Avaya Aura® Communication Manager 6.3.118.0

Release Notes

Issue 1

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Changes delivered to Avaya Aura® Communication Manager 6.3.118.0

Communication Manager 6.3.118.0 Release Notes

Communication Manager Release 6.3.1.0 and later uses the following services pack naming convention. This is a four digit number format as described in the following example:

Communication Manager 6.3.4.1, where

- 6 - major release field (Communication Manager Release 6)
- 3 - minor release field (Communication Manager Release 6.3)
- 4 - service pack field (Communication Manager Release 6.3 Service Pack 4)
- 1 - special release field, typically used for a re-issue of an existing service pack (Communication Manager 6.3 Service Pack 4.1)

Note that:

1. To avoid confusion, unused fields to the right might not be shown. For example, Communication Manager 6.3 will be used in documentation related to the minor release instead of Communication Manager 6.3.0.0
2. The special release field may be used for atypical software releases other than service pack re-issues which will be explained in the documentation for the special release software (e.g. release notes or Product Correction Notices).
3. This naming change applies only to regular Communication Manager service packs and does not apply to special service packs such as Security Service Packs, Kernel Service Packs, Pre-Upgrade Service Packs and VMware Tools Service Packs.
4. Communication Manager service pack file names will be unaffected by this naming change. For example, Communication Manager 6.3 service packs will still have file names with the Communication Manager GA load string and a unique five digit identifier like:03.0.124.0-12345.tar.
5. The service pack version information displayed on a running system will not change and will still show the Communication Manager service pack file name format like:03.0.124.0-12345.
6. This naming change does not apply to service packs for Communication Manager Release 6.2 and earlier which will follow existing naming formats.

Communication Manager releases and service packs are cumulative, and all changes in the previous service packs are included in Communication Manager 6.3.x.x. Changes delivered to the Communication Manager 6.3.x.x are grouped as follows:

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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

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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.
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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.

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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 FW 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 6.3.2.0

Table 1: Enhancements delivered to Communication Manager 6.3.2.0

Problem	Keywords	Workaround
The Calling Party conversion screen is enhanced to introduce a new column named Incoming number format, and support to enter any in the CPN Prefix field has been added.		
A new field, Invoke ID for USNI Calling Name, is added to page 3 of the ISDN trunk-group screen. The system displays the new field when the trunk-group field is set to isdn with Carrier Medium set to pri/bri or atm, and the Supplementary Service Protocol field is set to b. When the value of the new field is set to variable, then a new Invoke ID is selected each time the USNI Calling Name is sent to the far end. If the value of the new field is set to fixed-1, then the Invoke ID will be fixed as the number 1. This is required for interoperability with some equipment provided by other providers.	130481	
When Communication Manager runs in a VMware environment, each time Communication Manager VMware reboots, information about memory assigned to the VMware, CPU resources, and hard disk space assigned to the VMware is sent to the syslog and it shows up in the /var/log/messages folder	130871	
Communication Manager, Call Center, and Communication Manager Messaging license usage data is now sent to WebLM.	130936, 131440	
This is an enhancement to the GRIP 3587/4742 - Mute speakerphone when in shared control with Avaya one-X® Communicator (1XC) feature that was delivered to Avaya Aura Feature Pack 1. With this enhancement, the deskphone is not muted in an ASAI initiated Single step conference while in the shared control mode with OneX Communicator.	131072, 131422	
When OPS mapping is created for a dual registered H.323 station, the call limit is synchronized with the number of call appearances administered for the station.	131109	
This is a new Message Tracer Analyzer version 6.4.5.3 that includes following: • Correction of CMS messages • Parsing of multi-digit r2mfc messages • Notifications of Internal Call Process and the Call Record fields • Parsing of the ASAI endpoint registration/de-registration message	131744, 131890	
Video SRTP will be supported with OneX Communicator Release 6.2. For more details, see OneX Communicator Release 6.2 release notes.		

Enhancements delivered to Communication Manager 6.3.6.0 (FP4)

Table 2: Enhancements delivered to Communication Manager 6.3.6.0 (FP4)

Problem	Keywords	Workaround
The RAS Limit Threshold has now been increased from 50% to 65%. When the CPU now reaches 65% occupancy IP phone registrations will be throttled.	131503	
The number of Tenant Partitions has been increased from 100 to 250 without having to turn on the special application (SA8993).	131664	
Transferred calls to One-X CES controlled extensions will now show the original calling party in the call log instead of the party that transferred the call.	132502	
It is now possible to select a stronger certificate request signing algorithm on the Certificate Signing Request SMI page.	140116	

Enhancements delivered to Communication Manager 6.3.7.0

Table 3: Enhancements delivered to Communication Manager 6.3.7.0

Problem	Keywords	Workaround
Video SRTP will be supported with Scopia 8.3 SP1.		
Communication Manager now does not send asterisk (*) to the OneX-CES call logs.	131353	
H323 phones capable of Transport Layer Security (TLS) can now establish a TLS connection to a CLAN board.	140401	

Enhancements delivered to Communication Manager 6.3.8.0

Table 4: Enhancements delivered to Communication Manager 6.3.8.0

Problem	Keywords	Workaround
Avaya one-X® Communicator in the telecommuter mode will now display the actual calling number on the telecommuting extension instead of the Avaya one-X® Communicator number like it used to.	132408	
A SIP visiting user that makes an emergency call can now be reached through the PSTN call back.	140826	
Calls between IP Office v9.1 and Communication Manager will now be compliant with Special Application 9122.	140868	
The auto keyword for the **For Toll Compliance, Treat As** field on the trunk group screen has been added for Toll compliant administration of trunks on Communication Manager.	140929, 141184	

Enhancements delivered to Communication Manager 6.3.9.0

Table 5: Enhancements delivered to Communication Manager 6.3.9.0

Problem	Keywords	Workaround
The following fields on the off-pbx configuration set screen are now enabled for one-x, mobile-onex, and callback-onex configuration sets: • CDR for Origination • Post Connect Dialing Options • Barge-in Tone • Provide Forced Local Ringback for EC500	140752, 140945	
Duplicated Processor Ethernet for SIP is now obsolete and will no longer be available on Communication Manager.	140914	
The **`list trace hunt-group`** command will now print additional information to aid Avaya services to troubleshoot problems involving calls to agents.	141063	
Communication Manager will now perform a server interchange to release memory that is incorrectly held up in specific SIP call scenarios to prevent the system from getting into a state where further SIP calls cannot be processed.	141225	
The auto keyword is now an option for the **For Toll Compliance, Treat As** field on the trunk group screen.	141227	
A new field **Location to Route Incoming Overlap calls** is now available on the off-pbx configuration screen with trunk or station as values.	141237	
The use of embedded certificates is now removed.	141328	
Communication Manager is now RFC4040 compliant.	141339	

Enhancements delivered to Communication Manager 6.3.10.0

Table 6: Enhancements delivered to Communication Manager 6.3.10

Problem	Keywords	Workaround
The SSLv3 option has now been removed from the Avaya Aura Platform Communication Manager System Maintenance Interface.	141623	

Enhancements delivered to Communication Manager 6.3.11.0

Table 7: Enhancements delivered to Communication Manager 6.3.11

Problem	Keywords	Workaround
Communication Manager server interchange/restart/reset design strategy is being changed for duplicated server pairs. See PSN020191.	150044	
The filesync operation between Communication Manager servers will now only use TLSv1.	150177	
Communication Manager could sometimes incorrectly manage internal resources used for SIP calls. To recover, Communication Manager will now perform a software level 1 reset, or a server interchange. See PSN020047u.	141453	
Communication Manager is updating the "Simple Network Management Protocol (SNMP)" functionality to Net-SNMP. See PSN020171.	141106	

Enhancements delivered to Communication Manager 6.3.111.0

- Starting in CM Release 6.3.111.0, The Avaya Aura® Communication Manager updated the SNMP stack/engine to use net-SNMP.
- The G3-MIB was retired and replaced with two new MIBs:
  - The AVAYA-AURA-CM-MIB
  - AVAYA-AURA-CMALARM-MIB

The Avaya Aura® Communication Manager SNMP Renewal Quick Reference Guide outlines the changes that were made to the SNMP processes, SNMP administration, and SNMP SMI Pages. The Quick Reference Guide can be downloaded from Avaya Support

Enhancements delivered to Communication Manager 6.3.117.0

Table 8: Enhancements delivered to Communication Manager 6.3.117

Problem	Keywords	Workaround
Enabling "Block trunk seizure upon Busy Indicator button press" will not seize the trunk upon pressing "busy-indicator" button administered for trunk but will just show the trunk status.	15639	

Problems fixed in Communication Manager 6.3.2.0

Table 9: Fixes delivered to Communication Manager 6.3.2.0

Problem	Keywords	Workaround
DTMF could not be sent over a SIP trunk if the DTMF payload type was IN-BAND or Out-of-Band or RTP and PAUSE was required.	111735	
When a VDN service observer was observing a call and the call was transferred to a party that had the Can Be Service Observed? field set to no on the Class of Restriction screen, the service observer was not removed from the call.	120240	
Occasionally, there was one-way talk path on SIP calls that involved SRTP and EC500.	121260, 131438.	
There was wideband audio quality for calls made between Avaya SIP endpoints and Radvision XT endpoints. This was due to DTMF mode mismatch.	122111	
Orphaned TTI ports on the system caused the system to run out of ports. New TTI merges and PSA associates were denied because there were no ports available.	122983	
Occasionally, the **`monitor bcms system`** command did not show any data.	130157	Run the **`monitor bcms system 1-8000`** command.
Conference display was shown on a transferred call when SoftFlare was used to transfer a station to a held station.	130215	
The SIP network call redirection feature sent NCR REFER back to the party that initiated the transfer instead of the party that was on the call.	130223	
The display on bridged stations was not updated when a consult transfer was completed.	130261	
Call Admission Control did not apply for SIP to H.323 calls when Direct Media was enabled.	130315	
On a call made from Aastra to Communication Manager over Country Protocol 1b/1d (Telcordia), the endpoint on Communication Manager displayed the calling-party name and number. But on a call made from Communication Manager to Aastra over the same trunk, the endpoint on Aastra displayed only the calling-party number.	130361	
A Parallel-Forked Device could not be used to perform the following: • Deactivate Exclusion. • Bridge onto a Held call that had Exclusion deactivated The Parallel Forked Device was able to bridge onto a group-page call.	130383, 130580, 130885.	
A bridge appearance endpoint was unable to perform the Hold operation on the call when the call was already put on hold by the principal endpoint.	130395	
There was no video on a video call that was made from an Avaya one-X® Communicator H.323 endpoint on Communication Manager to another Avaya one-X® Communicator H.323 endpoint on another Communication Manager over a SIP trunk.	130430	
When the length of the calling-party number was greater than 13, Communication Manager truncated the calling-party number instead of removing the plus (+) sign.	130482	
The calling-party number was prefixed with an international access code	130506	

Problem	Keywords	Workaround
from the trunk location when a station and a trunk were on different locations and the incoming call was of type national.		
The value of the **Force Phones and Gateways to Active Survivable Servers** field on the **IP-Options System Parameters** screen could not be set to y. When the value of the field was already set to y, the changes could not be submitted to the Media Gateway screen. The system displayed the following error: `All MGs with the same BACKUP SERVER must have the same recovery rule`	130557	
Exclusion did not function properly on an endpoint when the 1XMobile SIP Dual Mode feature was activated.	130585	
After performing a handoff to the cellular One-X Mobile, a user on an iOS could not release the call.	130606	
There was no talkpath for calls made between stations in different Stub Network Regions (SNR) with no common codec.	130632	Perform one of the following: • Use common codec from SNRs to CNR. • Remove the connectivity to CNR-1. • Remove Media resources from CNR-1.
A conference call involving bridged appearances of various parties dropped when one party in the call dropped and the remaining parties were put on hold.	130657	
Occasionally, Communication Manager did not send the ISDN Presentation Restricted when Per Station CPN - Send Calling Number was restricted.	130673	
The **SMI Network Configuration DNS Domain** field allowed invalid Domain Names to be inserted in the / etc/hosts file. This caused failures in failover instances on duplicated servers.	130768	
The logged-in agent hunt group audit could run only the first 1500 logged-in agents of a particular skill. When there were more than 1500 agents logged into a skill, the hunt group audit did not run properly.	130818	
On RadVision H.323 video endpoints, when a mid-call feature such as Hold, Transfer, or Conference is activated on video calls, video is not re-established on the call.	130831	
AACC could not dial Feature Access Codes that start with a pound (#) sign on the SIP station.	130879	
A dual registered (DR) Flare iOS endpoint and an H.323 endpoint were being used. The DR Flare iOS endpoint was used to make a video call to a SIP station. The DR H.323 endpoint then bridged onto the call. When the DR Flare iOS endpoint disconnected the call, the call dropped.	130893	
Communication Manager profiles were not properly restored during a migration from 5.2.1.	130901	
Communication Manager restarted when a 96xx SIP endpoint performed the Hold operation on a call.	130947	

Problem	Keywords	Workaround
When two or more Multiple Device Access (MDA) devices were on a call and one MDA device activated Exclusion, Communication Manager sent the BYE message followed by a PUBLISH (Dialog State Event Notification) message to the MDA device. When Session Manager received the PUBLISH message before the BYE message, the MDA device that was dropped from the call displayed an idle call appearance instead of an active bridged call appearance.	130969	
The History Info messages generated in the invite message were different when the invite message had VOA and when the invite message did not have VOA.	130972	
After a Busyout followed by a Release operation on a DS1 board, Communication Manager sent a service acknowledgement message with an out-of-service indication on some of the PRI trunks right after the service-in service message had been sent. Even when Communication Manager sent additional Restart messages to the B channels, some vendor ISDN implementations did not process the requests properly. This rendered some trunks out-of-service until service and in-service messages were sent by Communication Manager.	131002	
Calls were stuck on the standby trunk when Digital Enhanced Cordless Telecommunications was forced back to the main server.	131053	
Occasionally, the CMS link dropped.	131065	
When encountering CAC limitations and call coverage on the called SIP station, the SIP caller did not hear call progress tones for around 50 seconds.	131077	
There was no talkpath on a SIP endpoint that was a whisper page group member.	131084	
An H.323 endpoint registered to an ESS got the incorrect IP address of the primary server in the Alternate Gatekeeper list. This caused the H.323 endpoint to fall back to the incorrect IP address.	131091	
A conference call hosted on an H.323 integrated multipoint control unit (MCU) was interrupted with MOH when one of the conference participants performed the Hold operation on the call.	131108	
Communication Manager reset on certain types of transfer operations, such as blind transfers.	131114	
A Flare endpoint was used to make a call to another Flare endpoint, and Music on Hold was enabled. One party on the call performed the Hold operation. The window of the endpoint that was used to perform the Hold operation still popped up allowing video operations. Ideally, after performing the Hold operation, the endpoint should not display the window.	131116	
The endpoint that was used to answer a pickup-group call displayed the trunk name instead of `Anonymous` when the incoming trunk call had no CPN.	131119	
Incoming Call Handling Treatment was applied to the calling numbers even when the SIP signaling group was administered to be in the Evolution Server mode.	131125	
Customer could not disable CDR1 and CDR2 on page 2 of the survivable-processor screen.	131128	
There was no video on video calls made between endpoints from unrecognized vendors or unrecognized video-endpoint models.	131129	

Problem	Keywords	Workaround
A SIP video endpoint was used to make a call to a Dual Registered (DR) extension. An audio-only DR H.323 endpoint was used to answer the call, and then a DR iOS Flare endpoint bridged onto the call. When iOS Flare escalated the call to video, there was no video on the call and the call dropped after 32 seconds.	131149	
Persistent intermittent port-network connectivity failures caused an overload condition that resulted in trunk groups going out-of-service.	131156	
Queued calls from ICR were not dropped automatically after the Session Establishment timer expired.	131157	
An outbound call transferred to an agent via hunt group showed only ANSWERED BY and no extension on the endpoint.	131165	
Occasionally, all ISDN PRI trunk calls failed due to internal software resource exhaustion.	131166	
When Communication Manager received two Hold REINVITE messages with a change in the SDP version, it did not send back the response.	131174	
Calls made from the attendant to an extension that were forwarded to the attendant override call forwarding when Chained Call Forwarding was active.	131189	
Occasionally, Communication Manager underwent reload.	131193	
Occasionally, attempting to send a call to an agent caused the CMS link to go down.	131195	
The **IMS Feature Sequencing** field was enabled when the station type was changed to a type that does not support IMS Feature Sequencing.	131210	
The display on a bridged appearance was not updated when a Facility Message with the Calling Party Name information was sent after a delay since the initial SETUP message.	131215	
An H.323 IP endpoint remained in the out-of-service state after a call on a media gateway went into the connection-reconstruct mode and then dropped.	131219	
A video SRTP-enabled SIP endpoint was used to make a call to a dual-registered (DR) extension. A video SRTP-enabled DR Flare endpoint was used to answer the call, and two-way video was observed on the call. A DR audio-only H.323 endpoint bridged on to the call. Depending on the SIP phones involved in the call, no video and one-way video was observed.	131228	
Occasionally, H.323 endpoints did not migrate to the ESS when the network region was disabled.	131233	
With the **Override ip-codec-set for SIP direct-media connections?** field on the **change system-parameters ip-options** screen set to y and only none given in the Media Encryption section of the ip-codec-set, calls between two Flare endpoints established with audio encryption, but no video encryption.	131236	
Call Admission Control did not apply to a call made from a SIP endpoint to an H.323 endpoint when Direct Media was enabled.	131240	
On Communication Manager, heavy call load on H.248 media gateways caused the gateways to become unstable, resulting in unpredictable call behavior.	131245	
There was a segmentation fault on Communication Manager during	131248	

Problem	Keywords	Workaround
duplicate Processor Ethernet server interchange.		
When a call has to be made from an H.323 Avaya one-X® Communicator endpoint to an H.323 Radvision Elite 5000/6000 endpoint on an H.323 trunk, the caller can either dial into a video conference directly or via an IVR. There was audio and video on the call, but when mid-call operations such as hold were performed, the call was rendered audio-only.	131255, 131269, 131274.	
Calls were dropped when G.723-5.3K was configured, Shuffling was enabled, and Direct Media was disabled.	131256	
In a non-EAS environment, the hunt group members are unable to receive calls when a hunt group is changed from ACD to non-ACD.	131258	Remove the ACD hunt group and add it as non-ACD.
An ASAI redirection to a hunt group that is set up to be a SIP adjunct for MM was not acknowledged. But, it worked. The next request was denied because the domain control association was stuck.	131259	
XEN migration set is enabled on VE systems.	131260	
When an incoming R2MFC call that was made to an endpoint from a cellphone mapped to a EC500 station had ECF (Enhanced Call Forward) unconditional enabled to a SIP station, and if the SIP station did not answer the call, the call did not go to coverage of the endpoint that had ECF unconditional activated on it.	131268	
Any administration change using the change **ip-network-region** screen corrupted the backup server table on a previously administered server. This caused the Split Registration feature to not function correctly because the feature relies on the backup server tables for information to make network region auto disable and auto return decisions.	131285	
An SRTP call made to a TCP-registered CapNeg endpoint rang only on the bridged call appearances.	131286	
A meet-me paging call could not be answered from an IP trunk.	131298	
The SA8146 redirect display was incorrect for calls that were forwarded to a VDN with announcement vector steps.	131325	
Occasionally, large SIP messages were not parsed correctly. This resulted in truncated SIP headers.	131327	
When 128 simultaneous station firmware downloads occur, Communication Manager got into a state where new downloads requests were rejected. Phones that were rejected were not queued up again, and a station firmware download schedule did not complete successfully.	131339	
Administering the **Block Exclusion Event Notification** field on the Class of Restriction screen was blocked based on the Call Center Release number.	131346	
SA9124 enhancements did not work for ASAI 3PCC merge requests. The default trunk identifier was used.	131348	
For calls made over a SIP trunk to a VDN, the caller endpoint displayed the VDN name and number irrespective of the value of the ISDN/SIP Caller Display field in the hunt group screen.	131349	
Incoming trunk calls to a SAC station that was bridged on a DECT station failed to cover to MM.	131372	
An H.323 audio endpoint was used to make a call to an Avaya one-X® Communicator SIP endpoint on Communication Manager. The H.323 endpoint then transferred the call to a Polycom HDX endpoint on another	131386	

Problem	Keywords	Workaround
Communication Manager over a SIP trunk. The call dropped after the H.323 endpoint completed the transfer.		
A SIP call answered on a bridged call appearance did not have talkpath when SA8965 was enabled.	131397	
Occasionally, due to data corruption, legacy port-networks such as G650s went out of service. Data corruption could be caused by running the **`list trace station`** or the **`status station`** command on an IP endpoint that was on a complex call, such as a large conference or a group page call.	131405	
There was no ringback tone on calls received on Communication Manager through Session Border Controller and Intelligent Customer Routing.	131409	
When the system reset and the first IPSI was added to translations, the IPSIs did not start functioning until after the next system restart of Communication Manager.	131412	
CDR failed to record the access code dialed for LAR calls.	131421	
The Service Observing Next Call Listen Only feature could not be activated remotely.	131425	
After a Session Manager failover, the SIP phones that were behind an SBC and on the call had stuck line appearances.	131427	
The VDN name in UUI was displayed incorrectly for AAEP call transfers.	131428	
VuStats did not check tenant calling permissions while deciding whether a user can view information regarding an agent, trunk group, VDN, or hunt group.	131433	
When Send All Calls and OneX Block All Calls was activated, the caller was unable to leave Voice Mail messages.	131435	
Supervisor Assist did not check tenant calling permissions while deciding whether an agent can call the supervisor.	131441	
Q-Stats (Q-Time and Q-Calls) did not check tenant calling permissions while deciding whether a user can view information from the hunt group.	131442	
The Hold operation could not be performed on SIP endpoints that were configured with multiple media encryption policies and Communication Manager was filtering out the top encryption policy.	131455	
Communication Manager stripped the crypto attribute from video calls when the port was set to 0. Hence, endpoints could not be used join the AAC calls.	131457	
The bridged call appearance could not drop the call after bridging onto a call when the primary endpoint had performed the Hold operation on the call.	131460	
A call made to an EAS agent when redirected on no answer to a VDN failed to cover to voice mail.	131469	
The One-X Client Enablement Services server could not be used with Communication Manager when it was routed via Session Manager Release 6.3 or later.	131470	
ASAI 3PMerge as part of CSTA SST (single step transfer) to a cellphone failed.	131479	
There was corrupted talk path on SIP calls when non-default packetisation	131480	

