
Calls made from an Avaya 1000 Series Video endpoint to a Cisco 99xx via SIP trunk and registered to CUCM (Cisco Unified Communications Manager) resulted in one-way video.	113120	
A video call dropped, or the screen displayed a black window when the call was transferred to another video endpoint.	113160	
The SA8475(SOSM) did not work after a system level 2 restart.	113185	
No video was observed when an H.323 HDX, that was registered to Polycom CMA, called an Avaya 1000 Series Video Conferencing System.	113191	
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	
On calls made between an H.323 endpoint and a SIP endpoint, the H.323 endpoint received no audio when a Siren audio codec was chosen.	113194	
One-way video was observed on calls made from an Avaya 10x0 endpoint to a Cisco 99xx endpoint via a SIP trunk, which was registered to CUCM.	113210	
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	
Print jobs scheduled using Report Scheduler failed.	113223	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***7 of 16***

Problem	Keywords	Workaround
Executing **`reset system 2`** did not log out SIP ACD agents.	113227	
Duplicate station command displayed an error when the **Display Character Set** was set to **Katakana** and the **Display Language** field on the **station** screen was set to unicode.	113229	
Communication Manager requirements for the Linux syslog-type logs state that each log should be rotated based on independent limits for size and age. When both criteria were specified in a single logrotate configuration file, the logrotate utility only rotated the log file based on the second of the two entries. This update corrects that problem by using separate configuration files for the two limits.	113243	
An active call dropped when a SIP bridge tried to join the call on the principal station.	113253	
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	
Occasionally, logged-in agents could not call voice-mail.	113264	
Talk path was lost between stations after two successive interchanges of media resources in a duplex media processor configuration.	113270	
Incorrect display was observed at the calling party station when called party station had Enhanced Call Forwarding Unconditional(ECFU) or Enhanced Call Forwarding Busy (ECFB) activated.	113275	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***8 of 16***

Problem	Keywords	Workaround
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	
The memory usage for processes that used large quantities of memory was not displayed correctly while executing the fasttop and mfasttop commands.	120028	
For calls routed over a SIP trunk, UUI (User to User Information) sent with a switch classified call request did not appear in the INVITE message.	120032	
The video-codec priority was changed in the **Answer** field, but the new video-codec priority was not tandemed as per the changed priority. Instead, the codec priority of the Offer field was used to tandem the SDP.	120035	
When the Terminal Translation Initialization (TTI) feature had associated a phone with a display, the display would not clear.	120048	
One way video was observed on a call that was made by a 10x0 video endpoint to ADVD and blind transferred to an HDX.	120050	
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	
For calls that covered to a member of a coverage answering group, the stations monitoring the member did not play an audio alert.	120067	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***9 of 16***

Problem	Keywords	Workaround
Enabling video-debug prints caused the Communication Manager server to reset in systems using H.323 video.	120080	
SIP calls were dropped when the far end sent comma-separated diversion headers.	120081	
Occasionally, Communication Manager reset.	120087	
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	
A conference of more than six parties on an H.248 media gateway failed on Communication Manager with Application Enablement Services and a DMCC application enabled.	120124	
There was no ring back when an incoming ISDN trunk call terminated on a VDN, and then was routed to a SIP station.	120126	
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	
A call that was redirected to voice mail over a SIP trunk was reported as abandoned when the caller pressed zero to talk to an agent.	120142	
Communication Manager did not use UPDATE for session refresh which caused some SIP calls to drop.	120153	
Occasionally, calls failed after a firmware downgrade of a media gateway. This happened because the media gateway did not support some features the previous firmware had provided. Communication Manager was reset for the media gateway to correctly process the calls.	120159	
The aut-msg-wt button lamps for agents were not updated unless the agent was logged in.	120160	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***10 of 16***

Problem	Keywords	Workaround
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	
A call made from a Tandberg video endpoint to AAC or MMCS dropped when SIP Direct Media was enabled.	120178	
Holiday tables numbered above 255 could be administered but were not handled correctly in vector processing.	120183	
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	
Sometimes the UUI from a switch classified (predictive dial) call request was not propagated over an ISDN-PRI trunk.	120194	
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	
An asterisk was added to the form label for **Use VDN Time Zone For Holiday Vectoring** switch to indicate that it follows VDN override rules.	120202	
Multiple transfers of an Avaya 1000 Series video endpoint could result in lost video.	120209	
Occasionally, the SAT **`list ip-interface`** commands got into an endless loop. This resulted in a high occupancy condition.	120238	
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	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***11 of 16***

Problem	Keywords	Workaround
When a system contained a faulty H.248 gateway and three IP endpoints in one NR, the system established a three-party conference using a fully operational gateway in another NR. However, the system continually tried to move the conference to the faulty gateway in the NR of the phones. These constant move attempts repeatedly cut and re-established audio.	120257	
An auto-answer agent logged in on a DCP station with auto-answer = none caused the DCP station to lose voice-path. This occurred when the agent logged out by hanging up instead of using a FAC.	120265	
A patch could not be removed.	120273	
Communication Manager restarts could occur when certain unnamed registration station administration tasks were performed.	120297	
Calls, made to IP softphone, One-X Communicator, One-X Attendant, One-X Agent in the TeleCommuter mode, dropped.	120308	
Corrupted hunt group data prevented saving translations.	120320	
Music On Hold was played on a call when MOH Class Of Restriction was disabled.	120323	
An internal Communication Manager software error prevented a call from selecting a member from an outgoing H.323 trunk group even when the H.323 trunk group was available. The configuration required the far-end network region of the trunk group and the network region of the media processors in the originating port network to not have an administered ip-codec-set.	120328	
Occasionally, People+Content did not work on video calls.	120330	
One-way talk path was observed when a call that was made from a SIP capneg endpoint to another SIP capneg endpoint that had video softphone enabled was answered by an EC500 station that was a DCP endpoint.	120334	
When a user made an on-hook trunk call from a 96x1 H.323 station and a second call landed on the station, the subsequently dialed digits for the first call were displayed on the second call appearance.	120342	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***12 of 16***

Problem	Keywords	Workaround
The call-pickup lamp update was not sent to SIP endpoints that were part of a pickup group.	120345	
A call that covered to a station with enhanced call forward enabled dropped without covering to the coverage points.	120351	
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	
A call redirected to a pickup group made a station of the pickup-group ring endlessly.	120389	
Occasionally, no talk-path was observed on a SIP call when the **Override ip-codec-set for SIP direct-media connections** field was enabled on the station.	120434	
Communication Manager did not parse the uri-parameter of the Proxy-Authorization header in the incoming ACK message correctly, which caused the call to drop.	120450	
A call was made to an IP softphone whose Telecommuter is a SIP trunk. The call did not complete and went to coverage.	120456	
Occasionally, a station did not play the reorder tone for a SIP call.	120460	
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	
Occasionally, SIP calls either dropped or one-way audio and video was observed on them.	120485	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***13 of 16***

Problem	Keywords	Workaround
Calls were made from an HDX4000-SIP or an HDX9004-SIP endpoint to an HDX8000-SIP endpoint. Bad video resolution was observed when the HDX4000 or HDX9000 endpoint transmitted SIF (Source Input Format) video resolution.	120491	
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	
Customers were unable to make calls from OneX Communicator to HDX-SIP. This resulted in HDX-SIP transmitting one-way video.	120516	
Customers were unable to make calls from an HDX4000-H.323 or an HDX9004-H.323 endpoint to an HDX8000-SIP endpoint. This resulted in CIF (Common Intermediate Format) video resolution.	120520	
Occasionally, Communication Manager reset.	120521	
An HDX8000 endpoint transmitted one-way CIF video when calls were made from the HDX8000-SIP endpoint to an HDX4000-H.323 endpoint.	120525	
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	
On Communication Manager, the user could not connect to another incoming call when a service-link call was made using a SIP trunk. The user continued to hear the original connected call even after placing the call on hold.	120571	Drop the initial service-link call.
Service observed calls that were made over R2MFC trunks dropped when they were put on hold.	120578	
An IP softphone registered with a callback number had a call routed using a SIP trunk. The other party in the call was also a SIP station, and the call shuffled to Direct-IP. The call either dropped or lost talk-path that could not be restored when the COR of the IP softphone did not support Music on Hold and the SIP station put the call on hold.	120583	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***14 of 16***

Problem	Keywords	Workaround
Communication Manager reset when the signaling protocol for a SIP trunk call involved provisional reliable responses.	120591	
Communication Manager could not conference a soft Flare station in a call between a soft Flare and a 96x1 SIP station.	120596	
The system reset due to IP traffic over a SIP trunk.	120598	
Video was lost on a video endpoint in AAC after the call was put on hold and then resumed.	120670	
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	
Occasionally, Communication Manager reset when a user dropped out from a conference call between SIP endpoints.	120726	
Occasionally, there was no talk-path on a SIP call.	120728	
SIP-subscription refreshes failed when all SIP b-channels were being used in a signaling group.	120767	
One-way audio was observed on a call made from a Polycom VVX video endpoint to an Avaya voice-only endpoint.	120776	
There was no talk-path on a call between a SIP endpoint and an H.323 Direct Media endpoint when the H.323 endpoint first selected a different call appearance and then answered the call.	120783	
A call was established between a SIP endpoint and an H.323 Direct Media endpoint. There was no talk-path if the H.323 Direct Media endpoint disconnected this call and answered a second call from a SIP endpoint.	120784	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 ***15 of 16***

Problem	Keywords	Workaround
When an H.323 One-X Communicator user logged-in and made a call to a 1010 endpoint, a 1020 endpoint or a SIP One-X Communicator, no video from the far-end was observed on the endpoints.	120792	
A call was made from a SIP station to another SIP station. The EC500 endpoint of the called SIP station did not ring because the **IP Video** field on that station was enabled.	120841	
Occasionally, Communication Manager reset.	120844	
A video station was on a conference call with an audio station and another video station, and the call had two-way video. After the video station hung up from the call, the remaining parties were also dropped.	120875	
The /var/log/ecs/commandhistory log permissions are now 644.	120915	
When a non-Avaya H.323 endpoint hung up a call, it was dropped from a subsequent call after an internal Communication Manager timer expired.	120954	
Multiple CLANs were used for AES sessions with Communication Manager. When the AEP connections were lost in these sessions, a delay of several seconds was observed in message transmissions.	120956	
When a VSST (Virtual Server Synchronization Technology) AES 6.2 High Availability server turned off unexpectedly, which could be due to power failure on the active AES server, an AES session was lost. This resulted in the loss of all CTI associations.	120957	
A call could not be made from the One-x Mobile application installed on a cellular phone.	120993	

Table 8: Fixes delivered to Communication Manager 6.3 and 6.2 SP1 *16 of 16*

Problem	Keywords	Workaround
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: ● The problem was not seen with simplex TN2602 circuit packs in a network region but disabling critical reliability on the duplicated TN2602 pair did not alleviate the problem. ● The presence of additional TN2302 and/or TN 2602 circuit packs in the same network region as the duplicated TN2602s may or may not have alleviated the problem. ● The presence of an H.248 gateway in the same network region would alleviate the problem. ● The presence of TN2302s and TN2602s, and H.248 gateways in other network regions also alleviated the problem.	121094	

Problems fixed in Communication Manager 6.3 and 6.2 SP2

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***1 of 15***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP2.	112532, 112552, 112554, 112561, 113159, 113219, 120200, 120267, 120278, 120329, 120335, 120433, 120517, 120548, 120600, 120634, 120743, 120760, 120799, 120897.	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 *2 of 15*

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP2 (continued).	120913, 120916, 120929, 120990, 121001, 121011, 121014, 121027, 121028, 121031, 121052, 121055, 121070, 121115, 121116, 121144, 121151, 121204, 121224.	
Occasionally, Communication Manager was unable to route an incoming call over an R2MFC trunk to an outgoing ISDN PRI trunk.	110369	
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	When the system displayes the SMI error "`The size of the file(s) are too large to be analyzed by the SMI page", ` use the command line tools on the server.
The inter-gateway connection that was established to provide synchronization between media gateways was torn down after a link bounce.	111922	
Occasionally, there was no video on a conference call between two video endpoints and one audio endpoint.	112357	
Occasionally, the **IGAR Now** field of the **`status ip-network-region`** command displayed an incorrect value.	112360	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***3 of 15***

Problem	Keywords	Workaround
A caller hung up while a VDN of Origin Announcement was playing at a telecommuter station, and the softphone associated with the telecommuter station remained off-hook with no call appearance selected. When the softphone was defined as manual-answer, it could not answer automatically when the telecommuter station answered the next incoming call.	112607	
When a SIP agent with **Forced Agent Logout by Clock Time** set was in pending logout mode and the agent changed work modes, the logout pending button was disabled.	112702	
A customer could not use calltype analysis to convert extension digits and LAR with digit strings longer than 13 digits.	112812	
Occasionally, a data record got orphaned in the BCMS/VuStats tables. The same record showed up as a call in queue on the monitor or in the list bcms reports even when the call was not in queue for any hunt group.	112823	
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	
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	
An ASAI application could not drop an announcement party from a call by using a selective drop request.	113203	
Incoming calls to an EAS agent failed to cover when they were redirected to a VDN on no answer.	113207	
The SMI pages did not allow hostnames that started with a digit.	113215	
Chinese display updates were not displayed when the unicode script tag **Kana** was not set on the endpoint.	113230	
The Hold-Unhold operation between two SIP phones failed when MOH was enabled, and there were no media resources.	113234	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 *4 of 15*

Problem	**Keywords**	**Workaround**
A transferred external call could not receive VDN return destination treatment.	113250	
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 a button-module.	113277	
There was only audio and no video on video endpoints when an audio device was used to conference a SIP video device and a H.323 video device on Communication Manager.	120001	
There was no talkpath when two SIP parties on a call simultaneously initiated the hold-resume operation.	120036	
A phone remained in the Discovering mode when an incorrect extension was typed in the log-in field and the (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	
A user called a SIP phone on Communication Manager via Session Manager from an MS Lync server. Communication Manager rejected this call with the 403 `far end domain name is invalid` message.	120168	
Occasionally, Communication Manager prefixed garbage characters to the calling party number in the delivered ASAI message to AES.	120197	
A memory leak eventually caused a Cold-2 restart when SA8891 was enabled.	120203	
The display was not updated on a bridge appearance when there was a delay in sending a Facility Message with Calling Party Name information after the setup.	120208	
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, a caller could not hear music after the trunk to trunk transfer completed, and the **Music (or Silence) on Transferred Trunk Calls?** field was set to **all**.	120217	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 *5 of 15*