Problem	Keywords	Workaround
time was used for audio codecs.		
When the second AES NICE logger observed the shared control endpoint, there was no talk path for the AES NIVE logger.	131501	
Calls made from a non-Avaya SIP endpoint dropped.	131519	
After a reset board command for a later vintage TN2602 board (Pacifica version), only half of the board's capability was used to set calls up.	131529	
When the second preference was chosen under the following conditions: • An EC500 or ONE-X call invoked ARS or AAR • The administered off-pbx number required a digit-conversion step • The first preference failed due to LAR then digit conversion did not occur, and the call was routed incorrectly.	131530	
The Genesys agent stopped functioning because an ASAI 3PCC answer request was not responded to. This happened because media resources were not available when the answer request was made.	131531	
While using a CTI application that included ASAI 3PCC commands on SIP endpoints, requests NACK'd with a CV of 111 - protocol error were observed.	131555	
A SoftFlare endpoint was used to make an audio call to an audio-only endpoint. After the answer was called, the SoftFlare endpoint escalated to video. The operation failed. When SoftFlare performed the Hold operation, it stopped functioning.	131556	
A trunk failure was observed, and the ASAI call offered message to a VDN was sent with no calling-party or called-party information.	131558	
Preserved H.323 trunk calls were dropped before the preservation time of two hours.	131559	
A Radvision XT 5000 endpoint was used to make a call to a LifeSize 1020 endpoint. The XT 5000 endpoint was then used to make a conference call between a LifeSize 1030 endpoint, a Flare endpoint, and an H.323 Avaya one-X® Communicator endpoint. The H.323 Avaya one-X® Communicator endpoint was dropped from the conference call after some time.	131568	
The system displayed the VE_BUF_FULL error when the collected-digit buffer was full.	131570	
Communication Manager restarted due to a limited SIP video memory leak.	131574	
Due to toll fraud restrictions (SA9122), Communication Manager blocked EC500 after answer when multiple trunks were present in the route-pattern to EC500.	131575	
The alerting message for a SIP endpoint logged in as an EAS agent did not follow VDN Override administration for the VDN that routed the call to the EAS agent.	131584	
On a SIP SRTP video call, the session type parameter was not sent during the Hold operation with Music on Hold enabled.	131587	
In media-gateway registration, announcement boards displayed `no board (list config media-gateway)` for several minutes after other boards were inserted.	131588	
Occasionally, calls made over a SIP trunk dropped when the SIP trunk	131593	

Problem	Keywords	Workaround
was used for routing to a telecommuter destination.		
When ROIF was enabled, Auto Exclusion did not remove the Service Observer for a manual-answer H.323 endpoint.	131595	
Communication Manager logs filled up with proc errors while using the ISAC (Internet Speech Audio Codec) codec, G.722.2, the iLBC (Internet Low Bitrate Codec), or the SILK codec developed by Skype.	131596	
A Communication Manager system (CM A) was routed to another Communication Manager system (CM B) through Session Manager, and the session refresh timer of CM A was less than the session refresh timer of CM B. CM B was connected to yet another Communication Manager system (CM C) by a SIP trunk that had Direct Media disabled. When an H.323 station (Station A) on CM A was used to make a call to another H.323 station (Station B) on CM B and Station B had an EC500 extension on CM C, both Station B and the EC500 extension alerted. When the call was answered on either Station B or the EC500 extension, the other stopped alerting and the call dropped.	131600	Enable Direct Media on the direct SIP trunk from CM B to CM C, or set the session refresh timer on CM A to a value greater than or equal to the value of the session refresh timer on CM B.
There was only audio on a video call made from a Radvision XT-H.323 endpoint to an Avaya one-X® Communicator SIP endpoint. The DTMF mode was RFC2833 for both the endpoints.	131624	
A SIP endpoint (SIP A) was used to call another SIP endpoint (SIP B). There was two-way talk path on the call. SIP A initiated attended transfer for an H.323 endpoint (H.323 C). Music On Hold was disabled. After SIP A completed the transfer, there was no talk path between SIP B and H.323 C.	131629	
When pound (#) is inserted before the digits of an outgoing call in a route pattern preference for a SIP trunk, the SIP INVITE has no digits.	131639	
VuStat values reset every 30 or 60 minutes depending on the administered measurement interval.	131644	
EC500 calls dropped when bridged appearances were administered on an IP DECT endpoint.	131645	
The endpoint displayed the name of an incoming SIP trunk call incorrectly when the username consisted of alphanumeric characters.	131648	
VP-MPP (Voice Portal) did not disconnect a call due to a lamp update received from Communication Manager. When VP changed its port to CTIACTIVE, and the port entered into CTI-only control mode, the call failed due to no CTI application.	131652	
Occasionally, Communication Manager reset during video calls on H.323 stations.	131654	
An SIP endpoint had features such as Bridged Call Appearance, Call Forward, Send Calls on an H.323 extension, and the Location field of the SIP endpoint on the IP Network Region screen was set to blank. During the button download of the H.323 endpoint, Communication Manager reset.	131657	
A SIP call could not be initiated because the CONN_M had a port in a bad state from a prior ASAI 3PCC merge involving a SIP endpoint that controlled the transfer.	131659	
A call was made from an Avaya one-X® Communicator H.323 endpoint to a Radvision XT5000 SIP endpoint. The XT4200 SIP endpoint then was used to call a XT5000 SIP endpoint and a three-party conference took place. The Avaya one-X® Communicator H.323 endpoint was dropped	131682	

Problem	Keywords	Workaround
within three minutes.		
Occasionally, there was no talk path on SIP calls that use SRTP.	131711	
Occasionally, a segmentation fault was observed on Communication Manager when an H.323 endpoint that had the EMU (Enterprise Mobility User) feature enabled had a bridged call appearance administered on the 24th button on the Station screen.	131714	
On a duplex server system, a system recovery that escalated to a Linux reboot did not complete and stopped before terminating all processes.	131720	
When an agent call with a bridged-call appearance was dropped, Communication Manager restarted due to an internal software trap.	131734	
There was no talkpath on incoming H.323 trunk calls. This happened when the signaling group of the trunk did not have Direct IP connections enabled.	131775	
When connection preservation was activated on call, a memory leak occurred and the transaction table filled up. Therefore, no more SIP processing could take place. This was observed only on systems that do not support UPDATE for session refreshes. This includes Communication Manager Release 6.0.1 systems. In Communication Manager Release 6.2, session refreshes are modified to use UPDATE instead of INVITE for refreshes. UPDATE handling does not encounter this problem.	131850	
When SIP downstream forking and reliable provisional responses were used simultaneously, the SIP transaction table filled up and SIP traffic was stopped.	131851	
A generic greeting was heard when a call that was made to a SIP endpoint covered to voice mail.	131959	
In a configuration where SIP messages associated with a call that was tandemed from a Communication Manager system to another over non-OPTIM SIP trunks, any one of the Communication Manager systems logged multiple UPDATE failures when the display name of the called party consisted of quotes. In some cases, the Communication Manager system reset.	131918	
ASAI Transfers and Conference operations from non-SIP stations that had EC500 or any other OPTIM feature enabled could not be performed.	131982	

Problems fixed in Communication Manager 6.3.2.1

Table 10: Fixes delivered to Communication Manager 6.3.2.1

Problem	Keywords	Workaround
When a principle station was active on a call and a bridged station attempted to originate another call, the bridged station was bridged on to the principle station's call.	132141	

Problems fixed in Communication Manager 6.3.3.0

Table 11: Fixes delivered to Communication Manager 6.3.3.0

Problem	Keywords	Workaround
An H.323 endpoint was used to make a conference call between an Avaya Desktop Video Device (ADVD A) and another Avaya Desktop Video Device (ADVD B). ADVD B used MMCS to include a third Avaya Desktop Video Device ADVD C (ADVD C) to the active conference. When the Conference button on ADVD B was pressed, an MMCS conference was established. All parties on all the endpoints could hear each other. On ADVD B, 2 contacts were shown: ADVD C and the Conference contact. After thirty seconds, the Conference contact was dropped from the spotlight. ADVD B had no moderator privileges and the remote operation buttons were unavailable.	122681	
When a call was answered on a bridged line appearance and then the principal endpoint was used to bridge on to the call, the monitored station with the Busy-indicator button did not light up with the busy alert of the principal endpoint.	130222	
A SIP phone displayed an incorrect message when it was used to log in an agent who was already logged in to another server.	130294	
Unadministered DS1 board warning alarms were not raised after Communication Manager was rebooted. This caused an inconsistency in the alarm system because when a DS1 board was inserted in the system and not administered, the system raised a Warning alarm. A system reboot clears all alarms, but when the alarms are still relevant, they should be regenerated.	130418	
Restricted Calling Party number did not function correctly when a call that had the Privacy set routed over a SIP trunk and tandemed over an ISDN or an H.323 trunk.	130694	
The endpoint displayed the incorrect calling-party number when an incoming SIP trunk call was tandemed over an ISDN trunk and the calling-party number was modified in the **tandem-calling-party-num** screen.	130750	
The display was not properly updated when Multiple Device Access (MDA) devices were on a conference call.	130867	
Incoming calls made from a cellular phone failed when Communication Manager had tenant partitioning enabled, the called endpoint and the EC500-mapped endpoint were in different tenants, and inter-tenant calls were restricted.	130951	
When an incoming SIP trunk call was mapped to an EC500 endpoint over an ISDN trunk, the calling number format was set to international even when the incoming calling number over the SIP trunk did not have a leading plus (+) digit in it.	130955	
Communication Manager was unable to tandem iLBC codec correctly to the called party.	131044	
Calls that routed using ARS or Calltype analysis to a pattern with two preferences where the first was unavailable and the second required an authorization code failed because the user was unable to enter the code.	131097	
A file descriptor resource leak caused sockets to stop working. No new sockets could be created, which is why calls made over on H.323 trunks failed, H.323 and SIP trunk groups could not go into service, H.323 stations could not be registered.	131140	
RPM installation failures in updates made the system inconsistent after a rollback attempt.	131151	
The **`list measurements tone-receiver detail`** command displayed the peak allocation values that exceeded the port network allocation.	131154	
A Polycom video endpoint on a Communication Manager system (CM 1) was used	131179	

Problem	Keywords	Workaround
to make a call to a Radvision RMX endpoint on another Communication Manager system (CM 2). The Radvision RMX endpoint is connected to CM 2 via an H.323 trunk. The Polycom endpoint is behind Video Border Proxy (VBP) which is connected to CM 1 via an H.323 trunk. After it was answered, the call connected as an audio-only call.		
Calls between 2 port networks or a port network and a media gateway failed when: • The PN cabinet was assigned to IP NR X • The PN consisted of a TN2302 or TN2602 media • Processor assigned to IP NR Y • The remote PN had a TN2302 or TN2602 assigned to IP NR Z or the remote gateway was assigned to IP NR Z • Connectivity was allowed between IP NR Y and IP NR Z and disallowed between IP NR X and IP NRZ	131314	
The endpoint displayed the wrong calling-party name when local calls were transferred to a VDN.	131324	
Calls made to a SIP agent who is in the Auto-Answer mode dropped.	131354	
Dial plan call-type with enbloc extension was unreachable from the VoiceMail button.	131400	
On a SAT terminal, the **status socket-usage** screen displayed a zero in the **Registered IP Endpoints with TCP Signaling Socket Established** field even when there were multiple registered H.323 stations with TCP sockets.	131451	
On a SAT terminal, the **status socket-usage** screen displayed a zero in the **Registered IP Endpoints with TCP Signaling Socket Established** field even when there were multiple registered H.323 stations with TCP sockets.	131451	
Incoming trunk calls made to a virtual station with coverage to a remote cover point failed and returned a busy tone.	131468	
Station users and call center agents observed the incorrect calling-party name and number when the user or agent was involved in a path replacement "trombone" trunk elimination operation.	131472	
Two calls were ringing for the same extension and the extension was bridged on to two other H.323 phones. When both bridged phones went off hook to answer the calls, then the endpoint that was used to answer the second call did not update the display.	131516	
Station A and Station B were configured as H.323 stations on Communication Manager. Station A had SAC enabled. Also, Station A was the bridged call appearance of Station B. When there was an incoming call on Station B, Station A displayed a visual alert only and no audio alert.	131538	
Occasionally, an MDA extension may be dropped from an AAC conference.	131551	
When a call was made to an IVR system, Communication Manager outpulsed the last digit twice when a call was routed using LAR.	131620	
When a customer used the SIP downstream forking and reliable provisional responses at the same time, the SIP transaction table filled up and stopped SIP traffic.	131621	
On Communication Manager, H.323 stations did not have talkpath on second call appearances when there were multiple bridges on both the primary and the secondary call appearances. The user switched from one active call appearance to another. This was observed when H.248 media gateways were used primarily for VoIP resources and ephemeral caching was turned off.	131627	
The `logmst` command did not display the full release string of Communication	131633	

Problem	Keywords	Workaround
Manager in the MST trace.		
Occasionally, agents did not hear the zip tone before a call connected to the customer.	131634	
Communication Manager did not accept new CES servers once it exhausts all ten slots even when one or more CES servers got decommissioned. With this fix, Communication Manager can have a maximum of 10 active CES connections at any given instant.	131637	
Occasionally, Communication Manager reset.	131665	
OneX Mobile was configured as No ring and connected on the first call-back call. When the deskphone received a second call, the call was extended to OneX Mobile even when No ring was configured.	131679	
Communication manager did not switch off the speaker phone when the Personal Station Access (PSA) feature was used.	131693	
Incorrect busy-indicator state was seen on the monitoring station when the monitored station had 2 calls, 1 in the ringing state and another in active call, and the ringing call was dropped.	131700	
On Communication Manager, the use of particular types of H.248 media gateways in an IP network region where G.723 is a preferred codec resulted in calls with no talkpath. The following H.248 media gateways do not support G.723: G450, G430, J4350, and J6350.	131704	
The **`list trace station`** command did not output the music source number when the call was put on hold.	131705	
customer could not change the Console Parameters screen.	131708	
An incoming call over a tie trunk where the calling party identity (ANI) is sent via DTMF tones did not complete successfully after it was sent to a VDN.	131716	
The CDR record was missing when an agent transferred an incoming trunk call back to an IVR.	131737	
The **FIPN_ISSLC** field displayed correctly on the **dialplan parameters** screen.	131742	
When an endpoint retrieved a call that was on hold at the coverage point, the ASAI drop event for the coverage party sent the wrong calling-party ID.	131748	
Occasionally, Communication Manager reset when SIP signaling group number 999 was on a call.	131752	
After some types of transfers by a SIP-connected server such as Voice Portal, subsequent agent transfers resulted in IQ reports showing HOLD times that were more than the actual HOLD times.	131766	
Under a heavy socket load, the system restarted.	131767	
Occasionally, the Blast Conference feature did not work for certain extensions.	131770	
A OneX Mobile user was unable to change the destination number.	131776	
When a non-SRTP IP phone was in a network region that had only an encrypted codec, there was no dial tone on the second call appearance.	131777	
A denial event was added to indicate an incorrect configuration when a service link and a bridged call appearance were configured on the same physical IP station.	131780	
A call that was made to an SSC (Single Step Conference) party and was blind transferred to an endpoint dropped.	131783	
When a special character was administered in the user name, the OneX Client Enablement Services (CES) logs displayed an incorrect caller name.	131785	

Problem	Keywords	Workaround
Mute could not be enabled when multiple calls were ringing and OneX Communicator was used to answer one of the calls.	131800	
The calling-party name was missing after a transfer recall operation when the client room feature was enabled and the value of the **Display Client Redirection** field was set to y.	131814	
The calling party number reported by the voice mail adjunct for a message record operation was incorrect when the call involved ISDN channel negotiation.	131831	
Occasionally, Communication Manager reset.	131838	
The crisis alert feature required all users to respond even when the **Every User Responds** field was set to no on the **system-parameters crisis-alert** screen.	131855	
Occasionally, Communication Manager reset while processing SIP calls.	131858	
A customer could not remove a skill using *3820# where *38 is the FAC and 20 is the skill because the # was incorrectly removed by the digit processing.	131862	
OneX Client Enablement Services could not be used with Communication Manager when it was routed via a Session Manager Release 6.3 or later.	131879	
Occasionally, there was no talk path on SIP calls using SRTP.	131880	
On Communication Manager system, there was no talkpath on incoming H.323 trunk calls when the signaling group of the trunk did not have the value of the **Direct IP connections** field set to y.	131881	
Occasionally, when an agent call with a bridged line appearance was dropped, Communication Manager reset due to an internal software trap.	131883	
A Radvision XT 5000 endpoint was used to make a call to a LifeSize 1020 endpoint. The XT 5000 endpoint then conferenced in a LifeSize 1030 endpoint, a Flare endpoint and a H.323 OneX Communicator endpoint. After some time, the H.323 OneX Communicator endpoint dropped from the conference.	131885	
An Avaya one-X® Communicator H.323 endpoint was used to make a call to a Radvision XT 5000 SIP endpoint. The XT 4200 SIP endpoint then called a XT 5000 SIP endpoint and a three-party conference call was created. After some time, the OneX Communicator H.323 endpoint got disconnected.	131886	
When an H.248 Media Gateway registered with a server after a link bounce that lasted longer than the link loss delay timer (LLDT), ISDN PRI calls were dropped when there are several DS1 boards in the media gateway.	131893	
There was no talkpath on a secure call made from Communication Manager Release 5.2.1 Communication Manager Release 6.2 and later.	131915	
An H.323 OneX Communicator endpoint was used to make a video call to AAC. However, there was no video on the call after it was answered.	131919	
An H.323 telecommuter was setup with a permanent service link over a SIP trunk. One call was made to an H.323 endpoint and was disconnected. The SIP service link responded with 408/481 to the session refresh REINV/UPDATE sent by Communication Manager. After this, no new calls could be made to the H.323 telecommuter for a period of two hours.	131926	
When a call made to a SIP station that had EC500 enabled got covered to SIP-integrated Voice Mail, the caller heard a generic greeting.	131967	
In a configuration where SIP messages associated with a call that was tandemed from a Communication Manager system to another over non-OPTIM SIP trunks, the system logged many UPDATE failures and reset when the display name for either call party contained quotes.	131973, 131988.	

Problem	Keywords	Workaround
ASAI transfers and conferences could not be performed from non-SIP stations that had EC500 or any other OPTIM feature enabled.	131989	
Occasionally, the hunt group administration audit caused the log files to get filled up very quickly.	131990	
The password information for scheduled backups was not migrated when the system was migrated to Virtual Environment.	132008	
When a domain-controlled SIP endpoint went off-hook, then on hook, there was no ASAI call initiated event. If the user dials digits and proceeds, the ASAI call initiated event was sent.	132030	
Ringback was not heard for calls made from a SIP or an IP endpoint to another IP endpoint that had EC500 enabled over a SIP trunk.	132032	
When the EC500 feature was disabled, a call placed from the cellular endpoint of a dual mode device did not drop when the SIP client resident of the same device merged into the call.	132041	
Remote mute (SA9120) did not work when an endpoint had a bridged call appearance in the in-use state.	132044	
An endpoint displayed the active call icon in the case of in-use bridged call state.	132053	
OneX Communicator in shared control that had the bridged call appearance of the calling party was unable to answer the call using the call appearance on OneX Communicator.	132066	
A call was dropped after 2 to 3 minutes when a page call was active via analog bridge appearance.	132080	
When a principle endpoint was active on a call and the bridged call appearance attempted to originate another call, they were bridged on to the call of the principle endpoint.	132163	
When SIP Direct Media was enabled, emergency call failed when the call was routed through the ISDN PRI trunk.	132191	

Problems fixed in Communication Manager 6.3.4.0

Table 12: Fixes delivered to Communication Manager 6.3.4.0

Problem	Keywords	Workaround
AST SIP endpoints monitored by the Client Enablement Services server did not show any indication for incoming calls when they were set to ring silently on the Avaya One X Mobile client.	103257	
When a skill is added or removed when an agent is on a call, an update was immediately sent to Communication Manager. This caused the reporting to ignore the call.	123033	
Occasionally, on a call with exclusion active, the call would drop when another extension attempted to bridge on.	130823	
When a call was made to a busy station on I55 from Communication Manager, the busy tone could not be heard and the calling party was dropped from the call.	131251	
Occasionally, Communication Manager dropped a Dual Mode SIP IOS client registered in the Multiple Device Access (MDA) mode from a call.	131404	
After delivering a call to a VDN after 250 active calls on the first two trunk groups of the route pattern subsequent attempts beyond the first two trunk groups failed.	131545	
Single Step Conference calls dropped when a listen-only party, such as a recorder, left the conference.	131579	
The user had to enter a digit to join the conference when AAC was used to make a call to a SIP phone that had Auto-answer enabled.	131655	
A call that covered to SIP Modular Messaging did not contain the calling-party name if the call was made over an ISDN trunk to a virtual extension on Communication Manager.	131736	
When the main server was in the split-registration mode and the survivable core server was not connected to the main server, the registered media gateways and the IP phones could not return to the main server on time.	131747	
On Communication Manager, any feature that sends multiple limited-duration tones, such as zip tone, then confirmation tone, to multiple stations that used resources on H.248 media gateways failed.	131778	
On Communication Manager with the multi-national feature enabled, IP endpoints (H.323 stations/trunks, SIP stations/trunks) may not hear the proper tones for their location. It is also possible that these endpoints may not be able to allocate TDM VoIP resources, causing loss of talk-path or call failures.	131808	
An incorrect display was observed for incoming R2MFC trunk calls that were transferred to another IP station.	131825	
On Communication Manager that had the multi-national feature enabled, IP endpoints such as H.323 stations, H.323 trunks, SIP stations, SIP trunks did not hear the proper tones for their location. It is also possible that these endpoints were unable to allocate TDM VoIP resources, causing loss of talk-path and call failures.	131845	
When telecommuter calls were active and the port network went through a cold reset, the media resources in the port network were still shown as being used. This caused exhaustion of media resources when there were high number of telecommuter calls.	131863	
When a SIP CC agent went off-hook in the Available state, CMS, IQ, and	131868	

Problem	Keywords	Workaround
BCMS continued to display the Available state for the agent.		
OneX Agent failed to enter timed ACW following the drag-and-drop transfer of an ACD call to a station call.	131891	
The calling party information displayed on the ACR using the Conf-Dsp button was incorrect after the call transferred from the IVR over a QSIG trunk.	131894	
An inter-tenant call made to an attendant using the attendant vectoring that is placed on hold did not alert after the expiry of the 'Time reminder on hold' timer that is configured on the console-parameters screen.	131895	
In a configuration with multiple H.248 media gateways spread across multiple IP network regions, the measurement reports for media-gateway DSP resource usage were inaccurate.	131897	
On Communication Manager, with the multi-national and multiple-locations features enabled, SIP endpoints did not hear the correct tones for their location.	131898	
Occasionally, a disabled speakerphone was inadvertently enabled after the phone performed a "reset values".	131908	
When in the survivable core server mode, calls made over an H.323 trunk between Communication Manager and a CISCO server failed.	131910	
Occasionally, Communication Manager reset after modifying the route pattern screen.	131914	
The telephone event in an incoming SIP INVITE message to Communication Manager did not tandem when the preceding SDP attribute in the same message had an unknown codec. This may result in functionality such as click-to-dial not working.	131921	
On Communication Manager, SIP endpoints lost talkpath after going through a vector with a collect digits step while listening to an announcement. This happened when Prefer use of G.711 by IP endpoints was enabled on the change system-parameters ip-options screen.	131925	
On a SIP-to-SIP call, when Direct Media was off on a signaling group, the call tried to shuffle to Direct IP. When an endpoint tried to perform a Single Step Conference or bridged on to the call, Communication Manager tried to bring the call on TDM and no talkpath was observed.	131929	
The Partition Routing Table screen did not handle PGN (Partition Group Number) values greater than 999. The data was incorrect after the screen was resubmitted.	131934	
Occasionally, the system reset when a glare condition occurred on SIP trunks.	131937	
Station A was used to make a call to Voice Portal. Voice Portal answers the call and transfers it to a DCP extension, Station 2. Station 2 had SAC enabled, and the call covered to another DCP endpoint, Station 3. When Station 3 was ringing, Station 2 deactivated the SAC. The call was not answered at Station 3 and the call covered to Station 2. When the call was answered at Station 2, there was no talkpath.	131942	
In an outgoing MLPP trunk call, the CDR report displayed an incorrect dialed number.	131945	
Starting a call type UDP entry on the Dial Plan Analysis table screen with an asterisk (*) or a pound sign (#) did not route calls correctly.	131957	

Problem	Keywords	Workaround
On Communication Manager, H.323 clear channel data calls failed to work properly with newer H.248 media gateway firmware loads that are RFC4040 compliant.	131986	
The RHNPA table screen did not accept a value greater than 999 in the Pattern Choices field. The system displayed the following error message after the screen was submitted: Error encountered, can't complete request; check errors before retrying	131998	
The Multi Device Access (MDA) bridge-on feature was not supported for devices across SBC.	132000	
The **`display capacity`** command now shows the correct capacity as follows: • Group Members Per System: 0 1000 1000 • CMS Measured ACD Members: 0 1000 1000	132007	
A segmentation fault due to a memory leak was observed on Communication Manager when an INVITE without mandatory headers and parameters was received.	132012	
IP phones could not be registered after a WAN outage.	132013	With duplicated servers, a server interchange will resolve the problem. With a simplex server, a system restart will resolve the problem.
When an incoming PRI call did not have the calling party information and was routed to Voice Portal followed by a transfer over a SIP trunk to an agent on another Communication Manager, the display on the agent was updated incorrectly when the agent answered the call.	132014	
The system did not display any output when the list registered-ip-stations command was run with the release option.	132027	
hen an incoming R2MFC trunk call made to an H.323 station was transferred to a SIP station, the bridged call appearance of that SIP station was not updated with the incoming ANI.	132035	
An incoming SIP trunk call that is transferred using a Refer message from a voice portal was not dropped until 30 seconds after it was disconnected.	132045	
Certain Single Step Conference features did not function properly when Communication Manager failed to update the call appearance button after overlap dialing was used on an ISDN trunk.	132055	
The small and medium survivable servers backing up a bigger configuration are now changed to support the matching survivable servers memory size. Using **`display capacity`**: • Group Members Per System: 0 1000 1000 • CMS Measured ACD Members: 0 1000 1000 • Medium survivable backing up a large main. • Group Members Per System: 0 60000 60000 • CMS Measured ACD Members: 0 60000 60000	132063	
Occasionally, the **Prepend '+' to Calling/Alerting/ Diverting/Connected Number? y** field in the Trunk Group screen of the SIP Trunk stopped working.	132074	
Communication Manager reset when the far-end responded with fewer m=	132079	

Problem	Keywords	Workaround
lines in SDP in answer to the shuffle invite.		
Calls made to an invalid number that were directed to an attendant vector that routed ARS failed to select the second route pattern preference trunk group if the first preference trunk group was busy.	132093	
Occasionally, when a trunk call was made to a SIP station with the Secure Only SRTP mode, hold/unhold would not work.	132098	
The display on an IP telephone was in the wrong language when the Communication Manager setting for the station was set to unicode and the actual phone did not support Unicode.	132099	
ISDN-PRI trunk calls made to a busy X-ported station dropped instead of sending a busy tone to the calling party.	132103	
When an auto-answer agent received a call to a non-VOA VDN after a call to a VOA VDN that pointed to the same vector and the caller dropped while the VOA was playing, the agent could not hear zip tone when the call was cut through. This happened when the Hear Zip Tone Following VOA? field was set to n in the system-parameters features screen.	132110	
When an existing location parameter was changed in the change locations screen, the audio level updates were not sent to the associated media gateway VoIP media. The audio levels that have to be sent are administered on the change terminal-parameters screen.	132117	
A SAC enabled DCP endpoint did not clear the display on a bridge call appearance when the far-end dropped the call without the call being answered.	132126	
A call made from a OneX Communicator terminal in the Telecommuter mode caused Communication Manager to restart.	132129	
Users were unable to log into a OneX attendant after being placed in the night mode.	132134	
Occasionally, IP Bandwidth audits produced false error indications that showed up in the system error logs and in the status audits command.	132138	
When an EC500-mapped cellular phone was used to call a VDN over an R2MFC trunk in a transfer operation, the display on the station was incorrect.	132155	
A SIP trunk call made to a DCP endpoint on a different port network than the SIP trunk resulted in no ringback on the SIP trunk.	132156	
Mute could not be activated on the desk phone when a second call was made from OneX Communicator and the first call was answered on the EC500 endpoint.	132162	
If a principle station was active on a call and a bridged station attempted to originate a call they were bridged to the principle station’s call.	132165	
When a call encountered a vector collect step and a digit was dialed after the announcement was played the call was routed after fixed interval of 10 seconds instead of the administered value of Prompting Timeout field on system-parameters screen.	132167	
Occasionally, poor voice quality was observed.	132176	
While using a vustats button, the tenant check used the station information instead of the agent information.	132189	
Occasionally, with a large number of BRI trunk groups, the system would reset.	132221	

Problem	Keywords	Workaround
Calls are getting queued after hours when those calls were supposed to get a "closed" message based on the "Service Hour Table" treatment.	132222	
An H.323 video-enabled Avaya one-X® Communicator endpoint (Station 1) on Communication Manager (CM 1) was used to make a call to an H.323 audio endpoint (Station 2) on CM 1. The IP codec-set had wideband codecs administered and Station 2 was also wideband capable. Station 2 transferred the call to an H.323 video endpoint (Station 3) on another Communication Manager (CM 2). Both Communication Manager systems were connected via an H.323 trunk. After the transfer was complete and the call was answered, there was no video and the call was connected as audio-only.	132229	
Occasionally, a memory leak was observed when some SIP calls were cleared before being answered.	132240	
When SIP Direct Media was enabled, emergency calls failed when routed through the ISDN PRI trunk.	132241	
An incorrectly formed SIP INVITE message did not have the IP address of the media resource used by the SIP trunk in case of TDM trunk (MFC, TONE, ISDN-PRI) to SIP trunk call. This caused incorrect bandwidth calculations.	132263	
On Communication Manager, calls involving SIP trunks and SIP stations dropped when the port-network VoIP board or H.248 media gateway stopped functioning.	132281	
A SIP signaling link to Session Manager could not be used for ASAI if it was TCP.	132290	
On a SIP endpoint, when a principle user joined a call that was put on hold by a bridged user, the principle user could not drop the call after going on-hook.	132312	
Communication Manager underwent a software reset during simultaneous log-in and log-off attempts by users using the Personal Station Access (PSA) associate and dissociate code respectively.	132358	
Emergency calls made from a SIP station dropped after 3 minutes.	132370	
Occasionally, there was a segmentation fault on Communication Manager when SIP Direct Media was enabled.	132395	
A call was made from an MDA device (MDA 1) to a SIP or an H.323 extension. Before the call was answered, another MDA device (MDA 2) was used to bridge on to the call. Communication Manager allowed the bridge-on operation. Communication Manager should allow bridge-on only after the call is answered. Occasionally, when the bridge-on operation happened before 180 ringing, call dropped.	140000	
Occasionally, announcement playback failed when there were multiple boards in an announcement audio group.	140005	
When the principal station makes a call and the far-end answers it, the SIP phones with a bridged call appearance of the principle station displayed the trunk name instead of the dialed number.	140031, 140082	
If a SIP station was used to make an outgoing R2MFC trunk call and was attendant-transferred to a local H.323 station then the station to which the call was transferred did not display the digits dialed by the originating SIP station.	140049	
When the station set type was changed to 9608, 9611, 9621 or 9641, the OPS application type was automatically administered on the Off-PBX-	140063	

Problem	Keywords	Workaround
Telephone Station-Mapping screen. The OPS application type could not be removed through administration.		
The PROC error 7171 20592 was logged in after every H.323 phone registration.	140071	
On Communication Manager with H.248 media gateways, the system did not use the media gateway VoIP to its full capacity.	140128	

Problems fixed in Communication Manager 6.3.4.1

Table 13: Fixes delivered to Communication Manager 6.3.4.1

Problem	Keywords	Workaround
Occasionally, CMS and IQ reports for legitimate completed calls were incorrectly reported as abandoned.	131499, 131892.	
Occasionally, Communication Manager reset when an endpoint registered to a network region greater than 250 through the ip-network-map screen.	131875	
Communication Manager did not play the busy tone after receiving the SIP 486 response with the Retry-After header to the initial INVITE message.	132020	
Communication Manager received a translation corruption message when a SIP set type that had an OPS and EC500 entry in the off-pbx-telephone station-mapping screen was changed to H.323.	132297	
Communication Manager did not register the 1692-type phones when the endpoint assigned to a network region was greater than 250 and the processor ethernet interface where the phone registers to was in a network region less than 250.	132371	
A SIP trunk did not drop when the Network Call Redirection feature was enabled and the incoming SIP trunk call landed on a vector with a reroute step.	132479	
Occasionally, while processing SIP calls, Communication Manager encountered an internal error that incorrectly managed the system memory associated with the call causing a system restart.	140050	
Using the change locations screen could sometimes result in the users hearing wrong dial tone, not being able to register phones or, experiencing difficulty in making or receiving calls using the media gateways in specific locations.	140161	
Communication Manager reset when an incoming SIP message contained a non-numeric value in the time field of the session description body.	140203	
Occasionally, Communication Manager reset when a call involved ISDN or H.323 trunk calls and H.323 or SIP stations.	140239	
When Communication Manager used the ASAI link version 5 or above and the system had undergone a level 2 reset since the last reboot, then the next ASAI station status query caused a system reset.	140241	
When the failover group domain table on Session Manager was configured but the failover-grp-domain-map screen was left un-administered, then, under heavy SIP traffic, Communication Manager reset.	140279	
Occasionally during heavy SIP traffic, the system reset.	140289	
Occasionally, Communication Manager reset in call scenarios that involved SIP.	140462	
Occasionally, Communication Manager reset when an un-named H.323 station registered to it.	140485	
Occasionally, the system displayed the `Entry is bad` error message while submitting a screen.	140493	
Occasionally, SIP calls transferred by Modular Messaging resulted in a software reset.	140508	
On Communication Manager, SIP trunks in network regions without VoIP resources were unable to listen to music-on-hold.	140516	
Occasionally, Communication Manager reset when the source-based routing feature was used and a call originated via a TDM trunk.	140525	
When a full core file was being gathered using the corevector command by Avaya services on duplex systems, the resulting interchange caused the new active	140527	

Problem	Keywords	Workaround
server to incorrectly undergo a full system reload instead of the level one restart.		
Due to an internal resource constraint that began with external network problems, Communication Manager stopped processing SIP messages.	140591	
Occasionally, the additional level of SIP debug messages enabled by Avaya services resulted in a system restart.	140767	
Occasionally, SIP messages were not sent to the network.	140768	
Occasionally, Communication Manager reset when a non-ACD call that was put on hold at an ACD agent station dropped.	140807	
Under heavy traffic conditions, incorrectly managing internal resources resulted in Communication Manager undergoing a software reset to recover resources and services.	140819	
Intermittently, calls dropped after 32 seconds when SIP transactions involved provisional acknowledgements.	141045	
Improper handling of internal resources related to media sometimes caused Communication Manager to reset when processing SIP calls.	141078	
Communication Manager could undergo a level one reset when an ACD call dropped from a manual-in ACD agent's station while the agent had at least one additional call on hold and was not active on a call, such as in call transfers.	141119	
When network conditions caused active SIP calls to be considered in the connection-preservation mode, incorrect handling of internal resources caused memory exhaustion. This lead to a system reset.	141173	

Problems fixed in Communication Manager 6.3.5.0

Table 14: Fixes delivered to Communication Manager 6.3.5.0

Problem	Keywords	Workaround
A whisper page could not be initiated from a SIP endpoint that had an active or a held call.	113273	
The IP address information on Communication Manager could not be changed from System Platform when an alias address was configured and the new IP address information was on a different subnet than the alias.	121765	
In an environment with multiple Communication Manager systems, when a 96x1 H.323 endpoint transferred an incoming call from an H.323 Avaya one-X® Communicator endpoint to an Avaya iPad or a Windows Flare device, there was no video on the call and the call dropped after 32 seconds.	123009	
When a call was made from a device by using the Multiple Device Access feature, another device could be used to incorrectly bridge onto the call, thus causing the call to fail.	130072	
ASAI applications received agent state and login and logout notifications when skills were added, changed, or removed by using the change agent xxxxx auto command even when the CTI link was set to not send CMS Move Agent events.	130152	
The Call forward feature did not work for a SIP endpoint that was configured on the One-X Client Enablement Services server.	131052	
The Communication Manager license did not expire on the system even after it had expired in WebLM.	131360	
Intermittently, video calls made from an H.323 Avaya one-X® Communicator endpoint on one Communication Manager system to an H.323 Polycom HDX endpoint on another Communication Manager system that had encryption enabled over a SIP trunk using TCP dropped as soon as they were answered.	131622	
The MCH SIP Agent calls were reported as Idle instead of Active on the Agent Status screen of IQ when an ACD call that was on hold dropped even when the agent was on an active call.	131686	
Communication Manager reset under some conditions when debug prints were enabled on the system and the network connection to the processor Ethernet interface was removed during calls.	131731	
Occasionally, there was no talk path on a forwarded call that covered to the voicemail server.	131772	
A callback call from the One-X Client Enablement Services server changed the location of the corresponding SIP station on Communication Manager causing the location related features to function incorrectly.	131782	
Occasionally, Communication Manager reset when an endpoint was registered to a network region greater than 250 through the ip-network-map screen.	131875	
Transfer of call across multiple Communication Manager systems failed when Direct Media was enabled, the Initial INVITE with SDP for secure calls field was not set, and the ip-codec-set was set as Capability negotiation capable on Communication Manager.	131916	
A call transfer over a SIP trunk failed when the network region of the party completing the transfer failed the network region connectivity test.	131971	
After resuming a held call between two video-enabled endpoints while Music on hold and Direct media were turned ON, there was no audio and video.	131995	
When a user in a call pickup group called another member in the same pickup group, the user could see the Call pickup button flash on the endpoint, but could	132010	

Problem	Keywords	Workaround
not press the button to answer the call.		
Communication Manager did not play the Busy tone when it received the SIP 486 response with the Retry-After header to the initial INVITE message.	132020	
When a restricted call was made to a SIP station over a PRI trunk, the caller identity was incorrectly disclosed.	132042	
When all of the following conditions were met, calls did not route properly: • The call was forwarded. • The calling and forwarding endpoints were in different locations. • Digit conversion was involved in the routing of the call. • The digit conversion rules were different in both the locations. • LAR was triggered.	132109	
Loading more than eight trusted certificates caused none of the certificates to be loaded onto Communication Manager.	132114	
The display on a SIP endpoint for a call made to a team was incorrect when the calling name was longer than 15 characters and contained extended Latin characters.	132140	
The Call pickup button of an endpoint in a pickup group flashed incorrectly when the endpoint was used to make a call to the pickup group.	132161	
A direct agent call made from another agent caused the number to be truncated on the display screen of the endpoint of the called agent.	132179	
When an Avaya one-X® Communicator endpoint operated in the shared control mode for a 96xx station, `A=` appeared instead of `3=` when the Enhanced call forward feature button was activated and the display language was anything other than English or Unicode.	132185	
Occasionally, QSIG Path Replacements failed.	132202	
When an incorrect extension was typed on the Login screen, the endpoint remained in the Discovering mode.	132231	
Under certain internal conditions, a Radvision XT SIP endpoint was unable to start a slide presentation.	132233	
An H.323 endpoint did not fall back from the ESS server to the main server.	132256	
A call made from a non-Avaya SIP phone dropped.	132266	
The agent endpoint displayed the trunk group name instead of the calling party number when an incoming ISDN trunk call tandemed over a SIP trunk and the far end sent an UPDATE or a ReInvite without the number in the contact or PAI header.	132267	
Under some conditions, a SIP NOTIFY message sent the wrong call state in response to the SUBSCRIBE dialog when network connectivity was lost and restored between the two SIP endpoints.	132268	
When the location parameter value was changed on the Locations screen, none of the correct H.248 media gateways and port networks received the Location parameter update. Instead, all other translated media gateways and port networks in different locations were updated.	132270	
The endpoints in a pickup group were constantly ringing when multiple calls were made to the pickup group.	132284	
There was translation corruption when a SIP set type was changed to H.323. The SIP set had OPS and EC500 entries on the off-pbx-telephone station-mapping screen.	132297	

Problem	Keywords	Workaround
Outgoing trunk calls using LAI failed when a Progress message was received with cause value 31 and the call interworked at the far end.	132303	
When Communication Manager was used in the Feature server mode, the CPU usage increased due to shuffle reINVITE glare.	132311	
Calls made to a SIP station with SAC enabled covered to Modular Messaging but also continued to follow the second cover point.	132315	
When attendant vectoring was used to generate a VIP wakeup call, the station receiving the reminder to make the VIP wakeup call did not have the information about the party that needed the wakeup call.	132316	
Occasionally, a newly active server in a duplicated pair reset after a server interchange.	132317	
When all extension blocks were marked as remote (AAR) and the **`add station next`** command was run, the system displayed the `No available extensions in the system` error message.	132322	
Communication Manager did not send the names of Vectors, VDNs, trunks, agents, and hunt groups to IQ when there were no externally measured trunks or no externally measured VDNs, or no externally measured hunt groups.	132331	
The **Total Persistent Variables in Use** value was incorrect in the list measurement summary report.	132339	
The CDR record was missing when an agent transferred an incoming trunk call back to an IVR.	132340	
When a video-enabled SIP endpoint was used to call another such endpoint over a Direct media-enabled SIP signaling group, there was one-way video if the destination SIP station had EC500 configured over a trunk that had video and Direct media disabled, and the SIP 180 message from the EC500 leg was received after the call was answered at the destination.	132344	
Occasionally, calls made to the attendant that were routed to a VDN with attendant vectoring were connected to the wrong music source when the call was answered and then put on hold.	132350	
Occasionally, announcement playback failed when there were multiple boards in an announcement audio group.	132352	
Communication Manager did not route calls to the secondary Session Manager when the SIP 302 `Moved Temporarily` message was received by Communication Manager because the trunk to the primary Session Manager was down.	132363	
Communication Manager failed to register 1692 type phones when the endpoint assigned to a network region was greater than 250 and the Processor Ethernet interface where the phone was registered to was in a network region less than 250.	132371	
When Network Call Redirection was enabled and an agent tried to transfer the call, Communication Manager received INVITE with replaces followed by a REFER with replaces, and the transfer failed.	132373	
On Communication Manager, calls made using the Dial Plan Transparency feature failed when H.323 and SIP IP trunks were used, Call recording was active, and the H.248 media gateways were used for media resources.	132379	
Look Ahead Routing did not work when the CPN block or unblock Feature Access Code was used.	132382	
With the Conference display button, the calling number details of the original calling party were displayed when it should have been restricted.	132391	