Problem	Keywords	Workaround
Dial Plan Transparency was not invoked when an endpoint on a local survivable processor called another logged-off endpoint on the main server that had EC500 enabled.	120234	
Occasionally, some SIP and H.323 trunks were stuck so that no new calls could be made on those 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 calling party did not have the entire dialed digit string on the display while making an outgoing call over an overlap dialing trunk. This prevented the use of the call log on the phone to redial the same number.	120277	
A SIP call dropped when another SIP endpoint joined the call by using a bridged call appearance before the call was answered by the called party.	120301	
SIP signaling groups could go in and out of service when a backup Session Manager sent polling subscribers while the primary Session Manager was still active and controlling the SIP endpoints.	120312	
Dual ringback was played for an SRTP call made from an IP station to another IP station over a SIP trunk to Session Manager.	120344	
If one of the parties on a three party conference call had answered the call using a team button then all parties would drop when this user dropped from the call.	120347	
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	
Team button calls made to a station with OneX integration and Send All Calls activated did not ring audibly.	120380	
An IP telephone was registered to the wrong extension when it was changed from an unnamed registration to a named registration.	120382	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***6 of 15***

Problem	Keywords	Workaround
Occasionally, media gateway media modules were not inserted after the media gateway registration. This resulted in a no board situation when a **`list configuration board`** command for that board was run on the SAT. Also, since the board was not inserted, the board did not work. The board continued to not work until Communication Manager was reset.	120406	
Calls extended to EC-500 from the primary station were forced to priority ringing.	120410	
System accounts could be removed by Administrator Accounts SMI Pages. Now, these users are protected.	120415	
Users were unable to conference three monitored stations.	120423	
On a SIP station, an outgoing call that required an authorization code was dropped when another call came in on a bridged appearance.	120424	
Music on hold was not played when a call shuffled across port networks.	120440	
For an incoming SIP trunk call made to a VDN which was eventually routed to an agent the CDR recorded the agent extension instead of the VDN number even when the **Record VDN** field on the **system-parameter cdr** screen was set to **y**.	120445	
Occasionally, Communication Manager did not allocate memory for IP endpoints. This resulted in call failures or loss of talkpath.	120451	
When the lamp/display/button update periodic was run and an agent was in the converse vector step, the call-state of the agent changed, and the call failed.	120452	
Occasionally, a SIP call caused Communication Manager to restart.	120453	
When an ISDN call was answered by a station and the station transfered the call to another station whose coverage path was set to **all**, a generic greeting was played.	120455	
CPN was not displayed when calls made to an agent routed via a VDN. The agent station displayed **to VDN** instead of **CPN to VDN**.	120465	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***7 of 15***

Problem	Keywords	Workaround
Blind transfer of a OneX Communicator H.323 endpoint to an Avaya A175 Desktop Video Device resulted in low resolution (H.263, CIF) video.	120472	
Customers could not submit a SIP signaling group after setting the **IMS Enabled** field to **y**. The following error was displayed: `System management overloaded; please try again later`	120477	
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	
Customers saw an unadministered media gateway while running the **`list measurements ip dsp-resource gw summary`** commands.	120490	
IQ reports did not always have accurate data on incoming and outgoing non-ACD calls when agents were defined with their first measured skill that was not externally measured.	120493	
A SIP agent made a trunk call. The CDR produced did not capture the agent extension even when the **Record Agent ID on Outgoing** was set to **y**.	120501	
Abbreviated dial button calls from a DCP phone did not route correctly when ~s or ~p was part of the dialed string.	120528	
Attendant extended ARS calls for attendant groups in tenants greater than one routed to the wrong route pattern assigned in the partition-route-table based on the Time of Day Chart (Partition Group Number) assigned to the COR for that individual attendant.	120541	
The agent log-out tone was played to the agent and the caller when the resources for a call were provided by an H.248 media gateway.	120547	
Communication Manager had 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	
Members of a pickup group either received visual update messages or audible notification messages but not both.	120565	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***8 of 15***

Problem	Keywords	Workaround
Communication Manager converted incoming UDP and BFCP messages to lower case. Support has been added in Communication Manager to keep tge UDP and BFCP messages as upper case.	120605	
Occasionally, users over VPN using SIP service links did not have audio path.	120610	
The 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	
A caller heard ring back instead of busy after the transfer recall timer expired, and the call could not terminate on a station that had limited the number of concurrent calls feature.	120624	
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	
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 Telecommuter number updated on OneX Communicator was not updated in Communication Manager when OneX Communicator unregistered and registered due to a linkbounce or when a proper unregistration request was not recieved by Communication Manager.	120676	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***9 of 15***

Problem	Keywords	Workaround
The XML body of the feature-status-event NOTIFY message contained garbage characters.	120687	
There was no error message stating that a login ID is required when the SMI login page was submitted without entering the login ID.	120688	
Instead of playing the MOH, the Hold operation performed on a Cisco endpoint resulted in silence on the Avaya endpoint.	120691	
While changing a BRI station that had an X in the port field, the system displayed the following error message: `Error encountered, can't complete request; check errors before retrying`.	120702	
On an analog phone, the call did not disconnect when the user disconnected the call after pressing the flash button.	120714	
The called station on Communication Manager did not display the caller name when the call was made from a cell phone that was using the EC500 feature name extension over a QSIG trunk.	120715	
An incoming call was sent to Medpro even when SIP Direct Media is enabled on Communication Manager and the initial INVITE of the call contains c=0.0.0.0 in SDP.	120716	
A background audit caused the system to go into overload on a survivable server. This occurred when there were lots of translated stations and a file sync was done to the survivable server.	120721	
Communication Manager reset when calls were made over H.323 trunks.	120750	
BRI trunk d-channel (TBRI-PT) alarms were not upgraded correctly when Off-board Alarms (Other) were upgraded in the set options SAT command. The alarms always remained as downgraded warnings.	120758	
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	
A SIP station was logged off. The incoming SIP trunk call made to this station dropped when the call was answered by the EC500 destination.	120772	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***10 of 15***

Problem	Keywords	Workaround
When the user on a non-Avaya H.323 endpoint disconnected a call, it was dropped from a subsequent call after an internal Communication Manager timer expired.	120775	
IP signaling groups went briefly and erroneously out of service when an unexpected socket closure occurred.	120777	
On Communication Manager, an H.248 media gateway with full VoIP utilization caused trunks assigned to the media gateway region to stop functioning, thereby dropping calls in the process. The following conditions apply: ● One or more H.248 media gateways in a network region at full VoIP usage ● No other VoIP resources used in the H.248 media gateway region, that is no Crossfire boards (TN2602s) or Cruisers (TN2302s) ● No other connected regions exist in the H.248 media gateway region ● Trunks assigned to use H.248 media gateway region	120788	
The call timer did not start and the call was not logged when a user on an IP phone (46xx or 96xx) dialed a VDN using an autodial button with some of the digits and manually dialed the last couple of digits.	120806	
An incoming SIP trunk call failed to detect inbound digits when **Direct IP-IP Audio Connections** was set to **y** on the **SIP signaling group** screen.	120809	
When an agent pressed a button on the phone while listening to the VDN of Origin Announcement, the call was left ringing depending on the button that was pressed.	120828	
When all available SAT sessions were in use, there was no indication that an attempt to start a new SAT session failed.	120834	
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	
The **`status station`** command returned the error `Entry is bad` when statusing an endpoint in the shared control mode.	120847	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***11 of 15***

Problem	Keywords	Workaround
When Communication Manager was set up to use multiple media resources, there were timing issues during call setup, thus causing the call to not connect properly and to drop.	120860, 121206.	
The system displayed the following error message when a maintenance command was run on the SAT interface: `All maintenance resources busy; try again later`	120861	
Users were blocked from removing a station, and the system displayed the following error message even when the station was not assigned to any vector: `Extension must be removed from vector(s) before removal/change`	120866	
There was no video on calls made between Polycom HDX 8000 (SIP) registered with Session Manager and Polycom HDX 8000 (H.323) registered with Communication Manager.	120880	
Occasionally, Communication Manager reset while performing an operation related to the EC500 feature.	120883	
Busy Indicator for Phantom extension on SIP station did not work.	120885	
Occasionally, Communication Manager reset.	120901	
The domain control was not relinquished for a station that had bridged appearance administered when a call was answered at a bridged appearance and only the principal station was monitored.	120903	
Occasionally, Communication Manager reset.	120927	
When an agent on a Genesys softphone unheld a call the caller heard DTMF tones.	120958	
Occasionally, a SIP calling party heard the reorder tone when SIP Direct Media was enabled , and the called party answered the call.	120959	
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	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 *12 of 15*

Problem	Keywords	Workaround
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	
Users attempting to transfer a call between two parties and two service observers to a VDN received denial event 1746.	120974	
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	
A call was not forwarded and was dropped at a SIP station when ECFU was active on the SIP station and the call was made over a direct SIP trunk or over a QSIG trunk to this SIP station.	121003	
The **IQ** field did not include AAPC on the **Feature-Related System Parameters** screen.	121015	
The **Communication-Interface Processor-Channels** screen could not be edited.	121036	
There were three SIP stations: Station A, Station B and Station C. The users at Station A and Station B were on a call. The user at Station B performed attended transfer to Station C after session refresh timer. This resulted in no talkpath between Station A and Station C.	121041	
During SIP downstream forking, Communication Manager did not send PRACK for a reliable response which could lead to call failures.	121062	
A standby IPSI remained out of service even when the board was fully functional and a PKT-INT alarm with error 769 prevented the in-service transition. The alarm could be cleared only manually. A **`busy ipserver-interface`** command followed by a **`release ipserver-interface`** command was run. This problem was triggered by a network impairment that caused excessive transmission delays. The excessive delays caused the IPSI PKT-INT to fail test 886 - maintenance loop-around test. The test failures PKT-INT error 769 raised a major alarm and caused an IPSI interchange. When the IPSI entered the standby mode, it remained out of service until manually restored via busy/release.	121067	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***13 of 15***

Problem	Keywords	Workaround
Initial Invite SDP with video inactive caused some endpoints to incorrectly use video resources and open an RTCP (RTP Control Protocol) socket.	121134	
There were multiple system restarts and a flood of process errors logged against the LIP process due to memory corruption.	121177	
Media direction value was not tandemed when SIP Direct Media was enabled which caused call drops.	121182	
The **`status media-processor`** command caused a segmentation fault when there was an error in retrieving the DSP information.	121193	
A call made from an Avaya 10x0 video endpoint to a Radvision XT1000 series endpoint over an H.323 trunk resulted in one-way video for 30 seconds. Then, the call reverted to an audio-only call.	121194	
There were three SIP stations: Station A, Station B, and Station C. The user at Station A made a call to Station B. The user at Station B did not answer the call and the call locally covered to Station C. The user at Station C answered the call and put the call on hold. The call dropped when MOH was disabled and Maintain SBA at Principal was enabled..	121209	
A user on an H.323 endpoint on Communication Manager A called an H.323 endpoint on Communication Manager B. NCR and MOH were enabled on both the Communication Manager systems. SIP Direct Media was enabled on Communication Manager B and shuffling was enabled on Communication Manager A. The user at the called H.323 endpoint performed unattended transfer to a SIP endpoint on Communication Manager A. There was two way talk path between the called H.323 endpoint and the SIP endpoint. However, after session refresh, there was no talk path.	121251	
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	
Users were unable to answer calls on bridged call appearance.	121284	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***14 of 15***

Problem	Keywords	Workaround
Direct Media SIP trunk calls were dropped when they were made on an LSP server and the LSP became inactive (media gateways fell back to the Main server).	121288	
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	
When a system had only IPv6 media resources the system would restart.	121314	
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	
Calls that were made from Radvision video endpoints over H.323 trunks to SIP video endpoints registered to Session Manager resulted in no video.	121361	
In a large dial-out Radvision conference call, a OneX Communicator H.323 endpoint rings but drops right after going off hook. Other endpoints in the conference connect without any problem. When a OneX Communicator H.323 endpoint connects into a Radvision meeting room configured with a PIN, the endpoint can enter the PIN, but does not connect successfully.	121371	
Occasionally, there was no audio on calls made from an H.323 96xx endpoint to the Radvision bridge.	121399	
When an audio call was made to a video endpoint, Communication Manager reset.	121411	
The states of the line appearance of a SIP phone and the line appearance of Communication Manager was out of sync after the SIP station failed over from Session Manager 1 to Session Manager 2.	121435	
A call that was made from a SIP station to another SIP station covered incorrectly to voicemail over SIP.	121443	
A call did not cover after RONA (Redirect On No Answer) when Send All Calls was used as a coverage criteria.	121478	
A SIP call dropped when another SIP endpoint joined the call by using Bridged Call Appearance.	121527	

Table 9: Fixes delivered to Communication Manager 6.3 and 6.2 SP2 ***15 of 15***

Problem	Keywords	Workaround
SEMT (SIP Endpoint Managed Transfer) fell back to AST1 when the transferred phone had EC500, and Direct Media was enabled.	121628	

Problems fixed in Communication Manager 6.3 and 6.2 SP2.01

Table 10: Fixes delivered to Communication Manager 6.3 and 6.2 SP2.01

Problem	Keywords	Workaround
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.	121783	
Occasionally, Communication Manager reset, causing service disruption.	121786	

Problems fixed in Communication Manager 6.3 and 6.2 SP3

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 ***1 of 11***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP3.	120247, 120248, 120384, 120590, 120698, 121165, 121256, 121382, 121384, 121555, 122061.	
Using DDB to debug SAT problems caused existing SAT sessions to hang and prevented new SAT sessions from starting.	101603	
Resolution of a problem with synchronization over IP for a media gateway caused a segmentation fault, and Communication Manager restarted.	112982	
The SNMP walk of the G3 update MIB will now report back all the updates that are on the system as opposed to just one.	113051	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 ***2 of 11***

Problem	Keywords	Workaround
Successive snmpwalk and snmpgetnext queries on certain MIBs resulted in the some MIBs not reporting their first OID. Now, successive snmp queries on all G3MIB groups report back all OIDs.	113060	
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	
There was no video when a call was made from a OneX Communicator (SIP) endpoint to a Polycom-HDX (H.323) endpoint and then transferred to an Avaya Video 10x0 endpoint.	120303	
There was no video when a call was made from an Avaya Desktop Video Device to a OneX Communicator (H.323) endpoint and then transferred to a OneX Communicator (SIP) endpoint.	120470	
During the Explicit Call Transfer (Network Call Redirection) feature, a call failed when the service observer dropped the call.	120498	
When the SOSM application was monitoring a station that was part of a forwarding chain, no term event was sent when the call processing attempted to term to that station unless the station was the principle terminating point.	120515	
Occasionally, Communication Manager incorrectly displayed errors for Port Network and media gateway media processors during an audit.	120546	
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	
When there were lots of unnamed H.323 IP stations that were trying to register to Communication Manager at the same time, it caused Communication Manager to run out of message buffer and restart.	120628	
A CPU spike resulted in ALLOC_BUF and caused Communication Manager to reset in a duplicated system.	120643	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *3 of 11*