Problem	Keywords	Workaround
If a whitespace was entered as part of a username on some of the System Management Interface web pages, the tasks being performed by the web pages did not complete successfully.	132394	
The Release Link Trunk (RLT) feature failed to notify the PSTN when two trunk calls were transferred together. This caused the trunks to remain active when they should have been dropped after the transfer was completed.	132403	
The voice mail greeting was incorrect after the **last-fwd** option on the **Coverage Path for Incoming Diverted QSIG/SIP Calls** screen was selected.	132413	
When all of the following conditions were met, a call made between two Communication Manager servers over an H.323 trunk group disconnected without any feedback to the calling party: • The incoming H.323 trunk group was configured for overlap receiving. • The incoming H.323 trunk group inserted the Automatic Route Selection or Automatic Alternate Routing access code. • The calling party sent a complete number. • The incoming call obtained its VoIP resources from a Media Processor in a G650 Port Network.	132423	
When an agent migrated from the main server to a survivable server, the auto-in button continued to flash.	132425	
When an endpoint was used to make an R2MFC trunk call and the call was transferred to another local station by the originator, then the display of the transferred-to endpoint was not updated with the digits dialed by the originating station.	132429	
An unattended transfer from a Cisco SIP endpoint resulted in call drop.	132442	
Communication Manager outpulsed the last digit twice when a call was routed using LAR.	132444	
Service observing using a feature access code was denied from an Avaya one-X® Communicator endpoint that was logged in the telecommuter mode.	132456	
The AAR/ARS Patterns information on the display capacity screen was not updated correctly when route patterns were cleared out.	132469	
A SIP trunk call did not drop when Network Call Redirection was enabled and the call landed on a vector with reroute step.	132479	
Occasionally, the `list station` command could not be run and the server CPU occupancy would become extremely high.	132482	
When the Dial plan transparency feature was used, Call recording using the ASAI-based multiple endpoint registrations failed.	132488	
An ISDN-SGRP alarm was left up for secondary D-channel after it was removed from administration. The alarm could not be cleared without a system reset.	132497	
When the Dial plan transparency feature was used, Call recording using Service Observing failed.	132501	
When a user was attempting to setup a conference call but received and answered another call before they were done the conference operation was not aborted even though the **Abort Conference Upon Hang-Up** field was set to yes. When the user attempted to transfer this new call, the old call was transferred to the new call by mistake. **Note:** The name of the **Abort Conference Upon Hang-up** field is now changed to Abort Conference.	132504	
A call was stuck in the vector-collect digits step when **DTMF over IP** on the	132513	

Problem	Keywords	Workaround
signaling-group screen was set to out-of-band.		
When an unnamed H323 endpoint made a call over a SIP trunk, Communication Manager would not send the via header in the outgoing SIP INVITE.	132514	
The calling station hears silence after dialing a conference bridge when the vector had ~p in the route to step.	140001	
Emergency calls made from a SIP endpoint dropped after 3 minutes.	140003	
The codec preferences on SoftFlare and on Communication Manager were different. SoftFlare was used to make a video call to an H.323 Avaya one-X® Communicator endpoint. Two-way audio and video was observed on the call. When the H.323 Avaya one-X® Communicator endpoint conferenced in a SIP 96x1 endpoint, there was no audio on SoftFlare after the SIP 96x1 endpoint answered the call.	140009	
Occasionally, Communication Manager restarted when a 200 OK message had to be re-transmitted while processing SIP calls.	140021	
A server interchange caused corruption of service and feature fields on the route-pattern screens.	140026	
Occasionally, while processing SIP calls, Communication Manager encountered an internal error that incorrectly managed the system memory associated with the call. This caused a restart.	140050	
In a contact center, a call was placed in queue by ICR. After periodic intervals, Communication Manager updated ICR with the call-related information. Due to some internal error, Communication Manager failed to send this information and reset.	140053	
When an H.248 media gateway supplies media resources for a network region, the second call from one H.323 station to another in the same network region had no talk path when ephemeral caching was turned off.	140068	
On Communication Manager, SIP station calls to H.323 station calls did not have two-way talk path when the following administration was enabled: • Initial IP-IP Direct Media is set to y on the SIP signaling-group screen used by the SIP station • G.726A-32 is the first or only codec selection in the codec-set used between the SIP signaling-group region and the H.323 endpoint region • The SIP endpoint is capable of doing G.726 and is so enabled in its settings file (if applicable)	140087	
There was no talkpath when a call that was unattended transferred over a SIP trunk was answered.	140126	
R2MFC trunk calls made to an IP endpoint that were blind or supervised transferred to another IP endpoint displayed the trunk name instead of the DNIS.	140138	
Communication Manager could undergo a system reset in either of the below situations: • A video-enabled DCP endpoint was used to log in from a DCP physical endpoint and also from a video-enabled Avaya one-X® Communicator endpoint in the shared control mode. The incoming video SIP trunk call that was made from a SIP endpoint underwent a Hold and Unhold operation after the call was answered. • When the soft client was video-enabled, an audio call that the telecommuting user placed resulted in a system reset if the call was held and unheld.	140148	
Communication Manager did not properly exercise the full media-processing	140149	

Problem	Keywords	Workaround
capacity of H.248 media gateways.		
SIP phones with a bridged call appearance displayed the trunk name instead of the originally dialed number when the principal station made an outbound call and the far end answered the call.	140150	
If a SIP endpoint was used to make an R2MFC trunk call and then performed a supervised transfer to an H.323 endpoint, the display showed the trunk name instead of the dialed digits.	140151	
Under certain SIP call scenarios, Communication Manager did not properly release all system memory consumed by the call. After many occurrences of this scenario, over time, the system reset.	140182	
A port board that had translations associated with it was removed from the port-location screen. When there were no other associated translations, the board was removed from the circuit-packs screen. This resulted in a corruption when a different kind of board was plugged in and translated.	140212	
When the **`enable mg-return all`** command was run, Communication Manager restarted.	140213	
There was no talk path on an inter-network region call made from a SIP endpoint to a DCP endpoint when the **Direct IP-IP Audio Connections** field on the **SIP signaling group** screen was set to no.	140228	
Occasionally, Communication Manager reset while trying to determine the internal location details of IP endpoints.	140238	
Occasionally, Communication Manager reset during calls involving ISDN and H.323 trunks and IP stations.	140239	
Call Center agent reports had wrong cause value for a call on the third line when the second line was active due to consult action.	140244	
The endpoints did not display Mute for the Conference display feature when the far-end Mute button was activated.	140260	
When a SIP endpoint was used to make a call, it sometimes received the 403 No More Call Appearance response from Communication Manager.	140262	
When multiple calls were made to a call pickup group or when a call pickup group call covered to the coverage answer group, there was continuous ringing on one of the endpoints.	140310	
Call pick up alert did not work for SIP pickup group members.	140319	
Occasionally, Communication Manager log files were filled with error messages that were generated when an endpoint was assigned to a network region greater than 250 through the ip-network-map screen.	140340	
When the Locations screen was edited using Avaya Integrated Management products, incorrect dial tones were observed and phones could not be registered in the given location of a media gateway.	140367	
Communication Manager restarted when SIP video calls were made between Radvision endpoints and the calls employed multiple applications such as BFCP (Binary Floor Control Protocol), FECC (Far End Camera Control), and FEC (Forward Error Correction).	140372	
When R2MFC trunk calls made to a station were supervised transferred to another station, the trunk group name was displayed instead of the calling party number.	140373	
A monitoring station for the SIP team button feature continued to ring even when the call was answered by another monitoring station.	140377	

Problem	Keywords	Workaround
The system displayed the `Error encountered, can't complete request; check errors before retrying` error message when a SIP station that had an EC500 entry administered in the **off-pbx-telephone station-mapping** screen was converted to H.323.	140463	
Under certain internal conditions, SIP calls that were transferred by Modular Messaging caused Communication Manager to reset.	140517	
When the Source based routing feature was used, Communication Manager sometimes underwent a software reset when the call originated via a TDM trunk.	140525	
Under some conditions, the Communication Manager server incorrectly entered the license error mode.	140557	

Problems fixed in Communication Manager 6.3.6.0 (FP 4)

Table 15: Fixes delivered to Communication Manager 6.3.6.0 (FP 4)

Problem	Keywords	Workaround
A principal station is used to bridge onto a held call between another station and the EC500 endpoint of the principal station. When the principal station was dropped from the call, Music on Hold was not heard at the principal.	120033	
Communication Manager trunk capacity could be exhausted when: • Several QSIG-capable PBX servers were connected using QSIG trunks in a star formation, with Communication Manager in the center, and • A call traveled into and out of Communication Manager several times due to redirection and call transfer, and • One of the QSIG-capable PBXs signaled a QSIG Path Replacement Retain operation. In such a case, the Communication Manager QSIG path-replacement logic failed to eliminate the unnecessary trunks.	120918	
When a VDN is called over a SIP trunk the hunt group number is displayed instead of the VDN number.	121012	
CMS and IQ reports incorrectly showed an abandoned call when an agent on a conference call with the customer and another agent, dropped the agent that had placed the call on HOLD.	121623	
There was no video on audio calls made between a SoftFlare endpoint to a Radvision H.323 endpoint when the SoftFlare endpoint upgraded the call to video (non-wideband audio).	130320	
When an SRTP H.323 endpoint on Communication Manager called another SRTP H.323 endpoint and the call covered to and was answered by a third H.323 endpoint, the principal station heard noise when it bridged on if the MLPP feature was enabled.	130390	
The direct media call across two Communication Manager servers dropped when the call was placed on hold and SRTP and Network Call Redirection features were enabled.	130397	
When a SIP bridge appearance went off-hook the SIP PUBLISH message was sent with a wrong dialog state to the principal station.	130690	
When a HDX user on one Communication Manager called a Avaya one-X® Communicator user on another, the call established correctly. If the One-X user now transferred the call back to another HDX user on the originating Communication Manager then the call would be audio only and drop shortly after.	130942	
When an audio-only H.323 endpoint calls a Radvision video endpoint, and then transfers the call to the Radvision endpoint to another video-enabled endpoint, the resulting call between the Radvision endpoint and the video-enabled endpoint has audio, but no video.	131230	
In some circumstances a call can be stuck and cannot be ended at the calling party phone. This can happen when a call is made from a party using resources on one media gateway to another party on a different media gateway, and while the call is still ringing, the calling party's media gateway resets, and then the called party answers the incoming call.	131342	
Occasionally, FAX over SIP trunks failed.	131401	
When inter-region video calls were denied due to bandwidth limitations, there was no corresponding **exceeded bandwidth** peg on the Inter Network Region Bandwidth Status administration screen.	131466	
There was no talkpath between SIP stations after an unattended transfer if the SIP trunk had Network Call Redirection enabled and the SIP signaling group had **Initial**	131602	

Problem	Keywords	Workaround
IP-IP Direct Media set to y.		
SIP calls would complete even when the bandwidth limit had been reached.	131604	
The principal station could no longer bridge on to a call that was originated from the bridge appearance after it had undergone network recovery while the other two parties were on call.	131713	
An ASAI domain control for a SIP endpoint provided an extra endpoint registered or unregistered event when CM subscribes to the SIP REG event package for the SIP station.	131784	
When a Redcom endpoint performed a Hold or Release operation on a call, there was one way talkpath after call was resumed.	131947	
On Communication Manager, calls that involved SIP trunks and SIP stations dropped when the port-network VoIP board or the H.248 media gateway stopped functioning.	131958	
When using an ANI variable in a vector, if the call comes in from a SIP trunk with a plus sign (+) in the calling party number, the variable is not correctly processed.	131991	
An H.323 Avaya one-X® Communicator endpoint made a video call to an iPAD-Flare. After the call was answered, if the 1XC-H.323 stopped the video causing the iPAD to downgrade the video, then the call was dropped.	132029	
The Administrator Accounts SMI screen did not support special characters in the Password field.	132075	
The **SA9120-Turn On Mute for Remote Off-hook Attempt** field on the station screen did not work in the OSSI terminal when in interaction with the auto answer or the int-aut-in button.	132120	
When there are calls in queue to hear an announcement and at the same time the call record audit runs, the audit would throw several invalid software errors.	132249	
A video SRTP call transferred to a non SRTP endpoint dropped.	132269	
Occasionally, Communication Manager reset during an H.323 IP station registration and unregistration process.	132338	
Under some extreme circumstances, Communication Manager could exhaust internal message buffers that could lead to a system reset.	132345	
When a video enabled SIP station called a video enabled H.323 station on another Communication Manager registered as only audio capable then a hold-unhold operation by the H.323 station resulted in no talk path.	132357	
Under very rare circumstances, registration of a soft attendant console caused a Communication Manager software reset.	132359	
Hardware errors associated with VAL boards and announcements were logged.	132366	
When a telephone number cannot be routed by Communication Manager then the plus sign (+) in the SIP URI is changed to %2B.	132376	
A Video SoftFlare SIP phone made video call to H.323 Radvision endpoint. After the call was answered, if the SoftFlare client downgraded and then upgraded the call to video there was no talk path after the attempt to upgrade to video.	132377	
Communication Manager failed to originate a call from call log using auto call back when the calling number was stored with a plus sign (+).	132381	
Occasionally, escalating an audio call to video by a Flare endpoint caused another Flare endpoint in the conference to drop from the call.	132410	
An Avaya one-X® Communicator endpoint (SIP-A) called a 96X1 SIP endpoint that had **Ip Video** as y. After the call was answered, the 96X1 SIP endpoint performed	132419	

Problem	Keywords	Workaround
a blind transfer to another Avaya one-X® Communicator endpoint (SIP-B). SIP-B answered the call, there was no video and the call established was an audio-only call.		
When a video enabled flare SIP user called a 96x1 SIP station the call was audio only as expected. When the 96x1 performed a blind transfer to a video-capable Avaya one-X® Communicator H.323 user the call remained as audio. When the Flare user escalated the call to video the call incorrectly remained as audio only.	132424	
Avaya one-X® Communicator calls made across a SIP trunk with direct media and music on hold features resulted correctly in two-way video. If another Avaya one-X® Communicator endpoint called the originator resulting in the first call to be placed on hold followed by a transfer of the second caller to the first called party, there was no video on the call.	132439	
SIP Endpoint Managed Transfer failed when the transfer target had call forward enabled.	132460	
Communication Manager sometimes misinterpreted certain music on hold frequencies as FAX tones causing dead air when callers are placed off hold. This happened when T.38 was administered on the ip-code-set screen.	132462	
When a Windows Flare user on a Communication Manager system made a video call to a 96xx SIP user on another Communication Manager system and the 96xx SIP user then transferred the call to a Radvision XT endpoint, the resulting connection established as audio-only.	132486	
For a conference hosted by Radvision or Lifesize MCU registered to the Avaya Session Manager, SRTP enabled H.323 endpoints that joined the conference as the second party or later would be dropped.	132498	
Agents on Communication Manager that use H.248 media gateways for resources (either for VoIP or a physical port on the media-gateway) heard the incoming caller while the 'zip' tone was played to the agent.	132508	
An incoming call to an IP DECT station did not have a CPN prefix attached to the calling-party number.	140002	
Occasionally, an incoming SIP trunk call to Experience Portal resulted in no talk path.	140012	
A monitored station did not receive a CTI alerting event when it was busy on a call and had Call Forwarding Busy/DA enabled.	140014	
When two SIP calls were merged the resultant merged call may experience problems if the call underwent path replacement.	140019	
When the Direct media feature is enabled, ring back is played even when the CAC bandwidth limit was reached.	140034	
On Communication Manager, an H.323 or a SIP IP endpoint that belonged to an ip-network-region without VoIP resources was unable to connect to TDM services.TDM services are announcements, music-on-hold, listening to digits, talk-listen to other ports.	140044	
Lamp refresh update leaves resources allocated during the test on IP phones if the IP phone unregisters during the test. This causes several software error logs and it also leaves internal data elements unusable for a period of time.	140055	
When a SIP station makes an outgoing call, sometimes, it could incorrectly get the response that no more call appearances are available to make the call.	140058	
CDR was not generated for a conference call that was later transferred to another party and **SA8434 - Delay PSTN Connect on Agent Answer** was enabled.	140061	

Problem	Keywords	Workaround
IP telephones that have **Near End Establishes TCP Signaling Socket** is set to n did not recover cleanly after a duplicate processor ethernet server interchange.	140072	
An iPAD Flare endpoint on a Communication Manager system (CM A) was used to make a call to an H.323 96x1 endpoint on another Communication Manager system (CM B). A two-way audio path was established. Encryption was enabled on the ip-codec-set screen and both the endpoints supported encryption. The 96x1 endpoint was then used to make a blind transfer to an H.323 HDX endpoint on CM B. As expected after transfer, the established call was audio-only. The Flare endpoint then escalated to video. Audio became one-way and video did not start. The call dropped after 32 seconds.	140073	
An unattended call transfer to a SIP station would not cover to a remote coverage point when the call was not answered.	140075	
When an incoming R2MFC call was made to an agent after which the call was routed over SIP trunk using VDN return destination, the SIP INVITE message did not contain the calling party number received over the R2MFC trunk.	140094	
An external call to an H.323 based voice portal that is then transferred to a SIP station would not update the display until the call was answered.	140101	
A corrupted dialplan expansion entry caused the **`list extension-type`** command to skip administered stations.	140103	
After some stable transfers by a SIP-connected ICR, an incorrect messaging sequence between Communication Manager and the CMS caused IQ and CMS to report calls incorrectly as abandoned.	140125	
The call logs did not show the correct entry for calls redirected to DCP or H.323 stations after the off-hook alert time out.	140143	
When a 9608SIP, 9611SIP, 9621SIP, 9641SIP was changed to a non-SIP set type and there was a second entry on the off-pbx station-mapping screen, the OPS entry is not removed.	140147	
H.323 phone registrations that occurred while a call was ringing failed to properly update the ringer and the display of the newly registered phone.	140156	
Using the change locations screen could result in the users hearing wrong dialtone, not be able to register phones, or experience difficulty in making or receiving calls using the media gateways in specific locations.	140161	
Administration of an Automatic Message Waiting button to monitor the extension assigned in the **Extension to Receive Failed Wakeup LWC Messages** field of the system-parameters hospitality screen was blocked.	140171	
IP station users that called a busy station had the incorrect soft keys displayed.	140174	
Media Gateway recovery was delayed after a server interchange.	140192	
When a SIP CC station is used to make a call, the station receiving the call did not get a screen pop.	140198	
Under specific conditions, Communication Manager would not acknowledge the originator of a SIP call, thus resulting in the dropping of the call.	140199	
A SIP endpoint managed transfer failed when the **Special Dial Tone for Digital/IP Stations** field was not set to none on system-parameters features screen.	140202	
Communication Manager reset when an incoming SIP message contained a non numeric value for the time field in the session description body.	140203	
Intermittently, under some internal conditions, ASAI initiated SIP call transfers failed.	140205	

Problem	Keywords	Workaround
When the VDN administered in the **VDN extension used as Redirect on IP/OPTIM Failure to VDN** field on page 3 of the hunt group screen was removed from the system, the `Error encountered, can't complete request; check errors before retrying` message was displayed while removing stations, listing hunt-groups, or performing administration tasks on the hunt-group that was using the VDN extension.	140207	
When a native name was not configured but the language is set to Arabic, the principal SIP station displayed CONFERENCE in English when the call was answered at the bridge appearance and the principal station tried to bridge onto the call.	140209	
The **`list trace station/TAC`** command displayed the wrong calling name and number when SA9086 was enabled.	140211	
Under certain internal conditions, Communication Manager did not correctly release internal memory required for managing connections between media gateways resulting eventually in the resources to be exhausted causing a software reset.	140215	
Service observing failed when the call was answered on an analog extension using the call pickup feature.	140221	
Communication Manager sent incorrect information in the SIP contact header of the ReInvite/UPDATE message after the call via a SIP trunk reached a VDN with an announcement and was later routed out over an H.323 trunk.	140223	
When the Codec preferences on SoftFlare and on Communication Manager were different a video call made from the SoftFlare client to an H.323 Avaya one-X® Communicator resulted in two-way audio and video. When the Avaya one-X® Communicator endpoint conferenced a 96X1-SIP phone, there was no audio on SoftFlare.	140226	
When Communication Manager stations were recording using DMCC endpoints, switching between active and held calls caused the recording to fail.	140234	
Under certain internal conditions, Communication Manager incorrectly managed internal memory causing the resources to be exhausted, thus resulting into a software reset.	140246	
When a ASAI monitored station with an EC500 mapping originated a call to telecommuting extension, the call would drop.	140255	
Accessing the blank entry in the **Proxy Sel Rte Pat** field of the route pattern assigned on the locations screen while processing a call caused Communication Manager to undergo a software reset.	140269	
If a user on an IP station called a station that was forwarded to another station using an autodial button, the call was not recorded in the caller's call log.	140275	
Occasionally, Session Manager generated multiple call logs for a single call to a logged-out SIP endpoint. In such situations, Communication Manager incorrectly triggered Look Ahead Routing when the endpoint was logged out.	140279	
When a call is queued to skill, Communication Manager could intermittently undergo a software reset when processing a SIP 182 queued message.	140393	
Call pick up alerting did not work for SIP pick up group members.	140394	
Occasionally, calls made to an unregistered IP phone caused a system reset.	140397	
H.323 desk phones could not be used to dial DTMF digits into an IVR associated with a Radvision MCU. The digits either needed to be entered more than once or were recognized after 40 seconds.	140403	

Problem	Keywords	Workaround
When the calling-party name had 15 or more characters, the incoming call failed to cover to SIP voicemail.	140404	
Communication Manager will not come up on an upgrade if there are more than 500 trunk groups translated. The system would go into rolling reboots.	140410	
When there is no call center license on webLM server, Communication Manager does not forward the Communication Manager Messaging license usage statistics to the webLM server.	140414	
When the dual registration feature was used, a SIP station could not bridge onto a call that was originated by an H.323 station.	140439, 140461, 140479.	
Occasionally, Communication Manager reset when an un-named H.323 station was registered.	140485	
`Error encountered, can't complete request; check errors before retrying` occurred after changing an existing station type from SIP to H.323 that also had an EC500 entry administered in off-pbx-telephone station-mapping form. This error was seen after Communication Manager was restarted.	140490	
After a level 2 reset, Communication Manager reset again when H.323 stations were registering.	140499	
Video calls between two video-enabled H.323 Avaya one-X® Communicator phones registered on two different Communication Manager systems via a SIP trunk failed intermittently while some calls were reduced to only audio.	140502	
Occasionally, SIP calls transferred by Modular Messaging encountered a software reset.	140508	
On Communication Manager, SIP trunks in network regions without VoIP resources were unable to listen to MOH.	140516	
When the source-based routing feature was used and the originating party was a TDM trunk, Communication Manager reset.	140531	
Under specific internal conditions, Communication Manager could enter into the license error mode even with a valid license.	140559	
When multiple gatekeepers were involved in a system, Communication Manager incorrectly sequenced the alternate gatekeeper list that could, under rare circumstances, lead to a server interchange or reset.	140576	
Communication Manager may undergo a software reset while processing very specific and rare ISDN message sequence from the network.	140597	