Problem	Keywords	Workaround
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	
In the case of a VDN transferring an incoming call over a SIP trunk to itself, the display information changed to show only the number and not the name when the call was answered.	120705	
Communication Manager did not send display updates to an outgoing H.323 trunk when the call was received by Communication Manager with display on the incoming SIP trunk.	120708	
An upgrade to Communication Manager 6.0.1SP5, or 6.2 caused degraded software duplication performance and higher processor occupancy.	120795	
A call was made from a OneX Communicator SIP endpoint to another OneX Communicator SIP endpoint and then transferred call over SIP trunk. There was no talkpath after the transfer was made.	120819	
Calls failed when the primary Session Manager in an active-active mode went down and the agent controlled the call from an Agent Desktop client device.	120824	
Under certain circumstances, a music source connected through a TN763D circuit pack stopped playing music.	120851	
Cause value IE in ASAI Held Event indicated that the call was on soft-hold even when the user put the call on hard-hold.	120874	
An incoming SIP call came to VDN that routed the call to IPSP in the telecommuter mode with service link over a SIP trunk. After service link answered the call, there was no talk path and call dropped after 32 seconds. This behavior was observed when SIP Direct Media was enabled.	120882	
Communication Manager delayed updating the display of a SIP station for an ISDN trunk call from a PSTN.	120899	
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	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *4 of 11*

Problem	Keywords	Workaround
When a SIP agent was logged in with multiple call handling and RONA, and was on a call when another call started ringing, and did not pick up the ringing call, the agent was put into the AUX work mode when the ringning call redirects.	120940	
This MR addresses two inter-related problems: 1. On systems with IPv6 disabled, random call failures occured every 10 seconds for devices using the PROCR interface. This problem can be avoided by enabling IPv6 and administering an IPv6 PROCR address. Note: There is no need to enable the IPv6 PROCR interface. 2. On all systems, a race condition caused a PROCR socket file descriptor leak. After a long period of time, PROCR socket file descriptors were exhausted, causing all new IP/SIP trunk calls (non-shared signaling) to fail, as well as any other application that required a new PROCR socket. CLAN sockets were not affected. The PROCR file descriptor leak occurred when a socket was abandoned/closed while it was being set up. This could occur in either mode, client or server. This is a rare event, which is why it took a long time before the leak caused any problems. There is no way to avoid the file descriptor leak, but a server interchange will fix the problem. PROCR file descriptors are not shared across servers, so the newly active server starts out clean and the subsequent reload of the the newly standby server closes all the leaked file descriptors, making this server clean as well.	120943	
Occasionally, Communication Manager incorrectly displayed errors for Port Network and media gateway media processors during an audit.	120950	
The Login Account Policy SMI page settings were not synched between active and standby servers in a duplicated pair when translations were saved on the active server.	121016	
There was no video when a call was made from an ADVD endpoint to a OneX Communicator (H.323) endpoint and then transferred to Avaya Video 10x0 endpoint.	121019	
Occasionally, calls could not be made from SIP phones.	121020	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *5 of 11*

Problem	Keywords	Workaround
Occasionally, a SIP trunk call dropped after a glare condition.	121045	
A denial event has been added which will let the customer know about the misconfiguration in the proxy route.	121065	
Occasionally, there was no talk path on a SIP call after the hold-unhold operation was performed.	121069	
An audio call that was escalated to a video call failed.	121103	
When the principal station had a coverage point to another station with the **Coverage All?** field set to **y** and a bridge appearance to the coverage point station, then a call made to the principal station covered and dropped.	121112	
Occasionally, Communication Manager inserted the national or international CPNprefix (00) twice before the same number. This resulted in double prefix.	121159	
CDR for call to a hunt group and picked up by the team button did not show the monitoring party extension even when the **Calls to Hunt Group - Record:** field was set to **member-ext** on the **system-parameters cdr** screen.	121166	
The phone rebooted after server interchange.	121167	
When an incoming external call was transferred, the caller could not hear anything after the call covered at the transferred location.	121168	
Agents with usd-mia skills assigned in call pick-up groups serviced by Direct Department Calling (DDC) hunt groups could not receive calls.	121174	
A trunk call made to Station A. Station A blind transferred the call to Station B, and the call covered to a hunt group, and was answered by Station C. This resulted in a CDR record that contained the extension of Station B instead of the answering party that is Station C in the called party field. The **Special Application SA7311 - CDR Record Answering Party?** field was turned on.	121178	
An incorrect CDR was generated for an incoming PSTN call that covered to SIP Modular Messaging and was transferred back to Communication Manager.	121196	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 ***6 of 11***

Problem	Keywords	Workaround
Occasionally, an active call made on a station dropped due to an internal software audit.	121198	
When a busy station was dialed using the redial button from a 96xx phone, the softkey options showed Hold/ Conf/Transfer/Drop instead of the expected Redial/ Clear options.	121230	
When a call made to an agent was conferenced across the trunk, the CDR showed the hunt group number as the calling party number instead of the agent number.	121231	
There was loss of MOH when a third party sent re-INVITE (session refresh) with a=recvonly.	121233	
When an agent who was service observed tried to transfer a call to another agent by pressing the flash button on an analog phone, the call dropped.	121235	
A SIP trunk call made to a station that had Send All Calls (SAC) active dropped when a SIP trunk going to Voice Mail had SIP Direct Media disabled.	121245	
The station security code change operation using a feature access code in abbreviated dialing did not work with One-X Communicator.	121253	
Occasionally, the performance of Communication Manager degraded due to misallocation of CPU resources.	121262	
A SIP call made to an AAFT client application did not complete properly.	121269	
Occasionally, a network outage caused the system to reset.	121273	
Exclusion, VOA and SO tones did not work properly with SIP Call Center Agents.	121283	
When all TN2312 IPSI circuit packs lost their sockets, Port Networks restarted.	121291	
Previously, when a WAN link with a configured BW limit has very little available BW calls that succeed could cause the inter-region BW limit exceeded count to be pegged many times, when it should never have been pegged. The BW limit exceeded count appears on the status **ip-network-region** screen.	121307	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 ***7 of 11***

Problem	Keywords	Workaround
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 monitored IP DECT station rang with an internal ring pattern even when an external call was made to it.	121325	
The display on bridged appearance on DCP station was blank when it went on-hook when a call was ringing on the bridged appearance.	121347	
When **SA8852** was set to **y**, and a trunk call was made to a VDN that had a vector that did 'route-to' to Station A and with the **cov** on the 'route-to' step set to **y**, Station A did not show the VDN name.	121363	
Occasionally, not all uses of extensions and vectors were displayed by the SAT **`list usage`** commands.	121373	
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	
Warning tones were not applied when a service observer was bridged onto an auto-agent-handled call involving a SIP call center phone, VDN VOA and zip tones.	121390	
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	
A pickup group had two SIP phones and Direct Media had been enabled. A call was made from a SIP phone to one of the SIP stations in the pickup group and was answered using the pickup FAC from the other SIP phone. There was no talk path.	121433	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *8 of 11*

Problem	**Keywords**	**Workaround**
When a **`display port`** command was run on port 17 (the Ethernet port) on a TN 799 CLAN circuit pack or port 33 on a TN 2501 VAL circuit pack, the SAT command line displayed an `Error Encountered Cannot Complete Request (eecr)` error.	121439	
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	
Calls made to a logged-off station that had both call forwarding and send all calls active were routed to the forward destination instead of the cover path.	121441	
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	
When the tie trunk group was used for Malicious call trace, it failed with denial event 5034 `Invalid MCT trunk group`.	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	
During a SIP downstream forking, Communication Manager did not send the 200 OK for a subsequent UPDATE request, which caused the call to fail.	121488	
An internal Communication Manager software error caused reset 1 & 2.	121496	
While making a DAC call using a SIP phone, the display of the phone did not show the number of the agent that was called. Instead, it showed either the station or the DAC skill.	121505	
When a call was made from an RTP SIP station to an SRTP Capneg SIP station on the same Communication Manager and then made an unattended transfer to a DCP station on another Communication Manager, the call dropped.	121513	
SIP Endpoint Managed Transfer (SEMT) fell back to AST1 when the transferred phone had EC500 enabled and Direct Media was enabled.	121522	
Occasionally, there was no audio on H.323 96xx phone calls made to the Radvision bridge.	121538	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *9 of 11*

Problem	Keywords	Workaround
Calls made from Radvision video endpoints over H.323 trunks to SIP video endpoints registered to Session Manager resulted in no video.	121539	
Cached old ports on a media gateway flushed with the change in network region configuration, such as change in port range, which caused calls to fail.	121572	
MOH and NCR was enabled on Communication Manager A. SRTP SIP-A on Communication Manager A called SRTP SIP-B on Communication Manager B. After answering the call, SRTP SIP-A put the call on hold for more than the session refresh timer on Communication Manager B. After session refresh, there was no MOH on SIP-B, and there was no talk path after the unhold operation.	121575	
A recently unheld call was dropped when the hold was an ASAI hold or a hard hold using the hold button on a station, and a second call was dropped by the far end party. When a CTI application retrieved the held call, the on-hook from the disconnect message on the previously active call caused the unheld call to drop.	121589	
SIP trunks became inactive after a traffic burst.	121591	
Upon an audio-only endpoint bridging onto a video call, the resulting 3-party call could not have the audio connected properly. Then the call dropped.	121605	
When a call was made over SIP trunk via Session Manager with SA8481 enabled, the customer was unable to see the alternate calling party number that was provided in UUI of ASAI make call.	121610	
When an ACD call was answered after being in the queue, the skill level that was sent to reporting was not correct.	121616	
An unattended transfer between two Communication Managers with SIP stations failed intermittently when shuffling was enabled on transferred to Communication Manager.	121622	
When a 96xx station dialed the Page Line Retrieval Access code to answer a pager call, the user could not put the call on hold because the softkey options showed Redial/Clear instead of the expected Hold/ Conf/Transfer/Drop options.	121632	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *10 of 11*

Problem	Keywords	Workaround
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	
The call of a SIP CC Elite agent phone was stuck in Communication Manager after it failed over to the secondary Session Manager when the user tried to drop the call during the fail over.	121653	
Unattended transfer among SIP phones failed with Direct Media and IP video enabled for both the station and the signalling group.	121665	
A SIP call dropped when another SIP endpoint joined the ringing call using a bridged call appearance and dropped out before the actual called party answered.	121667	
There was no talkpath on a call made to a SIP station that had multiple EC500 destinations administered on it.	121676	
There was no video on calls that were greater than 8192 kbps involving H.323 devices.	121687	
Occasionally, Communication Manager reset.	121689	
On Evolution Server, when the transfer target phone was AST2 phone with EC500 enabled, the SEMT (SIP Endpoint Managed Transfer) failed.	121694	
SEMT(SIP Endpoint managered transfer) failed when the transferred phone had EC500 enabled.	121697	
When the transfer target phone was an EC500 endpoint, after SEMT (SIP Endpoint Managered Transfer) the principal phone could not join the EC500 call.	121698	
Communication Manager has certain vulnerabilities described in Avaya Security Advisory ASA-2012-298. To see this document, go to http://support.avaya.com and search for that number.	121715	
Calls made to a non-ACD hunt group terminated to and rang members whose stations were logged out.	121716	
Calls listening to the disconnect tone did not prevent the new ASAI 3PCC to make calls from the originating endpoint.	121754	
Occasionally, Communication Manager reset. This caused service disruption.	121785	

Table 11: Fixes delivered to Communication Manager 6.3 and 6.2 SP3 *11 of 11*

Problem	Keywords	Workaround
Agents using One X Communicator could not log in to the system.	121803	
The display update on one endpoint failed when three endpoints were on a conference call.	121829	
Occasionally, the /var/log/wtmp file was not rotated which filled up the /var partition.	121853	
Communication Manager reset when an UPDATE message came in an early dialog and the session expired value was greater than the administered timer on Communication Manager.	121863	
Dial out calls made by the moderator of an Avaya Aura Conference call to an H.323 OneX Communicator did not complete. The call dropped immediately after the H.323 OneX Communicator answered it.	121913	
SIP A called SIP B. SIP B then conferenced SIP C into the call. SIP C dropped from the conference call and the call appearance on SIP B disappeared. However, there was still talkpath between SIP A and SIP B.	121924	
An SRTP SIP call made across two Communication Managers lost talkpath after session refresh when the Communication Managers have reverse codec orders with respect to each other.	121934	
SIP A called SIP B. SIP-B then conferenced SIP C into the call. SIP B then pressed the **Add** button to add SIP D. However, the conference call appearance on SIP B became the bridged appearance and SIP B could not join four parties to the conference call.	121962	
Calls made by using CTI to an auto-answer 96x1 SIP Contact Center station did not complete.	122044	

Problems fixed in Communication Manager 6.3 and 6.2 SP4

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***1 of 13***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP4.	121692, 121730, 121877, 121908, 121951, 122017, 122182, 122257, 122588, 122372, 122423, 122499.	
On the SMI interface, the session ID was not regenerated after user authentication.	093302	
In a tandem incoming SIP call with the plus (+) sign as prefix, the plus (+) was stripped. This resulted in an incorrect match for the calling party number conversion for tandem calls.	110070	
Auto-answer was enabled for a SIP term station. The auto-answer zip tone was not heard at the term station when Direct Media was enabled.	110639	
Occasionally, Communication Manager ISDN calls failed.	110861	
The **Display Forwarding Party Name** QSIG field on the **QSIG Trunk Group Options** page of the **ISDN trunk group** screen was not used. Therefore, it was removed.	111756	
A SIP endpoint had a coverage point administered with the Send-all-calls feature disabled, and its bridged appearance had the Send-all-calls feature enabled. All calls made to the SIP endpoint were covered immediately.	112550	
Workmode change from ASAI was performed immediately even when the agent had put a call on hold.	113100	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***2 of 13***

Problem	Keywords	Workaround
SIP stations displayed incorrect information about the connected party after blind/attended transfer to VDN or huntgroup or TEG (Terminating Extension Group).	113226	
When a SIP station was used to make a call to an agent who was logged on to an IP/SIP/DCP station, the calling SIP station displayed the name/number of the station on which the agent was logged in and not agent name/number.	113246	
The button labels of buttons from button 9 onwards on a button module of a 1616 phone type were displayed in English even when they were administered to be displayed in a different language.	120091	
When the password was changed by using the SAT interface for a non-TTI enabled phone, the new password was not updated on the phone	120141	
When there were two parties in a meet-me conference and when one of the parties dropped out, the other party endpoint displayed `Conference 1` instead of the Meet-me VDN.	120218	
There was no video when a call was transfered by a 96xx endpoint to a Polycom HDX 323 endpoint.	120396	
On a shuffled call, the called SIP station displayed the name and number of the caller even when **Mask CPN/ NAME for Internal Calls** was enabled on the **COR** screen of the calling station.	120461	
When a SIP trunk call received 403 forbidden from the far end and failed over to an ISDN trunk, DTMF from the calling station failed.	120495	
In the Communication Manager Feature Server mode, when the incoming trunk had only one unused member, then a call made to another Communication Manager station did not work.	120554	
A video call initiated from an ADVD to a 96XX SIP phone on a different Communication Manager was not shuffled after performing the hold-unhold operation on the ADVD.	120706	
A OneX Communicator (H.323) was used to make a call to a Polycom RMX over a SIP trunk. The call dropped when Communication Manager shuffled the call.	120738	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 *3 of 13*

Problem	Keywords	Workaround
On a video call between a SIP video endpoint and an H.323 video endpoint being served by different Communication Managers, a hold/unhold by the SIP video endpoint did not restore video when Music On Hold was disabled.	120791	
The agent endpoint displayed incorrect information when the incoming call was routed by the Avaya ICR (Intelligent Customer Routing) over a SIP trunk.	120837	
Recvonly on service link SIP trunk was ignored.	120973	
SIP to SIP calls used too many buffers to store name display data. This caused the codeset facility information element for a call to not be stored properly when all the buffers were used up. For this to happen, it took about 20,000 calls to be up simultaneously.	121026	
Transfer made from a SIP endpoint failed when Direct Media was enabled, and the resources were spread over multiple network regions.	121102	
A call came in on a SIP trunk, and the SIP trunk transfered the call to a VDN with a VDN return destination that includes internal calls. The VDN return destination was not applied after the call was transferred.	121148	
When the VuStats button was pressed on a SIP station and there were maximum number of VuStats sessions active, the SIP station did not display a meaningful message to the user.	121202	
When NCR was used, and an outbound PSTN call was made, and the call was transfered to another PSTN number, the SIP REFER message's 'Referred-By' header contained the local extension and not the DID. Because of this, the SIP service provider rejected the REFER with a 603 Decline.	121205	
When music on hold was administered, an auto-retrieved call after call park timeout continued to ring on the principal station even after coverage.	121261	
When an Avaya H.323 phone was registered to the Processor Ethernet (PE) of a duplicated Enterprise Survivable Server (ESS) pair, and an interchange occurred, the phone using Time-To-Service (TTS) immediately started registration attempts to the main server, instead of remaining registered to the ESS and reestablishing signaling with the newly active ESS server.	121271	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***4 of 13***

Problem	Keywords	Workaround
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	
The final 4xx (400,481,482,489) error response sent by Communication Manager did not add a To-tag in it.	121459	
If the **Type of 3PCC Enabled:** field on page 6 of the **station** screen (only for SIP endpoints) was changed from Avaya to None and the station was domain controlled, then subsequent commands on the domain control association were still allowed.This has now been corrected and the domain control will be removed if the change noted above is made.	121467	
UUI data could not be viewed by the called party in the following scenarios: ● the consult leg of a conference initiated by a SIP station ● the consult leg of a transfer initiated by a SIP station ● a supervisor assist initiated by a SIP station	121468	
Off-PBX mobile users dial Idle Appearance Select Feature Name Extension, and then dial destination number. If the destination user is busy, Communication Manager plays local busy tone to Off-PBX mobile user for SIP trunk, but for H.323 trunks, Communication Manager disconnects the trunk immediately and service provider plays the local busy tone. This behavior was not consistent for SIP/H.323 trunk. Now, with this fix, Communication Manager plays local busy to SIP/H323 trunk for busy timer (45 seconds) and then starts Auto Call Back timer (40 seconds). During this 40 seconds, the off-PBX mobile user can activate the Auto Call Back feature by dialing Auto Call Back Feature Name Extension.	121481	
When a remote worker operating in the Telecommuter mode performed a blind transfer to another party, the call dropped after it was answered.	121500	
There was no talkpath on a switched-classified call over ISDN PRI with the **Trunk Hunt** field set to ascend/descend.	121503	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***5 of 13***

Problem	Keywords	Workaround
Avaya OneX Communicator for MAC was used to make a call to SIP-A. Avaya OneX Communicator then tried to transfer the call to SIP-B that was logged off. The transfer failed, and Unhold from OneX Communicator resulted in no talk path.	121509	
The called party did not get the 181 SIP message for a covered call, when the call was covered on voice mail SIP adjunct.	121511	
Calls coming in from an Avaya/Tenovis Integral 55 server over a QSIG trunk to Communication Manager were dropped when the call was covered to a coverage Answer Group on Communication Manager.	121590	
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 pick-up group, all the endpoints of the pick-up group were in the alerting state, that is, the pick-up 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 using reporting prior to R3V161 and SIP, it was possible that SIP requested a priority that was not supported by the messaging to reporting. In such a case, the message to reporting contained invalid data.	121633	
Global RTCP and SNMP data was not sent to IP stations when there were more than 2000 IP stations on the system. Testing had shown that, on an idle system, approximately 2500 stations could be downloaded in 3 minutes. Three minutes was the maximum time allowed to download global RTCP or SNMP data to all registered IP phones on the system. Also, affected would be QOS/DiffServ changes made in the IP network region form if there were much more than 2000 IP stations in a network region.	121637	
The Communication Manager SIP stack was generating UUI header with 'To' for "to" that does not comply with the UUI draft. This lead to Nice recorder not recording the call.	121640	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***6 of 13***

Problem	Keywords	Workaround
SIP-A on Communication Manager A was used to make a call to a CS1K phone. SIP-A then conferenced SIP-B into the call. MOH was disabled. After the conference, there was one-way talk path on CS1K phone.	121643	
Per-location routing to a remote number failed with the Send All Calls feature activated on a station that has at least one off-PBX station mapping.	121649	Administer the **location** field on the **station** form or enable DCS coverage on the **system parameters customer options** screen.
Predictive dialed calls had their Call Detail Record with the VDN number instead of the Agent ID when the agent was not available immediately and the call went through wait treatment in the vector to find an available agent.	121664	
Service observers on analog stations were allowed to flash and put the call on hold when it should be denied.	121672	
SIP calls dropped due to inconsistent SDP states.	121674	
Predictive calls failed when **Call Classification After Answer Supervision** was disabled on the **system parameters features** screen.	121704	
Occasionally, IGAR calls failed with a denial event.	121706	
The display on a single-line display phone changed when a call reached the bridged appearance even when the phone was dialing to make a call.	121707	
Vector redirected virtual SIP calls, in conjunction with Avaya Aura Experience Portal's intelligent customer routing system, dropped when the trunk connections data relation audit was run.	121708	
When an EC500 station was used to make a call to the other principal station that had EC500 configured, the name of the called station was not displayed on the calling station.	121712	
There was no talkpath on calls made from an H.323 station to another H.323 station over a PRI trunk with overlap dialing, shuffling and encryption enabled.	121717	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***7 of 13***

Problem	Keywords	Workaround
IP signaling groups on an ESS went into the disabled state when the ESS was actively controlling port networks and media gateways. Then, all the media gateways returned to the main server.	121718	
After a reset system 1, the integrated directory services stopped working with the response `DIRECTORY UNAVAILABLE TRY BACK LATER`.	121720	
A double ring ping was heard on the endpoints of members of a pickup group when a call rang on the pickup group within 5 seconds of the previous call drop.	121735	
A call forwarded due to Enhanced call forward No Reply was routed to the destination set for Enhanced call forward Busy.	121741	
The **`status media-processor board`** command incorrectly returned a command conflict at times. The **`error enable filexfer`** command also returned the command conflict error.	121748	
There was a wrong display on an H.323 station when a call coming in from the PSTN was transferred.	121757	
Occasionally, the CLAN did not accept new registration requests from IP stations.	121762	
Translation corruption occured after removing a media processor IP interface that had an entry on the Media Processors Measurement Selection screen.	121768	
Calls forwarded to a non-ACD hunt group terminated to and rang members whose stations were logged out.	121770	
In the case of Re-Invite, Communication Manager did not update the display of the station correctly.	121784	Set **Identity for Calling Party Display: From** on the **Trunk** screen.
OneX Communicator in the telecommuter mode was using the location of the incoming trunk group member instead of the location administered on the SIP station screen.	121790	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 *8 of 13*

Problem	Keywords	Workaround
A conference call was held via Avaya Meeting Exchange. SIP-A called SIP-B. SIP-C then called SIP-A. SIP-A answered the incoming call by putting SIP-B on hold. SIP-A then initiated the conference and unheld SIP-B. However, the unhold operation was unsuccessful and there was no talk path. MOH was disabled and SIP-B had EC500 over a SIP trunk which had Direct Media disabled. All other SIP trunks had Direct Media enabled.	121795	
Occasionally, while activating or deactivating a Service Pack, the **`ldconfig`** command would return `Segmentation fault (core dumped)` while the Server Setup steps were running.	121806	
The FAX offer in unhold REInvite from Lync caused call drop.	121808	
More than 19 digits could not be dialed in an ASAI call to a SIP endpoint.	121815	
A caller was repeatedly asked to dial the name of the person he was trying to reach when the Dial by name server was being used.	121816	
Calls made from an EC500 configured cell phone to a VDN vector failed with a Hop Count Restricted denial event. An extension on Communication Manager had a cell phone configured as its EC500 endpoint. A call was made from the cell phone on trunk 304 that had Hop Digit set to Yes. Call went to VDN with vector 7. Vector 7 had a route to number step. This number was in the uniform dial plan pointing to AAR. AAR routed call to route-pattern 910 that specifies a Hop Limit of 4 and routed to trunk 1401. Trunk 1401 has Hop Digit set to yes. With the Hop Digit on trunk 304 and Hop Limit on route-pattern 910 enabled the call failed as Hop Count Restricted. When Hop Digit disabled on trunk 304 disabled, call completed properly. With Hop Digit enabled on trunk 304 but route-pattern 910 Hop Limit, a blank call completed properly.	121819	
Skills above 2000 on a Communication Manager configured with no CMS or IQ were unable to log in.	121827	
When a user was on a call on the cellphone (via EC500), the user got another call on the station while the limit-call button was enabled.	121834	
Outgoing calls from Visitor EMU logged in on the DCP set failed.	121838	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***9 of 13***

Problem	Keywords	Workaround
SIP phones dropped the call that did not receive crypto attributes in the rejected SAVP audio media line in SDP offer when the other party tried to upgrade the call.	121854	
Occasionally, customers using Service Level Objectives in skills did not receive the Interruptible Aux notifications while using Calls Warning or Time Warning Thresholds.	121859	
The per-loc dialplan entry was used when a user dialed a number through ASAI third party calling, even when all-loc entry had a longer match.	121870	
An agent conferenced two trunks together and the call was recorded by NICE. The call dropped when the agent disconnected the call.	121874	
Communication Manager reset in the process of collecting internal announcement usage statistics.	121886	
A display update was not sent by Communication Manager over an incoming SIP trunk after the Communication Manager agent dropped from the conference.	121891	
An IBM Sametime client was unable to make a SIP call to a Communication Manager extension.	121892	
When a CS1K user called an Avaya Aura SIP phone and then placed the call on hold, Communication Manager dropped the call.	121897	
Direct Media was enabled and MOH was disabled. A 96xx endpoint was used to call an ADVD station. ADVD then performed the hold-unhold operation. Unhold failed, and Communication Manager sent 488 to the ADVD station.	121898	
Occasionally, using attendant number 414 caused translation corruption.	121901	
When the system started running low on memory and swapped, the output of top and free was sent to /var/log/log/messages.	121902	
This fix adds atd to the root authorization in the access file so that atc can successfully start Communication Manager after a migration from 5.2.1 to 6.3.	121905	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***10 of 13***

Problem	Keywords	Workaround
A trunk to trunk tandem call showed a blank calling number on the called party when the calling number had a matching entry in the tandem-calling-party-num screen.	121921	
Announcement in the vector step did not complete.	121926	
A reset occurred when an UPDATE message came in an early dialog with a session expired value more than the administered timer on Communication Manager.	121931	
A memory corruption in the PROCR caused the system to enter an infinite loop, using nearly 100% of the CPU cycles. This starved other processes and caused the process sanity audit to request a system restart. Since the requested warm restart failed to clear the corruption, the system went through a second warm restart and then escalated to a cold2 restart. The cold2 restart cleared the corruption and the system recovered.	121932	
A call made to a SIP station did not have talk path when multiple EC500 destinations were administered on the SIP station.	121942, 121947.	
Occasionally, removing a source failed, stalling the Synchronization Over IP process from recovering from media gateway outages.	121943	
While making a DAC call using a SIP phone, the display would not show the agent who was called. Instead, it showed either the station or the DAC skill.	121944	
There was no talk path on the SIP group page call when the group page had more than 2 SIP group page members and group page originator was a SIP station.	121945	
When a call was made to a terminating extension group which was configured with a SIP station and the SIP station was not administered in the first entry, the call failed on answer.	121945	
A call answered by the second SIP station in a Coverage Answer group was not successful.	121945	
Occasionally, when executing the **`list bcms summary agent`** command, the system encountered a reset system 2 or reset system 4.	121952	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***11 of 13***