Problems fixed in Communication Manager 6.3.6.1

Table 16: Fixes delivered to Communication Manager 6.3.6.1

Problem	Keywords	Workaround
When Communication Manager used the ASAI link version 5 or above and the system had undergone a level 2 reset since the last reboot, then the next ASAI station status query caused a system reset.	140241	
Occasionally, during heavy SIP traffic, the system reset.	140289	
Occasionally, Communication Manager reset in call scenarios that involved SIP.	140462	
Occasionally, the system displayed the `Entry is bad` error message while submitting a screen.	140493	
On receiving a SIP REINVITE message, Communication Manager incorrectly dropped a direct IP call intended for an H.323 station while negotiating codecs.	140520	
Due to an internal resource constraint that began with external network problems, Communication Manager stopped processing SIP messages.	140591	
An IP phone of type 4620, 96x0, and 96x1 that was recovering from a network disruption turned its speaker phone on after registering back to Communication Manager when the ip-direct call that was active on it dropped before the recovery was complete. This happened only when the **Near End Establishes TCP Signaling Socket** field was set to n for such phone types.	140637	
Occasionally, direct-agent calls made to an unstaffed agent with a coverage path dropped instead of following the coverage path.	140641	
Occasionally, SIP messages were not sent to the network.	140678, 140768.	
Occasionally, the additional level of SIP debug messages enabled by Avaya services resulted in a system restart.	140767	
Occasionally, Communication Manager reset when a non-ACD call that was put on hold at an ACD agent station dropped.	140807	
Under heavy traffic conditions, incorrectly managing internal resources resulted in Communication Manager undergoing a software reset to recover resources and services.	140819	
Intermittently, calls dropped after 32 seconds when SIP transactions involved provisional acknowledgements.	141045	
Improper handling of internal resources related to media sometimes caused Communication Manager to reset when processing SIP calls.	141078	
Communication Manager could undergo a level one reset when an ACD call dropped from a manual-in ACD agent's station while the agent had at least one additional call on hold and was not active on a call, such as in call transfers.	141119	
When the transmission of a SIP provisional acknowledgment failed due to a networking error, corruption of certain Communication Manager internal data was observed.	141139	
When network conditions caused active SIP calls to be considered in the connection-preservation mode, incorrect handling of internal resources caused memory exhaustion. This lead to a system reset.	141173	

Problems fixed in Communication Manager 6.3.7.0

Table 17: Fixes delivered to Communication Manager 6.3.7.0

Problem	Keywords	Workaround
Communication Manager incorrectly posted a 408 Request timeout SIP message instead of the more appropriate 480 Temporarily unavailable message when interworking calls between ISDN and SIP.	131032	
An additional whitespace at the end of a SIP message to Communication Manager resulted in garbage characters in the SIP response.	131834	
Due to an internal data corruption, IP trunks remained out of service even when the associated network regions had the media resources that were required to bring them up.	132061	
In some Department of Defense special configurations, the high-priority sshd process failed to start. Occasionally, this resulted in Communication Manager undergoing resets or frequent server interchanges.	132291	
When an H.323 station had an OPS application administered, the ASAI application incorrectly rejected the domain control request when the link was not administered as Proprietary.	132461	
When one member in a Coverage Answer Group with SIP members responded with a SIP 380 message, Communication Manager cancelled the call to all members, thus resulting in a flood of SIP messages in the network.	140079	
In case of an attended transfer, video was not initiated when a call that was transferred from a video-disabled H.323 station to a video-enabled SIP station.	140099	
When Avaya Communicator attempted to escalate an existing audio-only SIP direct media call with Radvision MCU to video, Avaya Communicator dropped from the conference.	140189	
When Communication Manager used the ASAI link version 5 or above and the system had undergone a level 2 reset since the last reboot, then the next ASAI station status query caused a system reset.	140241	
Incorrectly transferring a call to a logged-off IP station from an auto attendant triggered the Dial Plan Transparency feature causing the transfer to fail and to result in incorrect coverage treatment.	140258	
When the failover group domain table on Session Manager was configured but the failover-grp-domain-map screen was left unadministered, then, under heavy SIP traffic, Communication Manager restarted.	140289	
Calls made to a VDN with a VDN of Origin Announcement (VOA) that were put on hold during the VOA announcement forced auto-answer agents to answer the call manually.	140303	
SIP Endpoint Managed Transfer (SEMT) failed when SBC was involved and the system displayed the `480 SIPS not allowed` message for the call.	140312	
When Communication Manager was configured to Apply ringback for Auto Answer calls and VOA configured on the VDN screen, callers calling auto-answer agents through the VDN did not hear anything when the VOA was playing for the agent and they were expected to hear ringback.	140339	
When the **`change ip-interface procr`** command was used to disable one PROCR IP interface (IPv4 or IPv6), all the sockets on both PROCR interfaces were torn down, even though the other interface	140345	

Problem	Keywords	Workaround
remained enabled.		
The **`list measurements ip voice-stats`** command returned incorrect report data for media boards and network regions.	140351	
When a call is blind-transferred to a station that uses Per Button Ring Control and the call appearance was set to not ring, then the dialed number was not displayed.	140374	
When intra Communication Manager SIP calls were routed via Session Manager, the calling party information was incorrectly displayed even though the the name and number restrictions were enabled.	140375	
Trunk calls made to a station that had a SIP station bridged to it displayed the trunk name instead of the calling party number on the bridged station.	140379	
When a SIP phone that was used to make a call transferred the call to another SIP phone, the dialed number was displayed on the transferred-to SIP phone, but not on the SIP phones that were bridged on to the transferred-to SIP phone.	140380	
When EAS agents could not log on to an internally-measured skill due to exceeding the system limit of internally-measured agent/skill pairs, system administrators were not notified of the reason for the failure to log in.	140383	
The bridge appearance of a H.323 station on a SIP phone incorrectly displayed the DCS trunk name for an incoming call to the H.323 station. The call contained the DCS name and the ISDN calling party number information and the H.323 phone displayed the ISDN calling number correctly.	140390	
Even though the called party details were restricted, the called number was displayed when the **conf-dsp** button was used.	140391	
When the monitoring and monitored stations were in different CORs, the redirection override protection flag was incorrectly used from the monitoring station.	140400	
The redirect on OPTIM failure (ROOF) timer would inadvertently prevent a station-to-station call from a SIP station when IGAR was invoked.	140442	
Occasionally, the **Actual Outpulsed Digits by Preference** field on the list ars route-chosen and list aar route-chosen screens displayed incorrect digits.	140451	
When a phone that had custom labels saved was used as the source for the **`duplicate station`** command, all new phones duplicated incorrectly got the custom labels of the source phone.	140454	
Occasionally, Communication Manager underwent a software reset while processing SIP messages.	140462	
Occasionally, due to an incorrect EC500 interaction, a transferred call from a SIP endpoint resulted in a dropped call.	140468	
When changes were made to the Console Parameters screen while the IAS (Branch) was not displayed, the system displayed the following error: `Cannot enable both CAS and IAS`	140476	
When toggling between calls on a multi-line appearance phone, resuming a previously held call was not recorded by the DMCC endpoint.	140480	
Communication Manager did not display denial events of media gateways in the DSP usage report even when the system was running out of VoIP resources on the media gateway.	140483	

Problem	Keywords	Workaround
Occasionally, when a screen was submitted, the system displayed the following message: `Entry is bad`	140493	
Communication Manager did not correctly parse a SIP User-to-User header that contained a comma.	140497	
Calls that were transferred to a VDN when a VOA was playing intermittently forced auto-anwser agents to answer the call manually.	140498	
Calls made between an H323 Onex Communicator endpoint and a SIP Flare endpoint dropped when the Flare endpoint downgraded the call to audio-only after the H323 Onex Communicator endpoint stopped video.	140509	
On receiving a SIP REINVITE message, Communication Manager incorrectly dropped a direct IP call intended for an H.323 station when negotiating codecs.	140520	
An external call to a SIP endpoint that had Send All Calls enabled and had no bridge appearances of itself on other endpoints did not record the call in its call log.	140538	
When a call traversed to an IVR over a SIP trunk and then went through vector processing, the VDN return destination did not work.	140542	
Under some internal conditions, Communication Manager responded to a location request (LRQ) incorrectly with a Location confirm (LCF), instead of a location reject (LRJ), thus causing unpredictable call behavior.	140544	
When a transferred call was answered at the EC500 destination, and the principal station tried to bridge on to the call, the endpoint incorrectly displayed the information of the transferring party instead of the calling party.	140558	
Changing the remote endpoint address on the processor-channel screen left the channel in an unusasble state where no new connection could be established on the channel. The address change was made by removing and re-adding the channel on the change communication-interface processor-channels screen. The status processor-channels screen then displayed: `Session Layer Status: Awaiting Transport` and `Socket Status: Bound`.	140560	
When a station with EC500 enabled had the **Per Station CPN - Send Calling Number** field set to r, the EC500 endpoint did not display the calling party number.	140564	
Occasionally, Communication Manager underwent a software reset when SIP agents used the timed ACW feature.	140566	
SIP CAC was not applied correctly when Direct Media was enabled in an environment that only involved media gateways.	140583	
When an incoming call to the attendant was conferenced with the voicemail server through a messaging step in a vector, the generic greeting was heard instead of the personalized greeting.	140588	
During impaired network conditions, a DMCC call recording station registered as a shared control station inadvertently dropped the entire call.	140590	
Due to an internal resource constraint that began with external network problems, Communication Manager stopped processing SIP messages.	140591	
When the DTMF over IP option was set to out-of-band for a SIP trunk, an announcement in the second vector that was being processed was cut short and the call dropped.	140596	

Problem	Keywords	Workaround
When SIP Direct Media was enabled, a call that was answered from a non-SIP trunk EC500 endpoint resulted in no talkpath.	140602	
When a call came to an attendant under night service and went to a VDN and the VDN routed to a station with coverage, the call did not go to that coverage.	140613	
The IP interface screen could not be changed or removed when procr was used with G650 cabinets and the procr ip-interface was added before the G650 cabinets. Also, the system displayed the `Entry is bad` error message when the **`status station`** command was run and a station extension that was registered to the IP address associated with the procr was used.	140620	
Occasionally, calls transferred to IP agents dropped when the agent heard a brief tone to notify an incoming call and Communication Manager was configured for Multinational/Multiple locations.	140621	
Non-IP telephones experienced difficulties in entering DTMF digits into an IVR associated with a Radivison MCU. This included having to enter the digits more than once and waiting more than 40 seconds before the digits are recognized by the IVR.	140629	
In first-level overload (the first 20 seconds at an occupancy above 92.5%), the system did not deny the SIP station and trunk originations according to the overload mitigation selected on the system-parameters features screen.	140636	
An IP phone of type 4620, 96x0, and 96x1 that was recovering from a network disruption turned its speaker phone on after registering back to Communication Manager when the ip-direct call that was active on it dropped before the recovery was complete. This happened only when the **Near End Establishes TCP Signaling Socket** field was set to n for such phone types.	140637	
Occasionally, direct-agent calls made to an unstaffed agent with a coverage path dropped instead of following the coverage path.	140641	
Translations involving more than 500 IP softphones when synced with the survivable servers sometimes caused multiple software resets on the survivable servers when there should have only been one.	140645	
An internal software audit sometimes caused TTS-enabled IP phone registrations to fail and report incorrect socket usage counts on the status socket-usage screen.	140646	
There was no video after answering a call that was made from a SIP station to another SIP station over a SIP trunk connecting two Communication Managers with Direct Media disabled on one Communication Manager and enabled on the other.	140668	
Occasionally, Communication Manager experienced a Level 1 system reset when an IP telephone used the Unnamed registration feature.	140669	
A SIP One-X Agent in the telecommuter mode entering DTMF digits was not processed by Communication Manager.	140671	
When a station that was using the Unicode language was used to activate the **abr-prog** button, the display flashed temporarily with the correct information, and then went blank. After some time, the station went out of service because too many display messages were causing it to not function properly.	140677	
Occasionally, the system displayed an incorrect error message, thus	140682	

Problem	Keywords	Workaround
preventing a location number greater than location 256 to be removed.		
Incoming DIOD trunk AAR/ARS calls that were routed to a pattern with no preferences assigned caused a system reset when the calls were redirected to an unavailable attendant.	140692	
In a call center configuration with more than 1000 hunt group members, translation corruption occurred on small and medium survivable servers.	140696	
Occasionally, Communication Manager underwent a software reset when a non-ACD call that was put on hold at an ACD agent station dropped.	140807	
Communication Manager sometimes experienced a software reset while processing unusually large alphanumeric strings in the SIP URI field.	140815	
Under heavy traffic conditions, incorrectly managing internal resources resulted in Communication Manager undergoing a software reset to recover resources and services.	140819	
Occasionally, Communication Manager using the SBS feature could undergo a software reset.	140836	
An Avaya OneX CES call-back did not work when SIP Direct Media was enabled on the link between Avaya Communication Manager and CES server.	141065	
Parties joining an active conference call on the MCU that has **ALL** muted join the conference with active audio.	R123/ QC20032	Upgrade to Elite MCU 5000 V7.7.4 or later.
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.
With TLS and SRTP encryption enabled, Avaya Communicator or Avaya one-X® Communicator joining a Scopia MCU conference can sometimes lose audio or video when performing mid-call features (hold/resume, video mute/unmute, video de-escalation/escalation).	27015	Video SRTP and TLS encryption to Scopia 8.3 will be supported with Scopia 8.3 Service Pack 1. Disable SRTP and TLS to Scopia MCU and Scopia XT endpoints until Scopia 8.3 Service Pack 1 is available.
When using Communication Manager CAC, the SAT **`status ip-network-region`** screen does not show the correct tally for the **# Times Exceed BW Hit Today** field for video calls that are denied due to bandwidth limits.	131466	Run the **`display events`** command, and select **denial** as the category. You can give a date to narrow down the results. Look for denial event **2373: No Video BW available** in the **Evnt Cnt** column to ascertain the number of times the bandwidth limit was reached for a given date range. Note that the event count is for the entire system and not listed as per ip-network-region.

Problem	Keywords	Workaround
Transfers from VVX SIP to 96x0 H.323 fail.	AVA-1576	
Avaya one-X® Communicator SIP in an XT MCU conference loses video when the XT dials out to a 96x0/96x1 endpoint.	QC23240	Upgrade to XT V3.2 or later.

Problems fixed in Communication Manager 6.3.7.1

Table 18: Fixes delivered to Communication Manager 6.3.7.1

Problem	Keywords	Workaround
Occasionally, SIP messages were not sent to the network.	140768	
Occasionally, the additional level of SIP debug messages enabled by Avaya Services resulted in a system restart.	140767	
Intermittently, calls dropped after 32 seconds when SIP transactions involved provisional acknowledgements.	141045	
Improper handling of internal resources related to media sometimes caused Communication Manager to reset when processing SIP calls.	141078	
Communication Manager could undergo a level one reset when an ACD call dropped from a manual-in ACD agent's station while the agent had at least one additional call on hold and was not active on a call, such as in call transfers.	141119	
When the transmission of a SIP provisional acknowledgment failed due to a networking error, corruption of certain Communication Manager internal data was observed.	141139	
When network conditions caused active SIP calls to be considered in the connection-preservation mode, incorrect handling of internal resources caused memory exhaustion. This lead to a system reset.	141173	

Problems fixed in Communication Manager 6.3.8.0

Note: There could be loss of fixes if you upgrade to Communication Manager Release 6.3.8.0 from Communication Manager Release 6.3.4.1, 6.3.6.1, or 6.3.7.1 service packs.

Table 19: Fixes delivered to Communication Manager 6.3.8.0

Problem	Keywords	Workaround
Communication Manager underwent a software reset when the pound sign (#) was dialed after the digits to complete the call over an H.323 trunk and the ARS table had identical minimum and maximum values.	131297	
When SIP Direct Media was enabled on Communication Manager and a SIP phone called an Automatic Call Distribution (ACD) number on CS1000, the call dropped if all agents on CS1000 were busy.	131909	
An H.323 - H.323 direct tandem call involving Communication Manager, H.323 trunks, and an H.323 trunk to the Tenovis I55 dropped upon answer.	132396	
While transferring a call using the team button, the monitoring station was unable to override call redirection as it was supposed to.	140350	
Communication Manager could undergo a software reset due to missing AAR/ARS entries when reporting queue statistics for an agent on a SIP station.	140475	
An audio call made between an Avaya one-X® Communicator and Radvision XT H.323 endpoint could not be escalated to video.	140561	
When a Radvison XT endpoint originated a SIP call to a Polycom VVX endpoint, a hold/resume operation resulted in the loss of audio and video, and the call eventually dropped.	140579	
DMCC call recording failed because an incorrect calling party number was used after a hold and conference sequence involving an agent and external caller.	140584	
When the Avaya Aura Experience Portal (AAEP) was configured to use the SIP INVITE with replaces or REFER without replaces operation, some call scenarios involving transfers and conferences caused IQ/CMS to stop tracking the call.	140614	
When SA8702 was turned on to copy UCID on transfer, the UCID was not copied over when agents on SIP endpoints performed a transfer or conference operation.	140642	
In very high traffic situations, Communication Manager received an indication of exhausted audio resources from an H.248 media gateway and failed to establish audio for the endpoint from another H.248 media gateway.	140652	
When a coverage answer group extension was set before a SIP integrated voice mail server in the coverage path, the caller did not hear the correct greeting.	140665	
A SIP caller saw incorrect display when the attendant was called using a feature access code.	140701	
Calls would route incorrectly when the SIP REFER message contained a pound (#) sign.	140704	
Communication Manger failed to update the button information to soft clients in shared-control mode.	140707	
When Avaya services enabled additional logging to view display related messages, the additional information were not printed to the log files.	140709	
A port network sometimes did not recover cleanly after a network outage and required manual resetting of the IPSI board to restore service.	140717	
Calls to voice mail dropped after Communication Manager tried to send a calling	140721	

Problem	Keywords	Workaround
name with non-UTF8 characters to the voice mail server.		
When the inbound call arrived over an H.323 or ISDN trunk, Avaya Aura Experience Portal (AAEP) initiated a blind transfer to an Avaya Aura Conferencing (AAC) agent, causing the actual calling number to be replaced with the number of the AAEP.	140728	
Calls with Automatic Exclusion from a bridged call appearance to an attendant that were transferred to another station from the attendant dropped when the attendant released the call.	140732	
SIP stations did not SUBSCRIBE when a lower numbered SIP signaling group did not have administered trunk members even though a higher numbered signaling group had members administered.	140735	
Occasionally, custom button labels disappeared after the internal software maintenance audit.	140737	
An incoming crisis alert call over a SIP trunk triggered by a visiting H.323 user looked like it was originated locally.	140740	
The VDN name was omitted from the display of the agent logged into a SIP station when two or more announcements were played during vector processing prior to delivering the call to the agent.	140750	
The One-X CES mobile client did not show missed call notifications in scenarios where only the deskphone was set to ring.	140758	
The display message configured for an invalid number was not displayed when an invalid number was dialed.	140760	
Occasionally, the additional level of SIP debug messages enabled by Avaya services resulted in a system restart.	140767	
Occasionally, SIP messages were not sent to the network.	140768	
On the display of a Service Observing endpoint with Client Room Class of Service enabled, the reason code so was omitted when an observed station was active on a call.	140775	
After a server interchange, a software restart occurred on the newly active server after a discrepancy was detected in the Music-on-Hold status.	140784	
When a secure video SIP call alerted multiple endpoints and the call was answered by the endpoint that supported the video plus application session description parameter, then the call was dropped upon answer.	140785	
When the video portion of a call involving an H.323 Avaya one-X® Communicator user was ended by closing the video window, the video window continued to pop up unless terminated using the soft-key to terminate the video.	140796	
In the Dual Registration Mode, when the SIP station made an outgoing call and the H.323 station went offhook, it would automatically join the call instead of selecting a new line.	140798	
A call to a SIP station that had unconditionally forwarded all calls to voice mail continued to ring without reaching the voice mail.	140801	
Communication Manager sometimes mixed the incoming SIP trunk call to an attendant group with another incoming SIP trunk call and send wrong connected information to this second incoming SIP trunk call.	140806	
When multiple Avaya one-X® Communicator stations in the telecommuter mode were in a conference and the direct media settings between the SIP signaling and telecommuter entities over SIP trunks differed, then the parties could not hear one another in the conference.	140821	

Problem	Keywords	Workaround
When the **Ethernet Link** field was blank, adding a VAL announcement IP-interface resulted in the system displaying the following message: `Error encountered, can't complete request; check errors before retrying`	140831	
When Call Park Return Notification was enabled, and a call over a SIP trunk was returned from being parked, the (rt) reason code was omitted from the display of the station that parked the call.	140835	
SNMP retrieval of data failed from a critical reliability bearer IP interface when walking the MIB for the **`status media-processor board`** command.	140842	
When SIP direct media was enabled, the outgoing call from a SIP station did not pick the subsequent trunks using Look Ahead Routing (LAR) when there was insufficient bandwidth to route calls using the default trunk in the route pattern.	140843, 140412	
When the **`list directory`** command was running on one System Access Terminal (SAT), some other maintenance commands were blocked on other SATs. Also, if certain maintenance commands were executed on any SAT, the **`list directory`** command was blocked from executing until those maintenance commands were finished.	140849	
The ESS server with IPSI connectivity sometimes, after a reset, displayed an incorrect alarm that could not be resolved and did not exist on the main server.	140850	
Communication Manager underwent a software reset when a call was transferred by an Avaya one-X® Communicator and the display name was more than 15 characters.	140864	
Under heavy SIP traffic, SIPCC call center agents were sometimes moved to the AUX state with the Redirect On OPTIM Failure (ROOF).	140855	
In rare situations, a Communication Manager server interchange could escalate to a full system reload due to internal software conditions.	140874	
With trunk to trunk transfer set to restricted, service-observed users were allowed to transfer across public trunks when the operation should have been denied.	140881	
A **uui-info** button could not be added while running an **`add station`** command unless the **Station-Button Display of UUI IE Data** field on the CLASS OF RESTRICTION screen was enabled on the CLASS OF RESTRICTION screen for COR 1.	140886	
An incoming SIP trunk call to Communication Manager with codec G.729 and silence suppression turned on resulted in call drops when traversing multiple VDNs.	140892	
The Mask Calling Party Number (CPN) feature did not work when the call originated from a bridged appearance.	140894	
Avaya one-X® Communicator in telecommuter mode could not be used to activate the Enhanced Call Forward feature.	140897	
When a call intended for an Avaya one-X® Communicator station integrated with the One-X CES server was answered by another station of similar configuration, the missed call log was available on the station the call was intended for but not on the station that answered the call. This happened when the temporary bridge appearance was disabled for call pickup.	140905	
A call answered on a SIP bridged appearance could not be transferred to another SIP station that had the same bridged appearance button mapping as the station performing the transfer.	140909	
When three or more processor-channel links connected to CMS (mis) or IQ (ccr) adjuncts, an IQ or CMS link did not pump-up when a different CMS connected to Communication Manager failed to pump-up due to insufficient capacity	140928	

Problem	Keywords	Workaround
administered on the CMS.		
The Crisis Alert feature did not work in configurations using extensions with 11 or more digits.	140934	
When the Team Button feature was used, calls made to a hunt group that were answered by an ASAI monitored station displayed the wrong calling party information.	140943	
A cabinet that had no translations associated with it could not be removed. Instead the system displayed the following message: `Cabinet has announcement translations`	140947	
The **`server-if`** command when executed on the standby server caused the new active server to perform a software reload. When the command was executed from the active server, the software reload was not forced on the new active server.	140997	
Intermittently, Communication Manager denied a call that was placed from Avaya Communicator for Windows to Avaya Aura Conferencing.	141018	
When using a call recorder, the agent-hold time was not counted correctly on BCMS/CMS.	141039	
Communication Manager underwent a software reset when a SIP INVITE message contained a very large alphanumeric string for the request URI.	141041	
When the Separation of Bearer and Signaling (SBS) feature was used, Communication Manager could sometimes undergo a software reset.	141042	
The password strength options were not configured correctly on the standby server after a file synchronization from the active server. This caused the wrong password strength options to be used after a server interchange.	141087	
Avaya OneX CES call-back call did not work if SIP Direct Media was enabled on the link between Avaya Communication Manager and the One-X CES server.	141099	
Incoming calls over a DIOD (Direct In/Outward Dialed) trunk that are AAR/ARS (Automatic Alternate Routing/ Automatic Route Selection) routed to a route pattern with no available preferences, caused a system reset when the call was redirected to an attendant and all the attendants were busy.	141117	
Communication Manager underwent a level one reset when an ACD call dropped from a manual-in ACD agent station while the agent had at least one additional call on hold yet not active on a call, such as in call transfers.	141119	
An incoming call was not delivered to an EC500 destination when the user was provisioned for multiple devices that included the Avaya one-X® Communicator for Android and one of those clients lost WiFi connectivity abruptly	141138	
ASAI domain control messages with more than an eleven-digit calling party number were truncated to eleven digits.	141145	
Status station did not work correctly for a softphone registered as a DCP type station when there was no port network 1 administered. This could happen if port network 1 was added in cabinet 1, then port network 2 was added in cabinet 2, then cabinet 1 was removed, leaving only cabinet 2 with port network 2. This could also happen when there were only Media Gateways on the system.	141206	
Occasionally, Communication Manager could undergo a system reset when a call was answered from the Avaya Communicator for Android and then bridged-on from the desk phone.	141238	

Problems fixed in Communication Manager 6.3.9.0

Table 20: Fixes delivered to Communication Manager 6.3.9.0

Problem	Keywords	Workaround
In a dual registration configuration, the makecall request made by the third party failed when the SIP station was unregistered.		
An EC500-mapped mobile phone was used to make a call to another station in a different location. The call failed when it was made over an overlap trunk. **Note:** See 141237 under Enhancements delivered to Communication Manager 6.3.9.0.	120430	
The automatic message wait button on SIP phones did not update correctly for calls to termination extension and hunt groups.	130375	
Communication Manager failed to terminate a SIP call when the capabilities were not negotiated correctly.	131774	
When the Multiple Device Access (MDA) feature was enabled and the second device with the same extension bridged on to a call with the AAC, the first was dropped correctly but the second device did not receive a SIP call info header when the call shuffled.	132158	
When SA9122 was enabled, a call could not be made over a H.323 or SIP public trunk if the far end network region did not have a location administered.	140387	
When a device as part of the MDA feature joined an ongoing conference the display showed the domain name instead of the correct conference display.	140456	
Stations that were registered to the same extension as part of the MDA feature had no display when a call originated from the primary station.	140528	
When a DMCC station in the independent mode monitoring a held call on a SIP station unregistered, all other active calls on the SIP station dropped.	140554	
An H.323 call that routed to a coverage answer group over SIP trunks caused more bandwidth than what was necessary to be allocated.	140756	
Occasionally, Communication Manager reset to recover from problems caused due to the management of SIP session timers.	140764	
Communication Manager could not prevent internal memory exhaustion due to a routing loop between Session Manager and Communication Manager.	140783	
ASAI presence status query indicated the status as busy instead of idle for a logged-out H.323 station that had EC500 enabled.	140794	
On a one-X communicator H.323 to Avaya Communicator (AC) video call, the video window on AC remained up even after the one-X communicator pressed the stop video button.	140809	
During a network congestion event, when a call to a SIP station resulted in a ROOF condition, the OPTIM trunk port state was not cleared causing subsequent calls to any SIP station using the same OPTIM trunk to fail.	140814	
A call was erroneously dropped when the SIP phone returned a SIP 305 Use proxy response even though the EC500 call leg was ringing.	140818	
The UUI info was not passed to a station when the UUI treatment administered on the trunk group form was shared and the **send UCID** field was set to y.	140822, 140823	
When a Cisco Unified Communication Manager made an incoming SIP call to an Avaya Communication Manager where multiple media gateways were involved in the call, there was no talk-path.	140866	
Occasionally, an internal software audit in Communication Manager caused some	140867	

Problem	Keywords	Workaround
parties that were listening to integrated-music announcements to be connected to silence.		
When an agent with multiple call handling put a call on hold and then resumed the call, the agent did not receive any more calls until the current call finished.	140879	
When a user added a Dialed String with a length greater than 8 on the Precedence routing digit analysis table screen, the system displayed the following message: `Error encountered, can't complete request; check errors before retrying`	140887	
Occasionally, Communication Manager reset when Look Ahead Interflow (LAI) was used over Distributed Communications System (DCS) trunks.	140888	
The ASAI event for a transferred and conferenced call contained the incorrect called number when direct agent calling was enabled and the called party was an agent.	140895	
An internal race condition involving call shuffling sometimes caused Communication Manager to drop a SIP trunk call that was involved in a Single Step Conference.	140904	
An incoming PSTN SIP call that covered to an AAM via a SIP - adjunct hunt group when transferred out and back into Communication Manager routed the call incorrectly to the first VDN it went to instead of the newer VDN number administered in the vector. This caused the call to fail.	140908	
A call to a dual-registered station continued to ring on the bridged station even after the call dropped.	140913	
Calls made from Communication Manager to a SIP voice mail server were rejected because of a SIP 302 Moved Temporarily message in the SIP message sequence.	140936	
When the SIP direct media feature was enabled on a SIP trunk group, Communication Manager did not tandem the ACK towards the far-end if the SIP PRACK was sent after receiving the 200 OK INVITE. This caused the far end to drop the call.	140941	
SIP audio endpoints were unable to place calls to conference rooms on Radvision Multipoint Control Units (MCU's) that were H.323 integrated with Communication Manager.	140948	
When a non Time-To-Service (TTS) phone registered to a CLAN, the **`status link xxxx`** command displayed this IP station under the IP SIG GRPS & MEDIA GATEWAYS category instead of the H.323 IP PHONES category.	140959	
When a call was being transferred by the voice portal to another Communication Manager system and the agent involved in that call tried to complete the transfer, the resulting call did not have any talk path.	140963	
Occasionally, calls made to agents on H.323 phones were not recorded by the recorder because of a missing call established event.	140964	
Communication Manager sent the incorrectly encoded User to User Information (UUI) to the AES.	140978	
When the Agent/Caller Disconnect Tones feature is enabled on the system parameters features form, SIP trunks were not properly freed at the end of a call causing the CMS reports to be incorrect.	140987	
SIP endpoints were unable to invoke Calltype Analysis.	140991	
Calls that were directed to the Listed Directory Number (LDN) of the second tenant because the attendant group of the first tenant was in night service did not queue or complete to the attendant group of the second tenant.	140992	

Problem	Keywords	Workaround
Calls between Polycom SIP phones and Avaya H.323 phones resulted in one-way talk path because of incorrect payload type of the RTP stream.	140998	
When Communication Manager received a retransmitted SIP ACK message, there was no talk-path on existing calls.	141004	
For a call involving 96x1 SIP stations, when an unattended transfer is completed to a station whose EC500 destination has been logged off, the caller hears a denial tone.	141007	
An H.323 one-X softphone in the shared control mode took several minutes to register when the associated physical phone was not registered.	141012	
Occasionally, an ASAI registration status query returned incorrect states for SIP endpoints.	141016	
Incoming trunk calls to a Vector Directory Number (VDN) that performed a route-to step invoking Network Call Redirection (NCR) via the Nortel Release Link Trunk (RLT) feature failed to complete.	141017	
A one-X mobile (dual registration feature with another H.323 station) was used to make a call. If the associated H.323 physical phone went off-hook, it would result in the user joining the already active call initiated from the one-X mobile instead of initiating a new call on an unused line appearance.	141024	
On a Communication Manager system with multi-national administration, H.323 and SIP endpoints could not hear audio through the speaker on a group-page call when the media resources were provided through an H.248 media gateway.	141026j	
When misconfiguration, misadministration, or network problems caused a SIP CC station to incorrectly register that an agent was logged in, Communication Manager did not respond correctly to a change work mode operation to indicate that no agent had logged in.	141034	
When the **`list trace ras`** command was run from the main or the primary server, the system did not output the RCF (Request Confirmed) message in response to the KARRQ (Keep Alive Registration Request) message sent from the survivable server. The survivable server could be an ESS or an LSP.	141036	
Frequent IP Agent phone re-registrations caused the IP Agent license usage to increase incorrectly. This resulted in blocked registrations when the system limit was reached.	141037	
When an H.323 soft phone and physical station registered to the same extension on different network regions, the physical station was placed in the Unnamed registration. Upon registering the physical extension back to the extension the last registered network region information was lost causing Dial Plan Transparency (DPT) calls to fail.	141038	
When holding more than 40 skills, CCE agents were sometimes logged out when a **`change agent xxx auto`** command or a CMS Change Agent Skills command was run.	141040	
Occasionally, active SIP calls dropped after 32 seconds.	141045	
The call forward feature could not be activated using the feature access code on a one-X attendant endpoint when the endpoint was registered in the telecommuter mode.	141047	
When a DMCC station joined a call in the listen-only mode with direct media enabled, the call had no talk path.	141059	
Communication Manager suppressed a SIP REFER message when the 200 OK response after a reINVITE did not contain the allow-header. This caused the Network Call Redirection (NCR) feature to fail.	141071	

Problem	Keywords	Workaround
When a call reached the second coverage point that had a modular messaging server in that coverage point, the caller heard a generic greeting instead of the voice mail greeting for the called party.	141072	
Improper handling of internal resources related to media sometimes caused Communication Manager to reset when processing SIP calls.	141078	
The boot, cron and emerg logs were not sent to the syslog remote logging server when the Remote logging feature was enabled.	141105	
Calls that were parked by an attendant and subsequently timed out and returned to the attendant received the busy tone instead of moving to another idle attendant. This happened when the attendant that parked the call was busy when the call returned.	141129	
When the enhanced call forwarding feature was used, Communication Manager sent out incorrect call forward destination when the length of the external destination was shorter than the internal destination.	141133	
The **`list registered-ip-stations`** command displayed a blank network region for the DMCC registered endpoints.	141137	
Occasionally, Communication Manager converted SIP error response code 491 to a 480, thus resulting in no talk path.	141142	
The station does not display Button number 11 after the phone type is changed from 2420 to 1416 or vice-versa.	141146	
Communication Manager sometimes did not handle few number of m-lines in the SDP from the far end than what was offered causing the call to drop.	141149	
Occasionally, a very large number of calls over a H.323 trunk caused Communication Manager to reset.	141170	
When network conditions caused active SIP calls to be considered in the connection-preservation mode, incorrect handling of internal resources caused memory exhaustion. This led to a system reset.	141173	
An analog station connected to a media module on a H.248 gateway did not alert when the signal feature button was used from a DCP or an H.323 extension.	141185	
Communication Manager did not send the correct media direction in the SDP for a reINVITE message, thus causing one-way talk path when the initial INVITE was answered by the far end with a 'sendonly' tag in the SIP 18x and 200 OK responses.	141194	
The call log for a one-X CES client was not in the E.164 format when the call was picked up using the call pickup button.	141196	
Station users and call center agents observed incorrect calling-party name and number when the station user or call center agent was involved in a path replacement trombone trunk elimination operation.	141198	
When a survivable server became active, SIP signaling groups went into service for an instant and then immediately into the bypass state.	141200	
There was no talk path on a call when the following conditions were met: • The call that had an IP endpoint listening to a zip tone provided by resources on a port network • The call was transferred to another endpoint that used resources from another port network or media gateway	141201	
When the **Calling Number Style** field on the off-pbx-telephone configuration-set screen was set to PBX, Communication Manager sent incorrect calling number over the SIP trunk.	141203	

Problem	Keywords	Workaround
Communication Manager sometimes reused an internal trunk identification number too quickly and incorrectly, causing the CMS and IQ message sequence to be wrong.	141209	
Communication Manager incorrectly invoked SIP Look Ahead Routing (LAR) even after Session Manager detected a routing loop and responded with a SIP 604 response code.	141226	
When two different Avaya Communicator for Windows point-to-point video calls were merged in a conference using the Avaya Aura Conference, only two of the parties had video.	141231	
When a queue-to attendant vector step failed because there were no in-service attendants and the subsequent route-to step with coverage resulted in either the call being forwarded or sent to voice mail, the call failed.	141240	
A Polycom RMX call to an Avaya video SIP phone resulted in no video.	141278	
A queued call did not hear ringback when the SIP station was in the auto-answer mode and listening to VOA (VDN of origin Announcements).	141281	
Occasionally, a named H.323 IP phone could not re-register back to Communication Manager when the unnamed registration feature was turned on.	141283	
Customers could not enable SA8608 on the solution for Midsize Enterprise template.	141298	
When a direct SIP trunk group to another Communication Manager was fully occupied with calls, with at least one of them being a data call, then an internal trunk software audit placed the trunk group in the pending-busyout mode. This prevented newer calls from using that trunk group until the trunk group was busied out or released.	141319	
An erroneous attendant return call was placed at the attendant while the ATQA (Attendant Queue Announcement) was connected to a calling party that goes onhook before the ATQA is completed.	141320	
MLPP call preemption failed when the party that had to be pre-empted had a call waiting during the pre-emption attempt.	141321	
Call preemption failed when the preempted call involved an attendant. Call preemption to a station with a bridged call appearance sometimes caused the bridging station to lock up.	141323	
While processing MLPP SIP calls, Communication Manager encountered an internal error that incorrectly managed the system memory associated with the call, causing a software reset.	141324, 141326	
Occasionally, while processing SRTP calls, Communication Manager encountered a rare internal error that incorrectly managed the system memory associated with the call, thus causing a software reset	141325	
Occasionally, SIP data calls that involved media gateways failed. This happened when some of the media gateways supported SIP clear channel (RFC 4040) while some did not.	141329	
The system displayed an incorrect warning message when a SIP trunk group contained 255 members and a budget of 255.	141331	
The correct Block Precedence Announcement was not played to the calling party for a call that was made over an ISDN PRI trunk and blocked due to insufficient precedence level.	141332	
When an equal MLPP precedence level call was forwarded to a destination that was busy on another call, the calling party did not terminate to a Block Precedence	141333	

Problem	Keywords	Workaround
Announcement.		
A held call that was preempted by a higher precedence call did not get a preemption tone.	141334	
Incorrect DSCP values were used when an MLPP precedence call was made over a trunk group.	141335	
The Busy Not Equipped announcement was connected instead of the Block Precedence Announcement when the far end returned a SIP 486 Busy Here response.	141336	
In an MLPP call flow, call transfer from Communication Manager to a remote Cisco device failed.	141337	
If Enable Failover Event Package Subscription was turned on for SIP signaling group number 1, the system locked up and restarted.	141338	
Failover subscription messages were not routed properly through border controllers.	141340	
Communication Manager reset when the incoming SIP INVITE contained an unroutable number prefixed with a plus (+) sign.	141345	
Calls made to a VDN that routed to an agent and redirected via RONA to another VDN with a messaging step to modecode voicemail with an agent subscriber failed to complete.	141354	
An H.323 phone did not display the `Resources Unavailable` message when the bandwidth limit was reached.	141377	
A local station-to-station call was placed using short dialing. After the call was answered at the far end and an abbreviated dial button was pressed to send end-to-end DTMF tones, extra digits were sent before the digits under the abbreviated dial button.	141409	
When a call could not be routed to an agent due to a network anomaly, Communication Manager did not place the agent in AUX state to prevent further calls from being tried to such an agent.	141460	
In the feature server mode, Communication Manager failed to create a conference call consisting of SIP endpoints.	141480	

Problems fixed in Communication Manager 6.3.9.1

Table 21: Fixes delivered to Communication Manager 6.3.9.1

Problem	Keywords	Workaround
Occasionally, outbound calls could not be made from the first call appearance on SIP endpoints.The call appearance appeared to be in a hung or unusable state.	150028	
When a forced server interchange was performed, all subsequent interchanges, even interchanges that were expected to be non-service impacting, were service impacting.	150032	

Problems fixed in Communication Manager 6.3.10.0

Table 22: Fixes delivered to Communication Manager 6.3.10.0

Problem	Keywords	Workaround
Communication Manager failed to respond with a H.323 Location Reject (LRJ) message after receiving a H.323 Location Request (LRQ) message with an IP address that was unknown to Communication Manager.	141477	
Some fields on the 'list ars route-chosen' SAT (System Access Terminal) command report displayed the wrong information if a partition-route-table was assigned as the route pattern on the ARS (Automatic Route Selection) analysis table.	141458	
When the following conditions were met a call was erroneously delivered to the desk phone and also only updated the display without an audible ring: 1. The configuration included one-X Client Enablement Services. 2. The station extension was associated with a desk phone and a one-X Mobile phone. 3. The station was a member of a hunt group. 4. The user activated "Block call" from the one-X Mobile phone. 5. A call was made to the station's hunt group.	141526	
In rare situations, Communication Manager would undergo a software reset when a video enabled phone called a user on another Communication Manager server who had Multiple Device Alerting (MDA) active	140958	
Occasionally, when announcements in audio groups were transmitted between media gateways, the callers would not hear the announcement.	141384	
When Communication Manager underwent a level 2 restart the administered IP softphone count would become 0.	141243	
In some situations, Communication Manager could undergo a level 1 reset when the 'change calltype analysis' command was executed from the SAT	141571	
When a forced server interchange was performed, all subsequent interchanges, even interchanges that were expected to be non-service impacting, would be service impacting	150037	
In some conditions, a third party auto dial command was never executed by Communication Manager	141234	
When the agent login was performed through a 3rd party application the display on the agent phone would show the skill number(s) and would not clear	141435	
Incoming calls on an ISDN overlap trunk failed if a port-network was selected to service the incoming call and digit insertion/absorption was administered on the trunk	141091	
In very rare situations, IP phones would not register to the processor ethernet interface after an upgrade. The "list ip-interface" SAT command would also not Terminate	141437	
Frequent usage of the 'status ip-network-region' command could sometimes cause Communication Manager to undergo a software reset	141375	
When the station type was changed from a 2420 to a 1416 set type and the station had an active 'ringer-off' button then the message 'Object in use; please try later' would be displayed. Communication Manager would also reset the board used by such a station causing other stations using that board to go out of service	141284	
The 'list usage extension' SAT command would not list the station if it was being used as the enhanced call forward destination from another station	141473	

Problem	Keywords	Workaround
The data entered in the 'SIP trunk' field of a station would disappear if the same value was entered for this field twice in succession and the form saved	141518	
The OPS entry for a station in Communication Manager would be deleted if a bridge appearance button was added to the station through the System Manager	141633	
The 'limit-call' button would disappear from stations SIP stations after a level 4 reset of Communication Manager	141544	
In several SMI Backup web pages, the User Name would not allow "-" (hyphen) as a valid character.	141621	
The display on the bridge appearance would not update if the 'Bridged Idle Line Preference' was set to 'y' or the 'Per Button ring control' field for the bridge appearance was set to 'no-ring'	141540	
When tenant partitioning was enabled a call would fail to cover to the second coverage point if the first coverage point was a coverage answer group (CAG) and it did not answer the call	141524	
Under very high SIP call traffic conditions, an ad-hoc video conference call could cause Communication Manager to undergo a software reset	141580	
If the "mapping mode" was set to "both" for the EC500 application in the "off-pbx-telephone station-mapping" form, then when the EC500 phone originated a call, the User-User Information would be lost.	141428	
In some conditions, H.323 stations could not register to the IP interface in their own network-region.	141459	
Under rare conditions during a service pack upgrade Communication Manager may experience a system restart.	141596	
A 1692 Polycom conference phone could not register to Communication Manager if the security profile on the ip-network-region form was configured as pin-eke.	150083	
The 'status socket-usage' command would sometimes show incorrect data	141221	
When the 'Override ip-codec-set for SIP direct-media connections' is set to 'y', SIP phones participating in a Polycom audio conference bridge could sometimes cause the participants to hear disturbance.	141003	
Communication Manager sometimes managed internal resources incorrectly causing SIP calls to fail.	141451	
After the security code was changed from the station form for a H.323 station, no further outgoing calls could be made from this station	150082	
When an attendant console was logged in as an ACD agent and the agent transferred an ACD call, if the skill had timed ACW (TACW), the attendant would not go into ACW.	141475	
ANI based agent screen pops could sometimes fail when the call involved a transfer to the IVR over a SIP trunk before reaching an agent	141434	
CMS and IQ metrics on outgoing agent calls were not completely accurate when the first measured trunk failed to route the call and cause the Look Ahead Routing (LAR) feature to send the call over a different Trunk	141431	
Occasionally, outbound calls could not be made from the first call appearance on SIP endpoints.The call appearance appeared to be in a hung or unusable state.	150030	
When using TLS, media gateway registrations, SIP trunks, AES links could sometimes experience delays	141462	
After transferring a call from a Lync-SIP client to Lync-H323 client via a Lync-SIP client, all calls were dropped and all conversation windows were cleared	141626	