Problem	Keywords	Workaround
There was no talk path on an SRTP SIP call across two Communication Managers after session refresh, when the Communication Managers have reverse codec orders with respect to each other.	121953	
A customer changed a media gateway network region when the Synchroniazation Over IP feature was enabled.	121958	
Vector redirected virtual SIP calls with no media preference, in conjunction with Avaya Aura Experience Portal's intelligent customer routing system, were dropped when the trunk connections data relation audit was run.	121959	
Hold, transfer and conference operations were denied for hotline calls.	121960	
IP endpoints could not register and were rejected due to password error, even when the user entered the correct extension and password.	121965	
SIP phone A called SIP phone B. B then conferenced in SIP phone C which later dropped from the conference. When C dropped, the call appearance on B disappeared but there was still talk path between A and B.	121980	
A service link call over an R2MFC trunk failed when it was made after placing an earlier call on hold.	121983	
Short digit dialing was unsuccessful when 10 or more digits were administered as the location prefix.	121984	
When a call was placed on hold, MOH was not played intermittently.	121995	
Reference board administration was not sent to the media gateway after it was reset.	122018	
Calls did not drop when the caller disconnected while queued to a skill with an announcement playing and an SSC party on the call.	122031	
A reset level 4 occurred during a requested level 2 reset or a system upgrade.	122033	
A call with video endpoints could not be made when the called party sent two provisional responses with SDP e.g. 183(SDP) followed by 180(SDP). This caused Communication Manager to send a CANCEL for the INVITE transaction.	122053	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 ***12 of 13***

Problem	Keywords	Workaround
Occasionally, a transferred trunk call did not alert at the transferred to station. The caller heard MOH and eventually abandoned the call.	122071	
With two IQs connected to the same Communication Manager, one IQ could hang during DP pump-up if the DP on both IQ systems restart at the same time. This occured when a Communication Manager administrator typed **`busy mis all`** then **`release mis all`** on the SAT screen or when an IQ administrator restarted the DP on both the IQ systems at the same time.	122080	
If a customer had over 2000 logged-in SIP agents all with Qstats buttons, qstats stopped alerting.	122116	
On Evolution Server, SIP Endpoint Managed Transfer (SEMT) failed when the transfer target phone was a bridge phone.	122118	
Occasionally, when a customer with IQ added or changed the names of measured objects, there was a chance of system reboot.	122128	
The display on SIP auto answer stations while making a DAC call was incorrect.	122134	
Call transfer from IVR to Communication Manager using converse on data return feature access code did not work.	122141	
The system was unable to release call appearance after dialing an unregistered phone.	122155	
A call made over a SIP trunk failed when Communication Manager had no entry for the calling number in the public numbering table.	122157	
Transferred calls from OneX Communicator H.323 intermittently resulted in no audio and dropped after about 30 secs.	122165	
On Communication Manager, T.38 (FAX) calls using legacy port-networks for VoIP resources failed.	122216	
SIP One-X Communicator logged in the telecommuter mode was unable to originate any call.	122246	
In the case of cover all criteria, both principal and coverage point rang.	122261	

Table 12: Fixes delivered to Communication Manager 6.3 and 6.2 SP4 *13 of 13*

Problem	Keywords	Workaround
Send All Calls did not work properly over a direct SIP trunk.	122297	
Transferred calls from OneX Communicator (H.323) intermittently resulted in no audio and dropped after about 30 seconds and then call processing restarted.	122304	
Communication Manager did not send down the full list for all call-appearance buttons when a SIP station made a new subscription to an alternate Session Manager during the Session Manager fail over.	122312	
Communication Manager idle CPU occupancy increased on video calls.	122326	
Calls made from a SIP phone to a CS1K user dropped when the call went to coverage.	122330	
When a Windows-based Soft Flare attempted an SRTP call, Communication Manager encountered a reset system 2.	122369	
A mini core dump was observed followed by a Communication Manager reset when a call was transferred to a hunt group.	122370	
A race condition in the SAT process caused the system to restart.	122452	
Communication Manager reset when a call was transfered to a hunt group and hunt group members were also members of a pick up group.	122561	

Problems fixed in Communication Manager 6.3 and 6.2 SP5

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***1 of 13***

Problem	Keywords	Workaround
Issues associated with the following keywords were also corrected in Communication Manager 6.2 SP5.	120757, 121131, 121994, 121998, 122204, 122209, 122210, 122643, 122644, 122808, 122825, 122849, 122891, 130266.	
A personal CO line was assigned to an analog CO type board and there was no physical board in the slot. When the personal CO line was removed, the system displayed the following error message: `Error encountered, can't complete request; check errors before retrying`	102246	
The **Display Information for Failed Logins** option on the **Login Reports SMI** page displayed no information on servers with large amounts of login activity.	103052	
Type III registration counts were off for IP_Agent.	112115	
CMS and IQ reports displayed incorrect agent statuses when agents with MCH (Multiple Call Handling) took a second call. When the second call was released, CMS tallied the call as completed before the agent left timed after-call-work.	112590	
INADS alarming started functioning when INADS modem alarming was disabled and both SNMP INADS alarming and SNMP alarm abbreviation were enabled.	120030	
In case of forking, Communication Manager did not correctly handle 488 error response. This prevented the caller from being notified that the call could not be completed.	120151	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***2 of 13***

Problem	Keywords	Workaround
Communication Manager did not send the Comfort Noise indicator in the message when the **RFC3389 Comfort Noise** flag was set to Yes.	120549	
A call was made from one SIP station to another SIP station with Direct Media enabled. The calling party heard ringback even when the called party declined the call with 603 Decline.	120864	
When inter-gateway connectivity was absent, ringback was not connected to the caller.	121071	
Communication Manager incorrectly tandemed reinvite with a=inactive as reinvite with a=sendonly.	121378	
An incoming SIP trunk call made to an H.323 station was unable to get transferred to another H.323 station on the same Communication Manager.	121409	
Multiple hold and unhold operations on the B179 SIP Conference phone dropped active SIP calls.	121457	
Feature status button exclusion was on even after the call dropped.	121461	
Communication Manager did not send 181 response after a call was forwarded to another SIP station when the called party did not answer the call.	121495	
A SIP call that was not answered at the principal station and covered to a bridged appearance had no talk path.	121507	
For SIP calls, VoIP resources from a particular network region were selected more frequently than the other network regions.	121584	
The caller was unable to hear the ringback tone for ISDN-SIP-ISDN interworked calls.	121592	
The g3statsta MIB group would not report all stations if extensions were 6 or more digits, included punctuation, and any extension ended with the number 9.	121627	
A phone could not be registered in the AnnexH mode after sStoredData[] was full.	121644	
An H.323 station on Communication Manager displayed incorrect calling party information for a call that was forwarded over a SIP trunk from a third party PBX.	121645	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***3 of 13***

Problem	Keywords	Workaround
A 96x1 SIP phone was the SEMT transfer target. The internal CID (Caller Identification) on the line disappeared after the transfer.	121650	
Occasionally, calls could not be made and received until Communication Manager was reset.	121722	
When an incoming ISDN call to Communication Manager covered to a SIP integrated voice mail and then transferred out to a station on Communication Manager, the display on the station did not show the correct calling party number.	121756	
An application did not get a transfer message for an attended transfer.	121791	
Block Hang-up by Logged-in Auto-Answer EAS Agents was enabled on the system parameters features screen. Auto-Answer agents could not answer a call when they logged in using a CTI product and did not manually toggle a line appearance after logging in.	121796	
Incoming H.323 trunk calls failed intermittently when Communication Manager had a mix of media processor boards and media gateways available for media resources.	121824	
The generic greeting was played when a call was blind transfered from a SIP trunk to a vector with wait and route to another station and covered to SIP Modular Messaging.	121855	
There was an internal Communication Manager software buffer allocation error and the server reset when an incoming ISDN call was made to a VDN that routed the call to an Xported station with over 50 bridge appearances.	121883	
SIP calls dropped after 15 minutes when the session refresh timer expired because Communication Manager was unable to parse multiple parameters in the SDP FMTP line.	121894	
On a conference call in the Lecture mode on Avaya Aura Conferencing 7, there was no video on all the H.323 one-X Communicator stations that were part of the conference.	121923	
Agents on SIP CC stations with MCH remained in the After Call Work mode after releasing a held ACD call.	121929	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 *4 of 13*

Problem	Keywords	Workaround
A call that was blind transferred dropped on the called party after three minutes.	121937	
There was no video on a call that was made from an H.323 registered one-X Communicator phone to Radvision MCU via Radvision iView IVR (Interactive Voice Response). Then, the call dropped.	121969	
On Communication Manager, there was no two-way talkpath on calls made to agents using IP stations. This happened when the agent IP station used a legacy port network for VoIP resources, and the agent was supposed to hear an alert tone for the incoming call.	121970	
There was no talkpath on a call that was transferred using REFER by Voice Portal over SIP trunk with Direct Media on to a queued call.	121972	
Soft buttons were not updated on the 96xx station when a call was transferred using transfer-on-hangup FNE and a new call was made.	121978	
SAC failed for an R2MFC call made to a SIP station.	122014	
Occasionally, auto exclusion did not remove service observers.	122022	
When the Special Application feature SA8797 CTI Agent Call Capture by FAC was enabled, ASAI could not log in an agent on a CTI station.	122037	
There was no talk path when far-end did not support UPDATE and sent OPTION in the dialog, and the call involved the sending of display Re-INVITE after OPTIONS processing.	122062	
A call that terminated on an agent with skill level set to 2 and Redirect on No Answer enabled dropped	122069	
An incoming SIP trunk call made to Communication Manager via Session Manager was conferenced on Communication Manager. After the first party dropped from the call, the display of the calling party station was updated with the name of the HUNT group instead of the member connected to the call even when the **ISDN/SIP Caller Display:** field was set to mbr-name on the HUNT group.	122086	
The ZIP tone failed on SIP Refer calls.	122096	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 *5 of 13*

Problem	Keywords	Workaround
An IGAR call was made to a SIP integrated Modular Messaging system. The voice-mail greeting was cut off when SA9112: Sequential IGAR Call Setup was enabled.	122102	
A call did not cover to voice mail when Microsoft Exchange Server 2010 was used as the voice mail server.	122112	
The SIP phones did not display the calling party information until the call was answered.	122114	
Occasionally, Communication Manager reset.	122115, 122146.	
The call log showed `Unavailable` after a transferred conference call was answered and dropped.	122123	
The media gateway registration activity interfered with the Synchronization Over IP feature when a large number of slave media gateways were being resynchronized due to the addition or deletion of a master clock or tandem clock.	122132	
A call that was forwarded to an off-net forwarded destination was dropped and routed to the coverage point.	122163	
For processor-channel applications such as CMS and AUDIX, a burst of incoming data traffic caused a buffer overload condition that resulted in a temporary loss of communications with the application adjunct (session/ socket bounced). For the CMS, this caused a pump-up to occur when communications were restored. The CMS link traffic bursts were the result of SIP BSR polling with measurements enabled on the associated VDNs.	122184	
When a SIP station transferred an incoming call over an ISDN trunk, CDR generated incorrect data.	122197	
A corrupted filesync.conf file prevented the filesync program from running. This resulted in critical files becoming out-of-date because changes on the active main server were not transfered to the duplicate main server and the survivable core and survivable remote servers.	122206	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***6 of 13***

Problem	Keywords	Workaround
When a monitored service observer joined a call, the ASAI connected event reported an incorrect number instead of the number of the called party. This happened when the VDN Override feature was enabled.	122219	
There was no video on a video call made from a Cisco E20 phone to ADVD.	122223	
Customers could not remove the 2420 or 4624 station set type when the Display Language field set to unicode and the Display Character Set field was set to Katakana on system-parameters country-options screen and then changed to Roman and then again changed back to Katakana.	122225	
On the MUSIC/ANNOUNCEMENTS IP-CODEC PREFERENCES screen, the following fields were enabled: ● Prefer use of G.711 by IP Endpoints Listening to Music? ● Prefer use of G.711 by IP Endpoints Listening to Announcements? Due to the above administration, an IP call that normally uses G729 used G711 to listen to music or announcements. A call was put on hold while listening to music and announcements. When the call was reconnected, there was no talkpath.	122233	
Communication Manager reset or interchanged when a large number of service observing calls were made.	122240	
The system reset due to internal software trap in Communication Manager.	122241	
Communication Manager did not play ringback to the calling party of a call when the call was made to an H.323 station that was logged off and had EC500 administered but not enabled.	122242	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***7 of 13***

Problem	Keywords	Workaround
A system consisted of two Communication Managers (CM1 and CM2) and one Session Manager. The Session Manager connected the Communication Managers via SIP trunks. The **Preferred Minimum Session Refresh Interval** field for SIP trunk of CM1 was set to 300, and the **Preferred Minimum Session Refresh Interval** field for SIP trunk of CM2 was set to 900. A call made from CM1 to CM2 dropped after 600 seconds. CM1 negotiated Min-SE:1800 and CM-A sent UPDATE after 300 seconds from ACK for INVITE. CM2 replied 422 to the UPDATE and the call dropped after 300 seconds from the 422 response.	122247	
When a VDN with VDN override routed to another VDN, any service observer attached to the second VDN was not connected to the call.	122251	Set the VDN override to yes on the first VDN.
ICC boards that are slow to initialize in an H.248 Media Gateway caused conflict board minor alarms.	122253	
Occasionally, all ISDN PRI trunk calls failed due to internal software resource exhaustion.	122269	
Occasionally, customers with Enterprise LDAP (Lightweight Directory Access Protocol) servers observed manual changes overwritten after Communication Manager rebooted.	122277	
A station in a corp admin PBX was used to call VDN. The VDN routed the call to a OneX Communicator SIP phone. The SIP phone then transferred call to another SIP phone. When NCR was enabled, there was one way talk path after the call was answered.	122278	
On Communication Manager, a held SIP station call did not drop when it received VoIP from an H.248 media gateway, and the media gateway moved to another server due to a signaling link outage. User intervention was required to remove the held call appearance from the SIP station.	122284	
A call that covered off-net over a SIP trunk after covering to a coverage answer group dropped.	122294	
SIP trunks stopped functioning after a Third Party Call Control transfer.	122298	
When the /var/log/ecs/commandhistory log reached its file size limit it would overwrite the current file instead of beginning a new log file.	122299	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***8 of 13***

Problem	Keywords	Workaround
A SIP call using Direct IP-IP audio caused the system to reset.	122300	
In the case of downstream forking of a call, incorrect handling of the SIP CANCEL message caused the system to reset.	122301	
When the Verint device registered and unregistered several times, the Verint call recorder that executed an ASAI single step conference caused the call that was getting recorded to drop.	122304	
A One-x Mobile call initiated as callback failed when the incoming call handling treatment was applied on the incoming trunk.	122310	
When a call was covered to Modular Messaging using an alphanumeric handle, Communication Manager sent incorrect information in Contact and P-Asserted-Id in the reINVITE SIP message.	122324	
An Avaya Aura Call Center transfer failed when the transfer target tried to forward the call back to Avaya Aura Call Center.	122327	
The **Override ip-codec-set for SIP direct-media connections?** field on the **system-parameters ip-options** screen was not presented to the administrator after logging onto SAT using a customer login. Earlier, the field was located on page 4 of the command which was only presented to Avaya logins. Now, it is located on page 2 which is presented to all logins.	122328	
In a 5.2.1 environment, an H.323 phone was used to call a SIP phone (SIP1). The H.323 phone then initiated unattended transfer to another SIP phone (SIP2). Both SIP1 and SIP2 were registered to a SBC. The H.323 phone then completed the transfer. When NCR was enabled, the call dropped after 32 seconds.	122343	
An adjunct did not receive a route end for a call that terminated to an outgoing trunk.	122348	
There was a segmentation fault that caused an interchange when the **`list station`** command was run.	122356	
Users of one-X Client Enablement Services were unable to use the call log for calling back national and international numbers.	122358	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***9 of 13***