Problem	Keywords	Workaround
An adjunct initiated transfer using the virtual hold operation failed if the 'Prefer use of G.711 by Music Sources' option was enabled and the first codec in the ip-codec-set form was something other than G.711	141147	
Communication Manager incorrectly evaluated calling restrictions across CORs when performing a transfer using the SIP REFER without replaces action	141268	
In very rare situations, Communication Manager could incorrectly manage internal resources when servicing a large number of SIP calls and undergo a software reset	141130	
There was no talkpath at the agent phone when the incoming call over a SIP trunk was answered and a recording device was added using the single step conferencing operation	141442	
In very heavy traffic situations, Communication Manager sometimes managed internal resources incorrectly causing SIP calls to fail.	141306	
Adjunct initiated transfers, when performed too quickly, could sometimes cause the SIP trunk call to be dropped	141535	
Transfer attempt by a SIP phone behind a Sonus (TM) SBC failed.	141422	
A SIP trunk call between two Communication Manager servers would drop when the "Always use re-INVITE for Display Updates?" field was enabled	141454	
Occasionally, SIP trunk calls had no audio when they were involved in a Single Step Conference operation	141490	
SIP calls to Communication Manager failed when the SIP History-Info header included the 'tel' URI	141447	
A SIP call dropped after a blind transfer due to codec mismatch if 'SA8965 - SIP Shuffling with SDP' was enabled on Communication Manager	141457	
An ASAI alerting event was not sent when a remote EC500 agent answers an ACD call.	141258	
Ringback to the caller was not disconnected when the call to a SIP station failed because of bandwidth limitation	140983	
When a SIP station with an Enhanced Call Forward (ECF) button logs in and the ECF destination is no longer valid, Communication Manager could undergo a software reset	141471	
When an incoming trunk call is answered on a bridge appearance of the called party and is then transferred to a SIP voice mail adjunct using a second bridge appearance then, the call would not reach the correct voice mail box	141140	
When the Special Application "SA9086 - Mask CLI on PSTN calls" was enabled, the calling number was not masked by the replacement number configured on the trunk.	140988	
H.323 stations in a stub network region would not register if "Near End Establishes TCP Signaling Socket?" is set to "n" on network region form	141448	
An EC500 destination was not able to dial the "Idle appearance select" Feature Name Extension (FNE) if the call was routed over a R2MFC trunk	141484	
A direct SIP trunk between two Communication Manager servers would not be placed in-service when the 'Layer 3 test' field was set to 'y'	150086	

Problems fixed in Communication Manager 6.3.11.0

Table 23: Fixes Delivered to Communication Manager 6.3.11.0

Problem	Keywords	Workaround
Performing a 'busy-out' and 'release' operation on a SIP station could cause the station to be incorrectly placed in a disconnected state, instead of the out-of-service state.	132409	
When the SIP station that was registered over a TCP connection was already on a call, calls from an SRTP enabled H.323 station on one Communication Manager(CM1) to a SIP station on another Communication Manager(CM2) were dropped,	140513	
When the MLPP feature was enabled on Communication Manager, an incoming SIP trunk call to the user who was already on another call, caused the other user to lose the talk path.	140526	
When an H.323 endpoint from a different Communication Manager server: a. called into a Radvision or Lifesize MCU, b. was the second party (or later) into the call, c. the "SIP Endpoint Managed Transfer" was set to NO, and if the called station was also administered in the off-pbx station mapping form, the call would fail.	140685	
On page 4 of the Network Region form, changing a regular network region into a stub network region could blank out the link between the stub and the core network region.	140813	
In a Communication Manager setup with multiple MCC cabinets connected by fiber links, when a user in cabinet A, listening to an announcement transferred the call to a paging group with some users outside cabinet A, only the members in cabinet A would hear the announcement.	141025	
When the 'Prefer use of G.711 by Music Sources' field was enabled, "Music On Hold" may not be played in some cases.	141027	
Sometimes the advanced troubleshooting information required by Avaya Services was not generated by Communication Manager.	141094	
Before answering a call delivered to Communication Manager over a SIP trunk, an auto-answer agent that was "service observed" would not hear a zip tone.	141100	
When crisis alert was used without any watchers in a tenant, subsequent crisis alert calls would not work, as the crisis alert queue became full.	141154	
Under very high call traffic, a failure in one or more media gateways or port networks could sometimes cause Communication Manager to: a. incorrectly manage internal memory resources b. perform a software reset to recover the service.	141171	
When there was no previous SIP message that explicitly requested Communication Manager to allow or reject SIP UPDATE method through the allow header field, Communication Manager sometimes incorrectly dropped the call on receiving a SIP UPDATE method.	141241	
An unnamed registered station that placed an emergency call would not receive a callback from the Public Safety Access Point (PSAP).	141289	
Under very rare conditions, Communication Manager could perform a software reset, when performing "off-pbx: feature operations.	141342	
In some specific SIP messaging sequence, after the call was removed from "hold", there could be a loss of talk path.	141360	
When only two parties were left after a conference that involved a "QSIG trunk", the final display on the remaining parties did not show the calling/connected number	141394	

Problem	Keywords	Workaround
correctly.		
Even when the VDN was not involved in the call, unrelated VDN numbers were recorded in CDR records of the Communication Manager.	141430	
A One-X Mobile user that had "Send All Calls" activated could not receive calls made using "team" button on the mobile phone.	141446	
When using a. the dual registration feature with the One-X Communicator for Android and b. a 96x1 H.323 phone, the H.323 phone did not provide an audible alert.	141463	
One-X Mobile call back failed intermittently with denial event number 1316.	141464	
When a. SIP stations used the team-button and b. the "Peer Server" field on the SIP signaling group form was set to 'other', Communication Manager could sometimes undergo a software reset.	141478	
When an unregistered SIP endpoint placed an emergency call using the "emergency call" button, a callback from the PSAP would not work.	141487	
When a. "Chained Call Forwarding" was enabled and b. the bandwidth limit between two Network Regions was reached, the denial tone for the subsequent call between the two Network regions was not played.	141500	
Under some internal conditions, IP softconsoles could not register to Communication Manager.	141503	
Communication Manager user names longer than 10 characters were sometimes logged into the command history log with garbage characters that were not part of the name.	141507	
Communication Manager could undergo a software reset in configurations where: a. one Communication Manager with the Failover Event package enabled, networked via a SIP trunk to b. another Communication Manager server with the package disabled.	141511	
When a. the source based routing feature was used and b. a visiting user dials an emergency call, the resulting callback from the Public Safety Access Point (PSAP) over an ISDN overlap trunk to Communication Manager would fail.	141515	
Calls made from the One-X Communicator mobile application showed "Unknown caller".	141527	
Under some internal conditions, when executing the 'list audio-group' command, Communication Manager would display the error message "Error encountered, can't complete request; check errors before retrying".	141528	
In rare situations, when the "Reset Shift Call" feature was used, Communication Manager may undergo a software reset.	141530	
In rare conditions, on an IP soft console registration, Communication Manager could undergo a software reset.	141533	
Downstream UDP transport could cause SIP calls to drop because SIP messages could be delivered out of order.	141550	

Problem	Keywords	Workaround
Calls to a "logged in" agent that resulted in the call being covered would report incorrectly to CMS/IQ.	141551	
Communication Manager incorrectly added a new Session Description Protocol (SDP) attribute without increasing the SDP version number, causing the far end to drop the SIP call.	141555	
In rare conditions, after a server interchange, Communication Manager could experience an additional level 1 or level 2 reset.	141558	
A large number of simultaneous H.248 gateway registrations (more than 50) over TLS could drive Communication Manager into software overload, causing a disruption of service.	141559	
When the "Match BCA Display To Principal" in the COS form was turned on, the calls to a bridge appearance on 64xx type telephones would display only "ALL FROM" and no number.	141562	
When the SAC/CF (Send All Calls/Call Forward) Override by Dialing feature was enabled on the COR form of the called station, calls to that station failed to forward.	141565	
A "route-to vector" step that used ARS to route a call over an R2MFC trunk would fail.	141595	
Incoming calls to a SIP telephone would not display more than 14 characters.	141603	
Communication Manager would fail to make an ASAI third party call that included the trunk dial access code, the called number and #.	141606	
When an agent on the Avaya One-X Mobile client on a Mac a. was active on an outgoing call and b. received an incoming call he or she would experience one way talk path when returning to the call that was placed on hold after answering the incoming call.	141607	
When a. the System Access Terminal (SAT) "list trace station" command was run on a station that was simultaneously making a group page call, and b. the station buttons were pressed causing DTMF to be sent, such as an autodial button, Communication Manager would sometimes undergo a software reset.	141609	
When the call was not routed over the first preference in the route pattern, Communication Manager would not send the modified Calling Party Number.	141611	
SIP call transfers by INVITE with Replaces that included: a different UCID than the call it replaced or no UCID resulted in incorrect messages that caused CMS to count the call as "other" instead of "answered", even though the call had been answered by an agent.	141612	
When an incoming SIP trunk call to an agent that was "service observed" was placed on hold, it would sometimes result in: a. a SIP reINVITE sequence that could exhaust internal memory b. Communication Manager performing a software reset to recover service	141614	
When Communication Manager incorrectly set the Session Description Protocol attributes in the "SIP reINVITE" message, an outgoing SIP call from Communication Manager would drop after a SIP session refresh timer.	141616	
A fax transmission that fell back from T.38 to G.711 codec on Communication Manager would fail.	141622	
When a call across Communication Manager servers was made over a SIP trunk using AAR, the called station did not store the calling number in its call log.	141624	

Problem	Keywords	Workaround
For the calls routed over SIP trunks with "emer" call type, Communication Manager did not prefix the '+' to the calling party number.	141628	
Due to the incorrect management of the internal memory by Communication Manager, calls between network regions using IGAR would eventually stop working.	141629	
When there were more than 20 digits to be sent over an overlap trunk, the "ASAI third party make call" request failed.	141630	
When Communication Manager received a blind SIP REFER message to a measured VDN ,whose extension began with a digit that could have longer extensions defined, CMS recorded calls as "abandoned".	141632	
When a call was made to a VDN with the 'Destination' field set to the policy routing table number in that VDN, the 'list trace VDN' command output was incorrect.	150002	
When a. the call was executing a converse-on step in a vector after the call was queued to the same skill and b. an agent became available in a skill, an incorrect DABN message was sent from Communication Manager to the CMS/IQ.	150004	
Communication Manager logs could fill up with error messages related to SIP media attributes being rejected.	150005	
When a. a combination of ip-ip direct media and media encryption settings were used for a three party conference call between SIP phones, and b. one of the parties dropped from the call, there would be no talk path or one way audio.	150008	
When using the a. SHIFT-R feature for repeating and b. SHIFT-TAB for modifying the command line The Communication Manager System Access Terminal (SAT) terminal type 513 would not work properly.	150009	
If the "Initial INVITE with SDP for secure calls" field in the system parameters features form was disabled, an emergency call originating from a DCP station would not contain the IP address of the media gateway in the SIP via header.	150011	
When crisis alert was used without any watchers in a tenant, subsequent crisis alert calls would not work as the crisis alert queue became full.	150012, 141154	
When Special Application 9096 (Increase Paging Group Members) was enabled, H.248 Media Gateways became unresponsive, unregistered or underwent a reboot. This occurred if the gateway was providing DSP/VoIP resources for all members of a very large group page (over 100 members), when pages to the same group were performed very quickly in succession, partially due to too many simultaneous call drop messages being sent to the gateway.	150014	
In some cases, when the "Prefer use of G.711 by Music Sources "field was enabled "Music On Hold" may not be played.	150016	
When H.323 stations were service observed by two service observers, dropping a call between two H.323 stations using the drop button required the button to be pressed twice.	150017	
A large number of members in a group-page (up to 127 parties) using resources from single H.248 media gateway or TN2602/TN2302 in a port-network, could cause calls to fail.	150018, 150026	

Problem	Keywords	Workaround
When Communication Manager allocated resources from media gateways or port networks that were near their VoIP capacity, calls may fail or parties would not be added to a call.	150019	
After a hardware error condition and the subsequent alarm on the media module was resolved, the red LED sometimes was not turned off.	150020	
When the calls were a. transferred out of a SIP integrated voice mail server and b. queued to a measured skill, CMS/IQ incorrectly records the call as abandoned.	150027	
Under certain network conditions, H.323 station recovery was delayed.	150035	
An incorrect SIP message sequence related to internal timers could occasionally prevent SIP phones from originating calls.	150036	
When a. a forced server interchange was performed, and b. some interchanges were expected to be "non-service impacting", all subsequent interchanges were "service impacting".	150038	
When an auto-in agent went into pending AUX during an ACD call, and subsequently dropped, CMS/IQ would not record that the agent released call.	150041	
A call from a PSTN SIP trunk to a SIP station that was sent back out to the PSTN because of the call forward Busy/NA, could: a. experience a glare condition and b. be dropped after answer.	150042	
When Communication Manager was the second or later participant in a Radvision XT MCU based conference, there was no video.	150047	
An emergency callback that was routed through the Session Manager, from the Public Safety Access Point (PSAP), did not alert as a priority call.	150049	
When a. "music-on-hold" was provided via an analog port or announcement, and b. the parties listening to "music-on-hold" were direct-IP capable, and c. the parties listening to "music-on-hold" were registered in the same network-region as the music source, IP stations or trunks that listened to the music-on-hold sometimes heard garbled music. In some cases, the call would drop.	150055	
When a SIP station involved in an Avaya Aura Conference call was mapped to a different number in the off-pbx station mapping form on Communication Manager, the conference would fail.	150056	
Due to the missing certificates, the steps to upgrade to a new service pack could sometimes fail.	150057	
Under some internal conditions, even when there was a message waiting, Communication Manager could sometimes not light the message indication lamp on SIP telephones.	150058	
If the security profile on the ip-network-region form was configured as "pin-eke", the 1692 Polycom conference phone could not register to Communication Manager.	150059	
For a station that had: a. "Do Not Disturb" enabled and b. the field "Controlled Termination Restriction" set to 'announcement/attendant',	150063	

Problem	Keywords	Workaround
the "Leave Word Calling" feature could not be used.		
When a. the 'Prefer use of G.711 by Music Sources' and 'Prefer use of G.711 by IP Endpoints Listening to Music' fields were enabled, and b. the VoIP resources were provided by H.248 media gateways and the IP phones involved were allowed to go direct-IP, a call transferred or conferenced could sometimes result in no talk path.	150065	
When • Communication Manager was administered to use TTI and • there was at least one analog or DCP media module with some ports free, an emergency call made from such a system would incorrectly send the number administered in the "CPN, ANI for Dissociated Sets:" field in the system parameters features as the "Emergency Line Identification Number (ELIN)" instead of the emergency number field in the "ip-network-region" form.	150066	
An emergency call made from Communication Manager that used "Look Ahead Routing (LAR)" was not processed through the ARS digit-conversion table.	150067	
When the emergency call was made from a TTI or dissociated IP station, a "Public Safety Access Point (PSAP)" callback over an ISDN overlap trunk would fail.	150068	
When the station security code was changed from the SAT for a H.323 station, no further outgoing calls was possible from that station.	150084	
If the "Layer 3 test" field on the signaling group form was turned on, a newly added SIP trunk between two Communication Manager servers would not be placed into service.	150085	
When a. the "SA9123 - Re-ring CAG" members in "Adjacent Coverage Points" were enabled, and b. the call moved from one coverage point to another, no ASAI call redirect event was reported.	150090	
Communication Manager prevented the removal of a TN799 "C-LAN" board that was inserted in the slot A01 of a G650 cabinet.	150091	
For a duplicated ESS server pair that was active, forced server interchange did not place the IP signaling groups back into service.	150095	
Communication Manager incorrectly placed a SIP station in the "in-service" state after a Session Manager failover, generating duplicate ASAI domain control events.	150100	
A call from a "service observed" station to VDN which queues to an "auto answer" agent would not hear the zip tone.	150103	
Activation/Deactivation of the "Enhanced Call Forwarding (ECF)" feature by a SIP station did not notify System Manager of the change in the administrative state of the station.	150105	
In rare conditions, while processing responses to SIP messages under SIP traffic, Communication Manager could undergo a software reset.	150109	
When a. the emergency services call was processed through the tandem-calling-party-number form by traversing to the Session Manager and back,and b. the emergency services call was routed to the PSTN, the watchers on Communication Manager received truncated display of the number that called the emergency services.	150110	

Problem	Keywords	Workaround
In rare instances, after a server interchange when calls involved SIP stations or trunks, Communication Manager could undergo a software reset.	150115	
If an LSP: a. was active with registered IP stations and b. underwent a software level 1 restart, the IP stations remained registered but could no longer originated calls.	150116	
Between: a. a SIP One-X Communicator endpoint on a Mac and b. a Tandberg endpoint, an audio call could not be escalated to video.	150118	
When the Session Description Protocol contained one or more codecs over and above the clearmode codec, Communication Manager dropped an incoming SIP call.	150137	
When the System Platform CDOM was unable to establish connectivity with Communication Manager, the Communication Manager virtual machine would incorrectly be restarted.	150141	
In a JITC environment, a FIPS enabled phone, in the unnamed state with a security profile other than 'challenge', could not initiate a call.	150150	
When a call center agent entered AUX mode with a reason code, the reason code sent to CMS/IQ was the one that was previously in effect.	150166	
When a. The ‘Prefer use of G.711 by Music Sources’ and ‘Prefer use of G.711 by Announcement Sources’ fields were turned on and b. The ip-codec-set contained only the G.729 codec, there would be no talk path.	150178	
Internal conditions in Communication Manager led to a software reset.	5874	
When the KERNEL Service Pack was activated or deactivated on the S8300D system, the system would hang.	6371	
When: c. The ‘H.323 Station Outgoing Direct Media?’ field was enabled on SIP trunk, d. An H.323 station placed a call over that SIP trunk, e. The DTMF digits were entered in response to an announcement or prompt, the call dropped.	6444	
When the Retry-After header was received in a SIP message, Session Manager brought down the entity link between that Session Manager and Communication Manager.	6553	
When the ‘Prefer use of G.711 by Music Sources’ and ‘Prefer use of G.711 by Announcement Sources’ fields were turned on, a call to a SIP station, direct or through coverage, would drop 32 seconds after a resume operation was performed following the call placed on hold.	7028	

Problem fixed in Communication Manager 6.3.11.1

Table 24: Fixes delivered to Communication Manager 6.3.11.1

Problem	Keywords	Workaround
After installing a patch that contained the over-writeable patch 22038, Communication Manager prevented unpacking a patch or service pack . For more details see PSN020171	7991	
When an incoming call over an SIP trunk was received on an auto answer DCP agent that receives zip tone, no audio was received.	8093	

Problem fixed in Communication Manager 6.3.111.0

The changes that were delivered to 6.3.11.1 are available in 6.3.111.0.