Problem	Keywords	Workaround
Outgoing trunk calls that were transferred by an agent who was service-observed generated a CDR record showing the agent as the originator and a duplicate record showing the service observer as the originator.	122371	
Call shuffling failed over an H.323 trunk when the ISDN messaging sequence involved receiving a Setup Acknowledge message followed by the Progress message.	122375	
There was one-way talkpath when a call across a SIP trunk was put on hold and then released by both parties.	122384	
The link to reporting failed when unmeasured calls went through an unmeasured VDN for BSR polling.	122386	
After deactivation, Service Pack could not be removed.	122390	
The self-administering of EC500 failed when the first route in route pattern had FRL (Facility Restriction Level) greater than station FRL.	122397	
A call was made from a one-X communicator SIP phone (SIP1) to another SIP phone (SIP2). SIP1 tried to transfer the call to a third SIP phone (SIP3) that had logged off. The transfer failed, and there was no talk path between SIP1 and SIP2 when Direct Media and Shuffling was enabled.	122407	
The off-hook alert was too long for IP stations.	122413	
Initially, all the SIP trunk groups towards Session Manager were only outgoing. A SIP phone was used to call another SIP phone. The call failed because there was no incoming trunk group. When the SIP trunk groups were changed to two-way, calls still failed.	122414	
An agent was on a conference call with two external parties and created that conference from a call that was first held. IQ ignored the call.	122437	
Telecommuter call between a one-X agent and ADVD dropped because of tandem media mismatch.	122444	
When predictive dialing was used over a SIP trunk, the **+** digit was not prefixed to the calling number.	122445	
Calltype analysis did not work for incoming R2MFC trunks even when SA8904 was enabled.	122446	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***10 of 13***

Problem	Keywords	Workaround
On a system, MOH was off and Direct Media was on. A call was made involving three SIP phones: SIP1, SIP2, and SIP3. SIP3 had the Bridged Appearance of SIP2. The call was made from SIP1 to SIP2, and was answered on SIP3. SIP1 put the call on hold, and, immediately after that, SIP3 put the call on hold. After SIP2 bridged on the call, SIP1 could not release the call and the call dropped after 32 seconds.	122450	
The route of the outgoing invite header/record route header/contact header had inconsistent port number or transport type on the evolution server. This caused call failures.	122463	
When over 3000 agents were logged into the same skill, agents could not answer calls and the calls continued to ring at the stations for several minutes.	122465	
When a connection manager denial event was logged, running the list trace station command caused the system to reset.	122486	
Occasionally, Communication Manager reset when an H.323 phone tried to register to Communication Manager.	122487	
There was Communication Manager license server core dump on the VMware platform.	122488	
A call dropped on a system that was administered to clear callr-info after leaving ACW. Agents with Enhanced CallrInfo Display did not consistently receive the Enhanced CallrInfo Display while in timed ACW.	122489	
Call processing stopped in the vector collect step when the incoming digits were sent over an IP trunk via out of band messages and the IP trunk was translated to receive digits in band.	122492	
Calls made to an unstaffed agent that were answered by a coverage point dropped after a few minutes.	122494	
An EAS Call Center with Callr-Info administered to clear when leaving ACW has an agent administered with Enhanced Callr-Info Display. When the agent pended the AUX workmode and released the call prior to the customer, the Callr-Info Display appeared and remained on the agent station when the call dropped.	122495	
The use of entity tags was disabled in the web server.	122506	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 *11 of 13*

Problem	Keywords	Workaround
Customers could not see all inter-connected network-regions when running the **`list measurements ip dsp-resources summary`** command after activating a patch using the call-preserving steps.	122508	
A VDN with a VDN return destination queued the call to a SIP agent, and the SIP agent transferred the call. When the call dropped, the caller did not go to the VDN return destination.	122519	
The ringback tone disconnected after the first coverage point when the coverage point was a coverage answer group and the principal was an X-ported station.	122523	
A media processor was not properly programmed when a SIP forked call was routed over an ISDN trunk.	122541	
SIP trunk routing previously optimized for Communication Manager 6.0.1 could result in failed calls after an upgrade to Communication Manager 6.2, which required manual reconfiguration of SIP trunks to resolve.	122543	
The **`reset ip-stations`** command skipped IP endpoints when shared control stations were registered.	122544	
Calls made to a SAC station did not ring on bridged stations.	122555	
The SAT interface stopped functioning when the status media-processor command was run.	122559	
When a call was made from an H.323/DCP station to MSUM (Microsoft Unified Messaging), the call did not terminate to MSUM.	122560	
LAR did not work when the Coverage after Forwarding and Coverage off-net features were enabled.	122577	
SRTP calls with **Initial INVITE with SDP for secure calls** disabled failed.	122583	
When a Radvision XT endpoint disconnected from an established two party call, the other party could hear the reorder tone instead of being disconnected.	122599	
The line appearance on the principal phone disappeared when it was put on hold and the bridge phone bridged in.	122605	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***12 of 13***

Problem	Keywords	Workaround
Communication Manager did not use the correct value of Session Refresh Timer.	122606	
In a two-party call, the system played clipped greeting on Avaya Aura Messaging.	122612	
Calls were made from a SIP phone to a VDN. Occasionally, when an agent answered the call, the station did not display the VDN.	122616	
LDAP users in certain groups did not get the proper shell initialization files during account creation.	122624	
Communication Manager did not send the SIP UPDATE message when H.323 FAC message was received from the H.323 trunk. This caused the called party information on the originating station not to update after a transfer at the far end.	122670	
Communication Manager delayed ringback by 10 seconds on an incoming R2MFC call that was forwarded by the far end.	122688	
The SMI backup pages allow a leading space in the destination field. This caused the backup request to fail on the Backup Now page, and a leading space broke the ability to change or delete on the Schedule Backup page.	122697	
Customers received a Resolved Alarm trap when they should not be getting one.	122705	
For a call, the display was not updated on the stations of members of the pick-up group when there were more than 25 members in the group	122740	
A VDN had the **Allow VDN Override?** field set to yes and the **Display VDN Name for Route-To DAC** field set to yes. This VDN routed to another VDN that had the **Display VDN Name for Route-To DAC** field set to no. A call went through the first VDN and the second VDN queued the call to an agent. When the agent transferred the call and the transfer completed, the transferred-to station displayed the caller information for the second VDN instead of the agent that transferred the call.	122742	In the second VDN, have the call use a vector route step to get to the skill instead of a vector queue step to get to the skill.
Communication Manager reset while processing H.323 video calls.	122745	

Table 13: Fixes delivered to Communication Manager 6.3 and 6.2 SP5 ***13 of 13***

Problem	Keywords	Workaround
In the survivable core server and survivable remote server setups, incoming endpoint registration messages were not captured in the MST trace file.	122758	
There was no talkpath when an attended transfer was performed on a call involving Communication Manager and CS1000 SIP endpoints.	122761	
In Call Centers with SLM (Service Level Maximizer) on multiple skills, an agent stopped receiving calls for an extended period during an agent surplus situation.	122768	
A video call between an ADVD and a content-enabled device such as LS-1030 and HDX-SIP could not be put on hold. This happened only when the call was made from the LS-1030 or HDX-SIP device.	122783	
While checking the VoIP capacity on a media gateway that has 320 channels installed, the following message appeared after the media gateway had temporarily unregistered: `Note: The gateway is registered with a Communication Manager version which limits DSP resources to 240 channels.`	122870	
REINVITE with SDP + display change was split into REINVITE with SDP and then UPDATE/REINVITE with display change. For a direct IP call, Communication Manager tandemed REINVITE as a single REINVITE for both SDP and display change.	122885	
AST 2 transfer failed.	122908	
Calls made over H.323 trunks had no talkpath.	123095	
Calls that were made over H.323 trunks did not have talkpath after they were put on hold and then released.	123097	
Calls could not be made from a SIP Radvision MCU to an ADVD.	130188	

Problems fixed in Communication Manager 6.3

Table 14: Fixes delivered to Communication Manager 6.3 ***1 of 11***

Problem	Keywords	Workaround
Remote access and telecommuter calls using QSIG over SIP trunks (QSIP) did not work.	100896, 112182.	
The following SAT commands did not handle 2-digit product major versions for registered IP endpoints: ● list registered-ip-stations ● list tti-ip-stations ● status station	103292	
The calling-party information was not displayed on the terminating SIP endpoint when the call was made from AAC in an ADVD audio and video conferencing scenario.	110926	
The agent deskphone displayed incorrect information after LAI was triggered.	111047	
LAR failed when a call was made from the call log.	111379	
A SIP trunk call made from Session Manager to Communication Manager failed when the first signaling group was busied out and the second and third signaling groups were in service.	111865	
The endpoints of the pickup group members rang constantly when Session Manager failed or when a SIP endpoint logged in and there was already one pickup call ringing.	113266	
Occasionally, one or more network regions were not disabled when the survivable server was active and the **Force phones and gateways to Active Survivable Servers** field was set to y.	120317	
QSIG auto-call-back calls failed when the Automatic Route Selection digit tables were involved.	120427	
When IP ports were unavailable, port records got corrupted and caused issues with registration.	120892	
The system displayed No valid license is installed when either the Communication Manager license or the Communication Manager Messaging license expired.	121228	

Table 14: Fixes delivered to Communication Manager 6.3 *2 of 11*

Problem	Keywords	Workaround
The IPSI Connection Up Time (min): field on the **system-parameters port-networks** screen had an initial value of 0 instead of being blank.	121309	
Customer-created SMI access profiles were not correctly restored during a template upgrade.	121387	
TN2501 announcement boards with an assigned ethernet port failed Periodic Test 1511 when the board had different ethernet options settings than what was administered for the board in the **change ip-interface** screen. When this happened, the test stopped abrubtly. This caused the associated process to stop functioning after five minutes. Also, the system ran out of resources and Communication Manager restarted.	121449	
Communication Manager dropped the call when a mismatch between Offer and Answer SDP for iLBC codec mode occurred.	121677	
For telecommuter calls, the DTMF tones could not be heard on the service link.	121928	
An alarm will be raised if the duplication link between duplex servers is not 1 GigE.	122121	
Proc errors were observed when a call was made with SA8891 enabled.	122148	
The display update was not sent to the members of a pickup group when there were more than 25 members in the pickup group and the total number of stations administered as members of pickup groups exceeded 255.	122192	
There was no display on the called station for incoming ISDN-PRI trunk calls that had CALLING PARTY NUMBER IE but no calling party number and were transferred or routed through VDNs to a station type that performs tagging.	122239	
Communication Manager conformed to obsolete draft-ietf-sip-replaces-01 instead of RFC 3891.	122365	
The **Native Name** field on the **status station** screen was not synchronized with the corresponding field on System Manager.	122042	
g3rapt and g3pkrpat MIB Groups did not return data for all route patterns that were administered on the Route Pattern Measurement Selection screen.	122433	

Table 14: Fixes delivered to Communication Manager 6.3 ***3 of 11***

Problem	Keywords	Workaround
Memory corruption lead to data corruption for DECT stations.	122455	
The Request Uniform Resource identifier of the Refer-To: header was created using the From header for a SIP trunk that had Network Call Redirection enabled for blind transfer calls. This caused the call transfer to fail.	122475	
There was no talk path on an H.323 trunk call made between two Communication Manager systems using wide-band codec when a single-step conference party joined in.	122510	
When a call covered to voicemail, the caller was prompted with a generic greeting instead of going to the voicemail box.	122553	
There were internal memory errors due to H.248 media gateway operation.	122571	
The **`count`** parameter was unavailable for the **`list off-pbx-station-mapping`** command.	122591	
Customers using ISDN, H.323, or SIP trunks ran out of internal Communication Manager resources. This caused ineffective call attempts on ISDN, H.323, or SIP trunks.	122415, 122655.	
Calls dropped due to the parsing error in the Call-Info header of generic parameters.	122678	
Incoming DPT calls on an H.323 trunk failed if SRTP was enabled on the ip-codec-set screen.	122689	
Communication Manager generated core dumps and reset when the value of the Message Lamp Ext field on the station screen was set to the extension of the attendant.	122692	
The call appearance of an end point stuck when the attendant dropped the call and single-step conference was involved in passive mode.	122739	
Phones got stuck in the discover mode when an agent attempted to log in using a VDN extension number.	122743	
The ASAI link went down on a busy switch. There is a vector trace sent for every route request that was cancelled. This filled up the message buffers and the switch went down.	122767	

Table 14: Fixes delivered to Communication Manager 6.3 ***4 of 11***

Problem	Keywords	Workaround
A call made to an agent, when dropped by the caller, did not drop all the parties in the call if no-hld-conf was invoked by the agent and the conference party did not answer.	122771	
The system displayed all ports on a TN2602 board as out-of-service when the **`status media-processor board`** command was run even though all background and demand tests passed. There were no errors or alarms logged against the board or any of the ports.	122775	
When the **`status station`** command was run, the system did not display the **Alternate Gatekeeper List** page when the **Alternate Gatekeeper List** field was set to 0 for an IP network region.	122805	
The NATL/INTL prefix was not included in the CDR even when SA9099 was enabled for EC500-mapped calls.	122826	
Late PRACK for 183 provisional response caused a SIP call to drop when UPDATE reaches the far-end first.	122828	
When ProVision was used to administer a SIP adjunct message center hunt group, the system rebooted. Also, using ProVision and other System Management tools could cause the system to reboot.	122866	
IGAR calls failed when incorrect authorization digits were transmitted to the far-end.	122882	
The called station displayed its own extension after the call was answered.	122884	
A user on an analog phone was on a call with the attendant. The user attempted to put the call on hold. The hold request was denied, but the ISG did not receive an event_abort message which would NAK the request.	122894	
Customers were unable to make adjunct route requests, and the system displayed PROC_ERR to indicate that all CRVs (Call Reference Value) were allocated.	122899	
Customers could not duplicate stations that had custom button labels on buttons 11 and 12 on Button Module 1 of the Station screen for telephones that support custom button labels.	122909	

Table 14: Fixes delivered to Communication Manager 6.3 *5 of 11*

Problem	Keywords	Workaround
Occasionally, dual ring back could be heard for calls made over SIP trunks.	122913	
Occasionally, warm restarts occurred when media gateways were removed within an hour of running the `enable mg-return` command while the **Force Phones and Gateways to Active Survivable Servers?** field was set to **y** on the **system-parameters ip-options** screen.	122915	
A call went through a vector that had a collect step, a route-to step that performed LAR, and a wait step. The prompting timer was shorter than the wait step. When the route-to step failed, the caller heard the intercept tone and the call did not continue with vector processing.	122928	Configure the prompting timer in a way that it is longer than the wait step.
The ASAI drop event was not sent when a transfer failed and the agent who was transferring the call was monitored.	122929	
Trunk to trunk transfer failed for SIP stations.	122931	
When Special Application SA9114 was activated and an EVNT_UORIG was sent to the ISG, an IAP application query for the called party number failed. This caused the default trunk value of ##### to be sent to the IAP application instead of the dialed digits.	122935	
A generic greeting was heard when a multilength dial was administered and the call was routed to voice mail by using a VDN vector.	122956	
Calls made to VDN failed when the name had double quotes.	122960	
Calls made from endpoints such as CISCO E20 that do not send bandwidth information in SDP did not receive video. Only audio calls could be made.	122962	
There was no talk path on a call made to One-X agent using a SIP trunk in the telecommuter mode.	122971	
A TTS H.323 station stopped functioning.	122979	
Network Call Redirection using E.164 numbers and a vector variable did not send the preceding plus sign (+) that is required to redirect the call back to the network that it came from.	122980	

Table 14: Fixes delivered to Communication Manager 6.3 *6 of 11*

Problem	Keywords	Workaround
Ephemeral caching was disabled on a system. When Direct Media was enabled, call pickup among SIP stations failed.	122985	
The display for an 8434D set type was not cleared under the following conditions: 1. A call was made to an extension with multiple bridged appearances appearing on the 8434D. 2. The 8434D had SAC active. 3. The 8434D had per button ring control administered. 4. The Display Information With Bridged Call? feature was enabled on the system-parameters features screen. 5. The 8434D had a call appearance or bridged appearance active on another set. 6. A call to the bridged appearance was dropped after the call went to coverage.	123011	
Customers were unable to register IP agents in the shared control mode.	123019	
On Communication Manager, multifrequency signaling trunks using Russian shuttle protocol did not work when the trunk ports used belonged to an H.248 media gateway.	123020	
When SIP Direct Media was enabled, features such as call forward and EC500 that need collection failed when Communication Manager had a media gateway administered with it and ephemeral caching was disabled.	123027	
The call log was not updated on the DCP (14xx) phones when the called-party station had call forward activated on it.	123039	
The incorrect called-party number was sent in the ASAI alerting event for a call that was made to a monitored VDN and the associated vector had a converse-on step.	123049	
The correct digits for the called party were not sent in the originated message for outbound calls to an ISDN PRI trunk using media gateways. Also, there was delay in reserving the resources.	123051	
Occasionally, ringback was not heard on a call over an IP trunk.	123057	

Table 14: Fixes delivered to Communication Manager 6.3 *7 of 11*

Problem	Keywords	Workaround
An ISDN trunk call was made to Communication Manager that was routed to VP (Voice Portal). The call was routed to an agent station, and the station displayed `UNKNOWN NAME` after the agent answered the call.	123073	
Calls made to a station that had SAC activated did not cover to Messaging.	123093	
Calls could not be made when the One-X Communicator SIP phone was in the telecommuter mode and LAR was used to end the call on the cellular phone.	123096	
LAR failed when calls were made from the call log.	130006	
The display of an STE (Secure Terminal Equipment) BRI station was not updated when the station was busy on a call and received another call on a second line appearance. The second call was diverted to voice mail. The station should have turned off the second line appearance that received the second call and updated the display with the current call information. Instead, the station ignored the Info message and continued to display the outdated information for the diverted call.	130031	
EC500 calls that were made using AAR and ARS to route were not completed successfully when the AAR/ ARS feature access code was not administered.	130044	
When a measured SIP call was made with international formatting, the calling-party's number was not sent correctly to Reporting.	130050	
Slow ping test responses, such as test 1387 on IP signaling groups, that take longer than 4 seconds caused an error peg that could generate an alarm. This alarm could not be cleared because of the slow ping test response error peg.	130051	
The line appearance was stuck when the ARS code was dialed using the idle line appearance FNE.	130130	
The station displayed the team button label incorrectly when the station was set to the user-defined language and the team button was administered to display the name of the monitored station as the label.	130137	
After the transferred-to party answered a call, the call state was not updated on the active party.	130138	

Table 14: Fixes delivered to Communication Manager 6.3 ***8 of 11***

Problem	Keywords	Workaround
Users were unable to change their security code using TAE (Telecommuting Access Extension).	130139	
Communication Manager set the wrong signaling group into the bypass state after a TLS (Transport Layer Security) certification error.	130153	
The telephone displayed the incorrect calling number for a PSTN call when it was transferred by ASAI to an agent from Voice Portal.	130165	
An attempt to bridge on a call was denied for a SIP endpoint even when the principal endpoint had Exclusion enabled with call held.	130175	
Call logs showed invalid calling party names (invalid.unknown.domain or NO-CPName).	130183	
On systems that did not have auto-hold, the IQ agent reports showed Auto-In agents with no MOH skills as Idle instead on Initiating when the agents ended calls by pressing a new call appearance.	130195	
A DSP fault on an H.248 media gateway caused overload on Communication Manager. This happened because Communication Manager reallocated resources from the same media gateway, thus affecting call service.	130198	
When a SIP message of more than 9216 bytes in size was received, Communication Manager reset.	130219	
There was no talk path between the principal and the originator over an IP trunk when the principal and the bridge appearance of the principal endpoint used a different codec and network region to the IP trunk.	130265	
There was no talkpath when calls were made from an IP station to a SIP 1XC telecommuter.	130278	
Dial Plan Transparency failed when a softphone user registered with the shared-control mode. Dial Plan Transparency does not consider the shared-control softphones while determining whether a call follows the Dial Plan Transparency configuration for an endpoint.	130301	
Scheduled backups on the standby server failed when the **Save ACP translations prior to backup** radio button was checked and the **Run Now** button on the **Schedule Backup** SMI web page was also checked.	130344	

Table 14: Fixes delivered to Communication Manager 6.3 ***9 of 11***

Problem	Keywords	Workaround
When SA9114 was enabled, ASAI sent dialed digits when a station was administered as an international number, but dialed it using a UDP extension that was shorter than the international number.	130357	
Occasionally, Communication Manager resets lead to a reload.	130362	
The system displayed an error when the asaiuui type variable was added on page 15 for variable IS, a value for the collect type was entered in row A on page 1, and the value of the start field on page 15 was set to a number greater than 16.	130363	
When an IP Agent registered in the share control mode, the application froze and used all available occupancy.	130404	
Occasionally, H.323 stations could not register to systems with TN2501AP circuit packs.	130407	
An agent was unable to log on to a DCP phone when call recording was enabled from a third-party tool by using Device Media Call Control.	130417	
Outgoing calls that use overlap sending did not send originated events.	130433	
The UUI entered in a vector was lost when an LAI call made to QSIG trunk failed and vector processing started on the call again.	130446	
Occasionally, when an agent transferred a call, the telephone displayed the wrong softkeys.	130504	

Table 14: Fixes delivered to Communication Manager 6.3 ***10 of 11***

Problem	Keywords	Workaround
Communication Manager had certain vulnerabilities that are described in Avaya Security Advisories: ASA-2011-430 ASA-2011-375 ASA-2012-008 ASA-2012-043 ASA-2012-126 ASA-2012-094 ASA-2012-156 ASA-2012-136 ASA-2012-145 ASA-2012-199 ASA-2012-117 ASA-2012-196 ASA-2012-138 ASA-2012-166 ASA-2012-170 ASA-2012-208 ASA-2012-231 ASA-2012-238 ASA-2012-147 ASA-2012-250 ASA-2012-286 ASA-2012-285 ASA-2012-300 ASA-2012-325 ASA-2012-301 ASA-2012-362 ASA-2012-394 ASA-2012-344 ASA-2012-414 ASA-2012-397 ASA-2012-387 ASA-2012-429 ASA-2012-411 ASA-2012-430 ASA-2012-433 ASA-2012-386	112990, 112991, 113267, 120198, 120199, 120422, 120459, 120533, 120535, 120536, 120537, 120539, 120542, 120904, 120727, 120920, 121125, 121130 121137, 121300, 121436, 121463, 121608, 121655, 121656, 121713, 121714, 121737, 121864, 121865, 121866, 122040, 122041, 122043, 122048, 122050,	

Table 14: Fixes delivered to Communication Manager 6.3 *11 of 11*

Problem	Keywords	Workaround
ASA-2012-398 ASA-2012-441 ASA-2012-479 ASA-2012-471 ASA-2012-461 ASA-2012-434 ASA-2012-489 ASA-2012-524 ASA-2013-033 ASA-2012-064 ASA-2013-043 ASA-2013-066 ASA-2013-053 ASA-2013-064	122232, 122234, 122235, 122236, 122265, 122385, 122442, 122737, 122864, 122897, 130081, 130082, 130083, 130084.	

Problems fixed in Communication Manager 6.3 and Avaya Video Conferencing Solutions

Table 15: Fixes delivered to Communication Manager 6.3 and Avaya Video Conferencing Solutions *1 of 2*

Problem	Keywords	Workaround
The **`list trace station`** command for SIP video did not show the negotiated video codec.	120193	
There was no video on a call that was answered from a bridged Flare endpoint.	122750	
Calls between Flare Experience for Windows and XT H.323 endpoints encountering packet loss and jitter caused the remote video on Flare to blink periodically.	A79/ NGUE-14952	
There was only audio on ADVD calls to Radvision Elite 5000/6000 via IVR or Auto Attendant.	A96/ ADVD-9964	
Point-to-point calls made between Radvision endpoints and Avaya endpoints dropped when they used different audio codecs.	R39/ QC15046	

Table 15: Fixes delivered to Communication Manager 6.3 and Avaya Video Conferencing Solutions *2 of 2*

Problem	Keywords	Workaround
When an audio endpoint was used to coordinate a conference with two video endpoints, all endpoints had two-way talk path and the video endpoints had two-way video. When the audio endpoint was used to add another audio endpoint, there was no video on the call, but all the endpoints had two-way talk path.	121594	

SIP Trunk Capacity Guidelines

The following maximum SIP trunk capacities apply when Communication Manager Release 6.3 or later is installed on the system:

Communication Manager Evolution Server Environment.

- For general business use or for non-24x7 Call Centers with moderate call traffic: 12,000 SIP trunks.
- This maximum applies to the main server in a Communication Manager configuration. Strictly to configure redundant trunks in support of a fail-over, up to an additional 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 Sales Factory Design Center. Call Center designs that involve SIP trunking must go through Sales Factory. For more information, see the document titled Avaya Aura® Communication Manager and Call Center Release 6.3 SYSTEM SOFTWARE BASED CAPACITES, available at http://support.avaya.com.

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, might 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

Known problems in Communication Manager 6.3

This release includes the following known issues in Communication Manager 6.3.

Table 16: Known problems in Communication Manager 6.3 ***1 of 6***

Problem	Keywords	Workaround
If Communication Manager Messaging is configured for SRTP and the far-end doesn't offer SRTP, Communication Manager Messaging will not answer the call.	5336	Administer Communication Manager Messaging to RTP (non-SRTP) if far-end (endpoint, incoming trunk call from RTP environment) does not support SRTP.
In rotary analog stations, the inter-digit collection timer may expire too soon, preventing dialed calls from completing successfully. The workaround is the only solution to this issue since no Communication Manager software change has been planned.	101096	On the **`system-parameters features`** screen, page 6, there is a field called, **Short Interdigit Timer** (seconds). The default value of this field is 3 seconds. Increasing this value can fix this problem.
Communication Manager 6.x LSP servers cannot register with Communication Manager Main servers that are prior to the 5.2 release. If the LSP registers with a Communication Manager 5.1.2 or earlier Main server, you may need to enter the serial number of the media gateway 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 the Communication Manager software.	101016	

Table 16: Known problems in Communication Manager 6.3 ***2 of 6***

Problem	Keywords	Workaround
A migration backup that was passphrase-protected on Communication Manager 5.2.1 where pre-upgrade patch 02.1.016.4-18793 was loaded could not be restored on Communication Manager 6.x unless quotes were put before and after the passphrase. This issue has been fixed in the latest pre-upgrade patch for upgrading from Communication Manager 5.2.1 to Communication Manager 6.x. The patch name is 02.1.016.4-19401.tar.gz, and it is available at http://support.avaya.com and PLDS.	111855	
Path Replacement does not work with Private numbering format for QSIG/SIP interworking. This also affects path replacement on a Communication Manager-Communication Manager Messaging QSIG trunk for the Messaging Transfer feature. The workaround is the only solution to this issue since no Communication Manager software change is planned.	113124	Change the numbering format from **Private** to **Unknown**.
After an interchange, the newly active server can experience call failures and occupancy spikes to overload levels. The occupancy prior to the interchange is 57% ST+CP or greater.	113197	
A call made from a 96xx SIP phone on Communication Manager 5.2.1 with RTP/SRTP enabled to a 96x1 SIP RTP phone on Communication Manager Release 6.2 or later with direct media enabled and CapNeg off drops immediately upon answer. This problem is resolved on the Communication Manager 5.2.1 side by applying service pack 12.01 (19751) or later.	101218, 120129, 120205.	Either turn off IP video on SIP signaling group to Session Manager on Communication Manager Release 6.2 and later, or remove 1-srtp from ip-codec-set on Communication Manager 5.2.1.
A 2004 IP phone on Communication Server 1000 calls an 1140 IP phone on a Business Communication Manager. If the 1140 IP phone blind transfers the call to a 96xx SIP phone, there is no talk path.	120170	

Table 16: Known problems in Communication Manager 6.3 ***3 of 6***

Problem	Keywords	Workaround
S8300D main servers running Communication Manager with an unsupported medium or large memory configuration will be prevented from upgrading to Communication Manager Release 6.3 and later. S8300D survivable servers running Communication Manager in an unsupported medium or large memory configuration will automatically be converted to a small memory configuration during the upgrade to Communication Manager Release 6.3 and later. Medium and large memory configurations are not supported on an S8300D server, but previously administrators were not blocked from configuring these memory configurations. See PSN 100127 for further information. **Note:** Survivable remote servers with a small survivable memory configuration can act as survivable servers for main servers with a large, medium or small memory configuration.	130445	All embedded (S8300D) Communication Manager main servers incorrectly configured with a large or medium memory configuration must be retranslated into small memory configuration before upgrading to, or having translations restored to, Communication Manager Release 6.3 and later.
IGAR is not supported on SIP endpoints.	130565, 130571, 130844.	
CM-A and CM-B have a QSIG trunk between them with **QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path?** set to yes and **Diverted Party Identification** set to principal for both switches. SIP phone A1 on CM-A calls B1 on CM-B which has call forward active to SIP phone A2 on CM-A. SIP phone A2 has cover-no-answer active to a sip-adjunct hunt-group which points to Avaya Aura Messaging or Communication Manager Messaging. If A2 does not answer the call forwarded from B1, the caller (A1) will reach the messaging mailbox for A2 instead of B1 as expected.	130582	

Table 16: Known problems in Communication Manager 6.3 ***4 of 6***

Problem	Keywords	Workaround
Two Stub Network Regions (SNR1 and SNR2) each have a direct WAN link to the same Core Network Region (CNR1) but use different codec sets (no common codec). CNR1 also has a direct WAN link to another Core Network Region (CNR2) with the same codec set as one of the SNR links. With this scenario, a SIP call between SNR1 and SNR2 has no talk path. If the link between CNR1 and CNR2 is removed, there is SIP talk path between SNR1 and SNR2.	130632	
Communication Manager should not allow endpoints to bridge onto a call when the Whisper Page feature is active. However, if Session Manager Multi-Device Access is in use, other SIP devices which are sharing an extension through parallel forking can bridge onto the whisper page call and have two way talk path with the paging extension.	130897	
During deployment of the Communication Manager 6.3 Duplex vAppliance, the second vNIC labeled **Asset** is the Communication Manager duplication link and should be appropriately linked to the customer network. **Note:** After deployment this link can be found as "Network Adapter 2" within the Virtual Machine's properties and can be edited or linked from this location.	NA	

Table 16: Known problems in Communication Manager 6.3 ***5 of 6***

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. **Note:** The basic steps outlined in the workaround are included in the connection preserving upgrade instructions for duplex servers in the document titled *Upgrading to Avaya Aura® Communication Manager 6.3*, which is available at http://support.avaya.com.	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.

Table 16: Known problems in Communication Manager 6.3 *6 of 6*

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	

Known problems in Avaya Video Conferencing Solutions

This release includes the following known issues in Communication Manager 6.3 and Avaya Video Conferencing Solutions.

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *1 of 7*

Problem	Keywords	Workaround
Point to point calls between Radvision H.323 endpoints and Avaya SIP endpoints result in G.711 audio.	A22/122211	Administer G.711 codec on the Communication Manager ip-codec-set list.
Point-to-point calls between Radvision H.323 endpoints and Avaya endpoints do not support mid-call features, such as hold, resume, and transfer and may drop video or the call.	A27/121568	

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *2 of 7*

Problem	Keywords	Workaround
Far End Camera Control (FECC) does not work on point-to-point calls between Radvision H.323 endpoints and Avaya SIP video endpoints that support FECC.	A28	
There is no video on calls made from One-X® Communicator H.323 to Scopia Elite MCU over a SIP trunk.	A61/121975	Configure to use H.323 trunk as described in the Quick Setup Guide.
Video calls between Radvision VC240 and Flare Experience for Windows may result in low-resolution video.	A89/ SCAE-2403	On the Radvision VC240 web client, select **Configuration** > **Call Quality**, and set NetSense support to off.
There is no content-sharing between Radvision XT and Avaya 1000 Series endpoints for point-to-point calls and calls made via Elite MCU.	R1	
SIP outdialing from Scopia Elite MCU uses the wrong SIP domain.	R4	Upgrade to iVIEW 8.0 or use this workaround to change default SIP domain on iVIEW 7.7: Manually add the default domain to the following file on the iVIEW ==> c:\Program Files (x86)\RADVISION\iVIEW Suite\iCM\jboss\bin\vcs-core.properties "vnex.vcms.core.conference.defaultDomain=<domain>", where <domain> is the SIP domain for your system environment. Then restart the iVIEW Graphical User Interface.
SCOPIA Elite MCU shows SIP connection to iVIEW as down, but calls can be made successfully.	R6	Upgrade to iVIEW 8.0.
iVIEW does not strip the prefix digits for outbound calls from iVIEW to Communication Manager.	R13/ QC19493/ QC15404	Follow the admin steps in the Quick Setup Guide.

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *3 of 7*

Problem	Keywords	Workaround
There is intermittent audio quality when Siren audio codecs are used for calls between Avaya 1000-series endpoints and the SCOPIA Elite MCU.	R14/AGS-289	Ensure that the Siren codecs are not in the Communication Manager ip-codec-set list.
Calls made from Radvision SCOPIA Elite MCU to Avaya SIP endpoints drop after 30 seconds.	R15	At the initial install, ensure that a functional FQDN is used for the Radvision iVIEW installation as per Radvision documentation. If FQDN is not configured, then reinstall it.
Avaya 1000-series calls made to Radvision XT1200 fail when G.729/G.729A is in the Communication Manager audio codec list other than first position.	R75/ QC18567	Set G.729 and G.729A in the first position of the Communication Manager ip-codec-set list, or remove it from the ip-codec-set list.
(Avaya Video Conferencing Manager) AVCM allows endpoint discovery up to a /24 subnet (254 endpoints max or smaller subnet).	147	AVCM will not discover the endpoints, but instead manually enter them.
When upgrading the 1000 Series Endpoints "Upgrade License expired(15)" message may be displayed.	254	Ignore the message. Licensing is not required on the 1000 Series endpoints.
Sequential blind transfer of 10x0 endpoints may drop video.	255	If video is required after the transfers, drop and make a direct call.

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *4 of 7*

Problem	Keywords	Workaround
After a Session Manager outage, 1010/1020 may take up to 30 minutes to re-register. Incoming calls are blocked while unregistered, but outgoing calls are accepted and immediately initiate registration.	260	When you see a red SIP box in the bottom right hand corner of the 1010/ 1020 screen, try manually registering by making an outgoing call or perform the following steps: 1. Log in to 1010/ 1020 as admin. 2. Select **Communications**. 3. Select **SIP** and enter your login credentials, and enter the IP address of the Session Manager system you have to register to. 4. Click **Register**.
1030/1040/1050 may transmit higher bandwidth than requested. Occasionally, this can cause 5+ party conferences to fail on 1050.	288	Administer 1040/1050 endpoints to send no more than 2M video.
Calls from Windows Flare Experience to ADVD with H.263 do not establish video. The hold and release operations drop the call.	130041	Enable H.264 on the ADVD endpoint in the ADVD Settings File.
HDX H.323 calls to AV10X0's is audio-only in a Multi Communication Manager configuration.	122851	Set DTMF rtp payload.
RMX dial-out to AV1010/20 leads to one-way video (Connect with Problem).	AVA-1551	Use dial-in on RMX.
HDX SIP to One-X Communicator H.323 or SIP gives audio-only connection if H.239 is enabled.	ONEXC-5548 /130801	Upgrade to One-X Communicator Release 6.1 Service Pack 8. Alternatively, turn off content sharing with HDX 3.1 or stay with HDX 3.0.4.

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *5 of 7*

Problem	Keywords	Workaround
One-X Communicator does not get video with XT5000 embedded MCU.	ONEXC-6627	This interop is currently not supported with FP2.
ADVD may show severely distorted video with XT5000 embedded MCU.	A87/ ADVD-9909	This interop is currently not supported with FP2.
Radvision MCU dialout calls to Avaya SIP endpoints using the H.323 protocol, for example, dialing the outbound call using a mismatched protocol type, results in the call flowing over the H.323 trunk to Communication Manager instead of the SIP trunk to Session Manager. The call flow results in an audio-only call.	A92	When you are creating endpoints on the iVIEW suite, assign the matching protocol type: SIP to SIP stations and H.323 to H.323 stations.
Radvision MCU dialout calls to an H.323 One-X Communicator endpoint using the SIP protocol, for example, dialing the outbound call using a mismatched protocol type, results in the call flowing over the SIP trunk to Session Manager instead of the H.323 trunk to Communication Manager. The call flow results in CIF video.	A93	When you are creating endpoints on the iVIEW suite, assign the matching protocol type: SIP to SIP stations and H.323 to H.323 stations.
There is no audio and video on calls made from Elite MCU 6000 to One-X Communicator, and the calls drop after 30 seconds.	A108/ ONEXC-7003	Configure One-X Communicator to use dial-in to the Elite MCU 6000.
No warning is currently given when a Zone Prefix is used on the iVIEW Gatekeeper administration screen for a Communication Manager entry that matches with either a Service Prefix or the leading digits of an assigned Virtual Room. Matching prefix entries can cause calls to route incorrectly and the call will fail.	R118/ QC20634	Ensure that Zone Prefix for the Communication Manager Gatekeeper entry does not match any of the Service Prefixes or the leading digits of any of the Virtual Rooms.
Parties joining an active conference call on the MCU that has **ALL** muted join the conference with active audio.	R123/ QC20032	
iVIEW8 does not show stats for SIP participants on initial view of the stats pop-up window.	R136/ QC21009	The screen can be updated by either closing the meeting room details pop-up window and bringing up a new one or by selecting "More Information..." under the "Action" drop down menu on the endpoint details.

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *6 of 7*

Problem	Keywords	Workaround
Adding a new Communication Manager gatekeeper via Scopia Management may not update Scopia ECS.	R157/ QC21263	Manually update Scopia ECS to route calls to the new Communication Manager gatekeeper.
An audio call made to an Elite MCU 5000 or Elite MCU 6000 cannot be changed to a video call.	QC21476	Make a video call from a Flare device.
When using Siren codecs on a Lifesize endpoint with **Override ip-codec-set for SIP direct-media connections** set to yes on page 2 of the **change sys ip-options** screen on Communication Manager, the 1050 can be limited to 4-party conferences if any of the Lifesize endpoints have Siren codecs above G.722 and G.711 in their priority list.	130531	Make sure Siren codecs are below G.722 and G.711 in the Lifesize codec priority list. The list is accessed on the Lifesize endpoint at **System Menu** > **Administrator Preferences** > **Audio** > **Audio Codec Order**.
One-X Communicator or ADVD blind transfers to HDX SIP fails to connect and drops after 30 seconds when BFCP (H.239) is enabled.	130722	Disable BFCP (H.239).

Table 17: Known problems in Communication Manager 6.3 and Avaya Video Conferencing Solutions *7 of 7*

Problem	Keywords	Workaround
Environment: Communication Manager duplicated systems that have an H.323 signaling group to a remote direct routing gatekeeper. e.g. Radvision's ECS. The H323 signaling group uses discovery LRQ's and a near end listen port of 1719. Problem: After a software upgrade and processor interchange, the H.323 signaling group is no longer able to process calls.	130774	Perform the following steps: 1) Perform the Pre Update/Upgrade Step on the active server. 2) Upgrade the standby server. 3) Interchange the servers. The connection to Radvision ECS gatekeeper is now broken. 4) Busyout the signaling group. 5) Run the **`change signaling-group XX`** command. 6) Set **Near-end Listen Port** and **Far-end Listen Port** to 1720 and **LRQ Required?** to n. Commit these changes using the **Submit** key. 7) Run the **`change signaling-group XX`** command. 8) Set **Near-end Listen Port** and **Far-end Listen Port** to 1719 and **LRQ Required?** to y. Commit these changes using the **Submit** key. 9) Release the signaling group. The connection to Radvision's ECS gatekeeper now works. 10) Upgrade the new standby server.

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 gathered 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.

Appendix A: Acronyms

3PCC	Third Party Call Control
AAC	Avaya Aura Conferencing
AAR	Automatic Alternate Routing
ACD	Automatic Call Distribution
ACW	After-Call Work
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
AVCM	Avaya Video Conferencing Manager
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
CID	Caller Identification
CIE	Customer Interaction Express
CIF	Common Intermediate Format
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
CRV	Call Reference Value
CS1K	Communication Server 1000
CSS	Center Stage Switch
CTI	Computer Telephony Integration
CUCM	Cisco Unified Communications Manager
DAC	Direct Agent Calling
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
ECFB	Enhanced Call Forwarding Busy
ECFU	Enhanced Call Forwarding Unconditional
EMU	Enterprise Mobility Users
ES	Evolution Server
ESS	Enterprise Survivable Server
EWT	Expected Wait Time
ETSI	European Telecommunication Standards Institute
FAC	Feature Access Code
FECC	Far End Camera Control
FNE	Feature Name Extension
FRL	Facility Restriction Level
FS	Feature Server
HDX	A Polycom high definition video room system

HEMU	Home Enterprise Mobility User
IAC	International Access Code
ICR	Intelligent Customer Routing
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
IVR	Interactive Voice Response
J24	Avaya Digital Terminal for Japan
LAN	Local Area Network
LAI	Look Ahead Interflow
LAR	Look Ahead Routing
LDAP	Lightweight Directory Access Protocol
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
MCH	Multiple Call Handling
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
RONA	Redirect on No Answer
RTCP	RTP Control Protocol
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
SDP	Session Description Protocol
SEMT	SIP Endpoint Managed Transfer
SES	SIP Enablement Services

SIF	Source Input Format
SIP	Session Initiation Protocol
SO	Service observer
SMI	System Management Interface
SSC	Single Step Conference
SSH	Secure Shell
SSHD	Secure Shell Daemon
STE	Secure Terminal Equipment
SVNS	Simple Voice Network Statistics
TAC	Trunk Access Code
TAE	Telecommuting Access Extension
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TEG	Terminating Extension Group
TLS	Transport Layer Security
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
URN	Universal Resource Name
USNI	United States Network Interface
USB	Universal Serial Bus
UUI	User to User Information
VALU	Value-Added
VCS	Video Conferencing Server
VDN	Vector Directory Number
VEMU	Visitor Enterprise Mobility User
VLAN	Virtual Local Area Network
VOA	VDN of origin Announcement
VoIP	Voice over Internet Protocol
VP	Voice Portal

VSST	Virtual Server Synchronization Technology
VSX	A Polycom standard definition video room system