Problems fixed in Communication Manager 6.3.12.0

Table 25: Fixes delivered to Communication Manager 6.3.12.0

Problem	Keywords	Workaround
When SIP endpoint with Auto Answer enabled, tried to make a SIP trunk (Direct Media On) all across unlinked Network Region, Call Admission Control failed and call was not re-routed.	2127	
While installing Communication Manager 6.2 Feature Pack 4 or later, patching errors (for example: HUNK #1 failed) were incorrectly displayed during the server setup process.	2703	
While using the Enhanced Call Forwarding (ECF) feature, Avaya Aura Communication Manager forwarded call to an internal destination instead of a configured external destination for the call originated by a Public Trunk.	2714	
SIP phone didn't receive early media announcement when call was made using an ISDN PRI trunk.	3026	
An error resulted when: a media gateway was added for the first time; assigned to a network region that already had a BACKUP SERVER adminstered; and all previous media gateways were administered with 'none' as a recovery rule.	3318	
Buttons on BUTTON MODULE #2 and BUTTON MODULE #3 were not retained if set type changed from 9640 to 9630 or 9630 to 9640.	3509	
In list measurements, coverage criteria showed incorrect coverage reason.	3516	
If a user parked a call using the call park button and disconnected within 5 seconds, MOH (Music on Hold) was dropped from the parked call if "Drop Parking User From the Call After Timeout" was enabled on the system-parameters features form.	3517	
If a main server was upgraded from an older load to a new load, and the web profile base was changed at some point afterwards, the survivable servers did not allow legitimate System Management Interface access.	3564	
Avaya Aura Communication Manager did not add ANI in the CDR report if SA7311 (Record Answering Party) was enabled.	3940	
False Filesync alarms were generated due to some race condition on an Active Communication Manager server.	4055	
If the fields "Prefer use of G.711 by IP Endpoints Listening to Music?" and "Prefer use of G.711 by IP Endpoints Listening to Announcements?" were both enabled, and the ip-codec-set serving an H.323 endpoint included G729 only, then the endpoint did not get talkpath.	4089	
A translation upgrade from an older CM release resulted in '?' being displayed in the Number Format field for SIP trunk groups.	4619	
Executing the SAT command 'get forced-takeover ipserver-interface all' on a server with no port-networks resulted in the following error being displayed: ’Error encountered, can't complete request; check errors before retrying’.	4664	
Persistent tone-detector alarms occurred for some IP Switch Interface (IPSI) versions.	4723	
During link recovery, Communication manager forced to unregister the Time-To-Service (TTS) AES (Avaya Aura Application Enablement Services) phone.	4728	
EC500 user was unable to unhold the call with Special Application SA9106 enabled on Avaya Communication Manager.	4734	
Call between SIP and DCP station having EC500 mapping was dropped when it was answered by a remote coverage point over H.323/SIP trunk having direct	4855	

Problem	Keywords	Workaround
media turned off.		
Outgoing calls over a SIP trunk group with the "Unicode Name" field set to "yes" on the System Access Terminal (SAT) trunk-group form, sent the station name (the Name field on the station form) and the native name (e.g., configured on Session Manager or set from ASA) as the Unicode Name, instead of consistently sending the native name.	4962	
When a call to agent reached the voice-mail, the call was not recorded.	4964	
Under internal conditions, Avaya Aura Communcation Manager performed a reset system 4 after the server interchange.	5117	
Avaya Aura Communication Manager rejected a NOTIFY message for Message Waiting Indication without indicating that the signaling group has been orphaned.	5175	
The Phone Number field on the off-pbx-telephone station-mapping form was required to contain the Extension Number for Off Premises Station applications.	5216	
If tandem-calling-party-num (tcpn) was administered for incoming emergency call over sip trunk, the crss-alert stations got a truncated number as emergency caller.	5230	
Calls to a station that were diverted to another station logged the wrong information in the call logs of the calling station.	5291	
If the field "Initial IP-IP Direct Media?" was set to "y" on the signaling group, when the SIP station made an outgoing call to the WATS trunk, the call was not successful.	5302	
An incoming call resulted in two call log entries.	5343	
Changes to ip-network-region administration forms related to 'stub' network regions resulted in incorrect "display failed-ip-network-region" data and unnecessary NR-CONN alarms.	5370	
Outgoing calls over a SIP trunk group with the "Unicode Name" field set to "auto" on the System Access Terminal (SAT) trunk-group form, sent the station name (the Name field on the station form) and native name (e.g., configured on Session Manager or set from ASA) as the Unicode Name, instead of consistently sending one or the other.	5375	
After Network Call Redirection(NCR) feature was invoked on Avaya Aura Communication Manager, the calls got dropped.	5376	
Rebooting the Tomcat web server process on System Manager rebooted Communication Manager Local Survivable Processor (LSP) in standby mode.	5381	
An unattended transfer call involving SIP stations experienced talkpath problem when call got forwarded or covered from the original transfer recipient.	5405	
While adding OPS entry on off-pbx-telephone station-mapping form, MCD occurred during the following steps: 1. Add OPS entry 2. Give right phone number and try to submit the form. 3. Cursor moves to "Trunk Selection" and it throws error "Required data not specified". 4. Press right arrow to move to next field.	5414	
When the field "Mask CPN/NAME for Internal Calls?" was turned on, an internal call resulted in the Calling Party Number being sent.	5454	
CMS ignored incoming trunk calls that had been answered, then transferred using the "reverse transfer" out over a SIP trunk which then returned the call using a REFER with Replaces after the returned call was answered.	5462	
When a REFER failed with 408 or other codes, CMS stopped tracking the call	5488	

Problem	Keywords	Workaround
because CM sent CMS an IDLE on the wrong trunk.		
Avaya One-X CES mapping for CM extension failed in multi-CES environment.	5562	
When there were two Service Observer(SO) for a user, and call was transferred, the second SO for user was not added to call.	5566	
If the inc-call-handling-trmt table was not configured, an internal call resulted in two call log entries on an MDA (Multiple Device Access) device.	5589	
If the SIP station made an outgoing trunk call, and if the far end trunk did not send back connected or called number, the MDA device on SIP station could not log any number in the call log, hence it was not be able to call out.	5636	
Call back from PSAP (Public Safety Answering Point) failed when the originating party had used bridge appearance to initate the emergency call.	5658	
Stub IP network regions with Dynamic bandwidth were allowed to be duplicated resulting into "duplicate entry" errors.	5679	
When ICR (Intelligent Customer Routing) transfered a call from CM, the transferred call got dropped due to the timing issue.	5690	
Calls to a BCMS (Basic Call Management System) measured skill were not counted in the Vu-stat ACD (Automatic Call Distribution) call count if the VDN (Vector Directory Number) Return Destination feature was invoked because the agent disconnected the call before the caller dropped.	5692	
When emergency call originator got PSAP (Public Safety Answering Point) call back during call forwarding timer which was routed through SM, call did not show as Priority.	5697	
An incoming call to a SIP station with MDA (Multiple Device Access) resulted in two log entries.	5724	
Calls initiated by dialing without actually going off-hook first (onhook dialed), failed with denial event 1644(DNY_ORIG) if the user was prompted for an authorization code.	5739	
After Network Call Redirection(NCR) feature was invoked on Avaya Aura Communication Manager, calls were getting dropped because of vector music step.	5770	
If call was originated by ISDN trunk and routed to SIP trunk the INVITE did not have MG IP address as last via header, if "Initial INVITE with SDP for secure calls?" field on change system-parameters features was disabled.	5810	
When one user had viewed the Secure Shell (SSH) keys on the CM web page, then a different user logged in to view the keys, the key fingerprints could not be displayed for the second user.	5816	
Crisis alert queuing and Single Respond Mode was not working as per location for incoming emergency calls over sip trunk, if the calls were not active.	5871	
The calls transferred to IP agents dropped when the agent heard a brief tone to notify an incoming call and Communication Manager was configured for Multinational/Multiple locations.	5952	
While adding OPS entry on off-pbx-telephone station-mapping form, Mini Core Dump (MCD) generated during following steps: 1. Add OPS entry 2. Give correct phone number and verify 3. Call Limit field. 4. Remove Phone number and blank out Call Limit and again administer phone number back.	5958	

Problem	Keywords	Workaround
5. Submit the form. The error occured saying "Required data not specified". Press right arrow to move to next field.		
Board-removal alarms raised against "MG-ICC" never cleared once they were logged.	5960	
An incoming SIP trunk call when covered to SIP Voicemail experienced no media path & resulted in voice mail failures.	6010	
Covered Call to mobile voice-mail sometimes got bridged on to another active call. This happened when the called station was dual-reg with 96xx H.323 and Avaya Communicator for Android (ACA) SIP registrations, ACA having EC500 administered.	6071	
Avaya Aura Conference (AAC) initiated call (adjunct origination call) did not follow the location-based route of the single station user.	6084	
After activating Call-forward feature on a SIP station, Call forward destination extension was not displayed properly.	6126	
Calls using a voicemail button on an Avaya one-X Communicator in Other Phone user mode terminated. Instead the calls rang the voicemail destination number before the call to the other phone completed and answered.	6134	
For SIP endpoints with MDA (Multiple Device Access) both a missed call & a recieved call log entries were added at the MDA endoint for a single call, if caller was a H.323 station	6160	
The CDR field for Country code displayed a NULL string for an incoming call to an announcement extension.	6161	
An incoming call over an ISDN trunk could not hear busy tone if the called station was busy and if the incoming ISDN trunk call was routed out to a SIP trunk with Look Ahead Routing (LAR).	6166	
When CM received an unknown FNU (Feature Name URI), CM treated it as an OFF-HOOK FNU and created a phantom call.	6171	
Data disappeared from the station "SIP Trunk" field when the same value was entered.	6285	
Problem 1: An internal caller had two service observers (SO), one recording device. The called party had two service observers (SO), one recording device. When the call was made, the wrong service observers were attached. Problem 2: The call was then transferred. The party transferred to had two service observers (SO), one recording device. When the transfer was complete, the wrong service observers were attached.	6296	
Calls to a station that were answered by an IP-DECT wireless handset using the team button failed to update the station display with the calling party information.	6313	
SIP call was dropped due to the glare in SIP display re-INVITE message.	6315	
Avaya Aura Communication Manager was sending default DTMF payload value instead of administered DTMF payload value set for the SIP trunk.	6333	
Incoming trunk calls dropped after the attendant released from the call, to an attendant that was transferred to an outgoing destination over the same trunk group with NCR (Network Call Redirection) enabled.	6394	
When a call went to VDN Return Destination, CMS stopped tracking the call if the call had previously been an ACD call (routed to a measured hunt group) and the dropping party was either a trunk or a simple user that did not receive the call via a group such as a hunt group.	6410	

Problem	Keywords	Workaround
An incoming SIP trunk call failed to complete when its SDP contained connection address as 0.0.0.0	6416	
The 'list trace hunt-group' command failed to output the station extension of the idle agent that could not be connected to the calling party.	6445	
After a call to an IP station/agent that recently changed the security code, calls were stuck in queue when agents were available to take the call.	6446	
Pause Character (comma) on Incoming 3PCC call message was not populated in the SIP Refer-To header by Communication Manager.	6498	
When an attendant transferred a trunk call to a SIP station, the transferred-to endpoint displayed the number of the calling-party instead of the number of the attendant.	6513	
The CPN (callling party number) mask did not work if the called station is 13 digits long with a "+" in front.	6555	
Call failed to complete when CM received a SIP message having Alert-Info header with "urn:alert" value in it.	6607, 6608	
Facility Associated Signaling (FAS) ISDN PRI signaling groups on H.248 Media Gateways (MG) were put out of service if there were similar signaling groups in Port Network (PN) 1, and then PN 1 lost connectivity to the main server.	6719	
If a user dialed a VDN (Vector Directory Number) that had a route-to number step to the Telecommuting Access Extension, the call failed.	6818	
Duplicate entries were allowed on the digit-conversion administration forms for Automatic Route Selection (ARS) / Automatic Alternate Routing (AAR).	6886	
Adding a skill to a ’logged in’ agent did not always report the change to reporting adjunct before a call for that skill was delivered to the agent. This caused the reporting adjunct to restart the link.	6899	
An incoming call over an analog trunk to the "incoming destination" configured on the analog trunk group form intermittently caused a prefix of "1" to be added to the caller ID in CDR data and reports.	7039	
Service Observers of SIP auto-answer agents or stations were not re-connected to the observed call when the agent/ station took the call off of HOLD.	7040	
A call over a PSTN trunk configured as overlap/overlap, where the digits dialed to initiate the call were dialed slowly, a "#" was sent after the digits were dialed.	7041	
When an agent handled multiple calls at the same time including at least one personal call, the resulting stream of SPI messages were not handled accurately by CMS and resulted in inaccurate reporting of agent activity.	7051	
One way voice path was observed when the call was forwarded to external entity which sent media-release parameters in answer.	7119	
A call drop issue was observed when video call went on hold between Avaya Aura Communication Manager releases CM7.0 and CM6.3. The problem was observed when Hold/Unhold Notifications? field was enabled from SIP trunk between two Communication Managers.	7272	
H.323 endpoint was unable to send Dual Tone Multi Frequency(DTMF) digits when it made an outgoing call over a SIP trunk which had "H.323 Station Outgoing Direct Media" field enabled on it.	7527	
Adding or changing multiple entries on the digit-conversion administration forms for Automatic Route Selection / Automatic Alternate Routing prompted a CM reset.	7536	

Problems fixed in Communication Manager 6.3.112.0

Table 26: Fixes delivered to Communication Manager 6.3.112.0

Problem	Keywords	Workaround
The snmpconfig tool was modified to allow a user to explicitly set the Engine ID for the server in question.	7392	
Few CM SNMP MIB descriptions are corrected as follows. - Replaced "display trunk" with "list trunk" exmple from avCmListMemTrunk and avCmListMemTrunkRange MIBs - changed read-write to read-only in those MIBs that don't have set commands	7428	
When there are no Codec Sets administered for “inter-region ip-network-regions” on page 3-20 of network regions form, the subagent restarted due to multiple Master Agent retries.	7510	

Problems fixed in Communication Manager 6.3.13.0

Table 26: Fixes delivered to Communication Manager 6.3.13.0

Problem	Keywords	Workaround
"The Mask Calling Party Number (CPN)" feature did not work when the call originated from a bridged appearance of a SIP Endpoint.	2415	
The 'status mst' SAT command displayed a misleading status when a trace was disabled by overload control.	2837	
Occasionally, a user or an agent in Telecommuter mode over a PSTN permanent SIP service link would experience issues with talkpath.	3355	
When two SIP signaling groups with the same near-end/far-end IP address, the same near-end/far-end listen ports, but different far-end domain were configured they would be put into a bypass state for a short period of time.	3496	
Communication Manager underwent a software reset when a call to a SIP extension with Multiple Device Access (MDA) configured was answered by two or more devices simultaneously under traffic conditions.	3510	
On the Communication Manager System Access terminal (SAT), the command 'list calltype route-chosen <dialed string>' failed when the dialed string was longer than 17 numeric characters.	3700	
One way talkpath was experienced by one of the parties in a conference when the Avaya Media Server (AMS) was used as the media resource and encryption was enabled on the system.	4152	
Communication Manager generated two ALERT messages for a call made to a SIP extension capable of dual registration as an H.323 as well as a SIP station when the call was being monitored by an Avaya Enablement Server (AES) Application.	5093	
During conditions when the system is in ISDN b channel overload state for more than 8 minutes, ISDN trunks would go out of service.	5384	
Communication Manager could experience one extra restart after Duplicate Processor Ethernet (DUP PE) server interchange if non-TTS (Time-To-Service) AES (Avaya Aura Application Enablement Services) phones were registered.	6007	
When "Enforce SIPS URI for SRTP?" was disabled on the Signaling Group of the SIP trunk a video call made over this trunk to an H.323 video Call Center Agent was completed with an unencrypted audio stream when it should have been completed with an encrypted audio stream. In some cases "static noise" was heard on the call.	6047	
When the caller, agent and observer were on a single port network, and the call was transferred by the agent to a VDN with a collect step, the caller was unable to enter digits.	6471	
A call to a hunt group with members as 1x-CES users in 'trigger' mode with Call Forward enabled would result in the caller hearing continuous ringback.	6568	
If the SDP offer in an incoming SIP call contained more than two crypto lines the call would fail.	6576	
The Network Region used is now displayed when running the "list trace" SAT command on a SIP station.	6603	
In a system consisting of primary and secondary Session Managers (SM) when a customer uses the secondary SM for handling SIP phones, the primary signaling links go into bypass mode when a large amount of SIP phones register to it even though the primary SM is up.	6703	
For an incoming call over SIP trunk, the calling party name on the called party's	6720	

Problem	Keywords	Workaround
display changed to display the trunk name when the following conditions are met: 1) The incoming UPDATE message contains display information in the 'From' header 2) The 'P-Asserted Identity' (PAI) header is absent 3) The 'Contact' header contains no display information.		
An incoming call made over a trunk with Direct Media disabled landed on a SIPCC agent after being connected to a VDN Origin Announcement (VOA) and was placed on hold by the agent. The agent would be unable to unhold the call if it was connected to the Communication manager over a trunk with Direct Media enabled.	6911	Disable Direct Media on the SIPCC Agent
Occasionally, when a DCP station came into service from a Personal Station Access (PSA) or Terminal Translation Initialization (TTI) state, its labels and buttons would not get downloaded.	7025	
Communication Manager underwent a WARM reset when the SIP message transaction count went beyond 10,000 under SIP traffic.	7049	
When an Avaya One-X Communicator in roadwarrior mode was configured with multiple button modules, the button labels would not be downloaded.	7075	
Occasionally, an unrelated trunk call would be dropped unexpectedly if another call was dropped due to codec mismatch.	7079	
Calls made using the "PIN Check for Private Calls" Feature would fail once the caller dialed the Feature Access Code for "PIN Check for Private Calls", followed by the PIN.	7116	
Communication Manager allowed agents logging in through Application Enablement Services (AES) associated Applications to login overriding Tenant Permissions.	7142	
When a One-X CES Application has the options "Ring Phone" and "Call Back Phone" configured on two different Mobile Numbers then the mobile phone which was already busy on a call would get notification for a second incoming call overriding the One-X CES configuration.	7251	
Large number of Log files would be generated as a result of some unwanted Proc Errors being logged under high traffic conditions of SIP to SIP station and trunk calls.	7253	
When a call made over a SIP trunk with 'Network Call Redirection' (NCR) enabled was merged into conference by the answering station, the calling party experienced one-way talkpath.	7327	
Under very specific conditions, the Media Gateway was prevented from registering to the Primary Communication Manager which had become available again after a Local Survivable Processor(LSP)/ Enterprise Survivable Server (ESS) failover.	7328	
Occasionally, a trunk call made using an EC500 mapped extension would get disconnected after a few seconds after answering under the following configuration: 1) Two station users belonged to different Tenant Partitions with Calling Restrictions between the partitions 2) An EC500 mapping on one of the users used a trunk group which was placed under a third Tenant Partition which did not have any calling restrictions 3) The call was made using the external EC500 mapped extension to the other	7430	

Problem	Keywords	Workaround
station user.		
For an X-ported station, the Call Detail Recording (CDR) report displayed null characters for the "country-to" field.	7461	
Calls to the Audix hunt group failed and were dropped when Audix answered the calls.	7511	
While making a One-X callback call to a destination requiring an Authorization Code, the customer was not able to enter the Authorization code.	7569	
An incoming SIP call being routed through a 'goto,if ani' vector step would not be routed correctly due to ANI mismatch if the incoming ANI began with a '+'.	7577	Set "Remove '+' from incoming Called/Calling/ Alerting/ Diverting/ Connected Numbers?" to "Yes" on the SIP Signaling Group form for the incoming call.
Under conditions of SIP trunk call traffic and congestion in the IP Network Communication Manager underwent WARM resets.	7608	
Customer could not access entries that were not defined in the Dial Plan Analysis Table form when executing "change public-unknown-numbering" or "change private-numbering" command to remove or modify entries. Customer would see the "Ext code inconsistent with dialplan" error.	7644	
Large number of Log files would be generated as a result of some unwanted Proc Errors being logged under high traffic conditions of SIP to SIP station calls.	7696	
When a SIP station originated a 911 call over a SIP trunk with Direct Media disabled being used as the PSTN trunk, a call back from Public Safety Answering Point (PSAP) failed to terminate on the 911 caller.	7717	
No ACK/NACK message was being sent to the Computer Telephony Integration (CTI) application for a third party call control (3PCC) call if a VDN Origin Announcement (VOA) was being played on the user's station. This caused CTI call control, for example, transfers, to fail.	7724	
The mobile extension was being displayed on the called station instead of the extension of the EC500 station to which the mobile station was mapped when the field "Location to Route Incoming Overlap Calls" on 'off-pbx-station mapping configuration-set' form was set to "trunk".	7747	
Users experienced loss of video when a point to point video Call was initiated between two Avaya Communicator for Windows (ACW) clients with video enabled and one of the clients was monitored by an Avaya Call Recorder (ACR).	7806	
When an incoming ISDN call made to a local station covered to an attendant who used the "Transfer to Voice Mail" Feature Access Code to transfer this call to the Avaya Aura Voice Mail (AAM) system connected to the Communication Manager via a SIP trunk, the Attendant would be incorrectly identified as the originator of the call instead of the ISDN Calling number who left the voice mail.	7815	
When a call was originated using Computer Telephony Integration (CTI) by an ASAI application integrated with the Communication Manager with 3rd Party Call Control (3PCC) enabled, '#' was outpulsed over the trunk once the call was answered.	7905	

Problem	Keywords	Workaround
After installing a patch that contained the over-writeable patch 22038, Communication Manager prevented unpacking a patch or service pack. For more details see PSN020171.	7991	
An administered cabinet would not be removed unless the ip-interface PROCR was removed first.	7992	
Occasionally, users experienced no talkpath for SIP incoming calls terminating to DCP agents on a Port network when the DCP agents were in auto-answer mode in a Call Center setup with multiple Port networks and Gateways.	8093	
Call Suppression did not operate correctly if the called SIP endpoint's extension number was modified through the use of inserting digits using a route pattern entry.	8102	
Third Party Endpoints E.g. CISCO endpoints capable of Binary Floor Control Protocol would not be able to join a call.	8136	
When a call involving a remote worker logged into Aura using Avaya Aura Session Border Controller for Enterprise (ASBCE) was made then under a specific SIP messaging sequence the signaling connection between Communication Manager and Avaya Session Manager (ASM) disconnected.	8146	
IP Agent 6.0 soft client couldn't register to Communication Manager after the CM was upgraded to 6.3.8.0 or higher.	8169	
In a very rare scenario where internal CM data ended up in a mismatched state, hunt groups were not being monitored when they were configured to be monitored by BCMS/CMS or other similar applications that utilize monitoring events.	8184	
When path replacement was used and an announcement had finished playing in a vector, it would propose a new path replacement, which could requeue the call thus affecting the oldest call waiting statistics.	8221	
Occasionally, an ISDN trunk call made using the OneX Communicator Redial feature caused the Communication Manager to undergo a WARM reset.	8236	
Users experienced no talkpath when a direct media call was auto-answered by an agent administered with an auto-answer Automatic Call Distribution (ACD) group and logged in on a SIPCC station that was not administered as auto-answer.	8247	
On rare occasions, a system interchange would fail and lead to a system reload.	8253	
Under very specific SIP messaging sequences erroneous Proc Errors were generated resulting in a large number of log files.	8280	
Call Centers where a large number of registrations and unregistrations occurred frequently and a large number of stations were added or removed consistently experienced translation corruption on the Communication Manager preventing further administration change relating to stations.	8302	
A message containing the incorrect extension number sent by Communication Manager caused applications like Proactive Outreach Manager (POM) to drop the entire on-going call instead of only the Audix Recorder which had been recording the call.	8303	
Under specific conditions misleading Proc Errors would be logged generating large number of Log Files when IP Terminal Translation Initialization (TTI) and Personal Station Access (PSA) were enabled on the system.	8307	
CM failed to provide the call identifier for a monitored station in response to a value query from Application Enablement Services (AES). This resulted in unexpected behavior from AES integrated applications utilizing this information,	8317	

Problem	Keywords	Workaround
depending on how the information was being used.		
Occasionally, Communication Manager fails to reroute a call correctly when SA8904 (Location Based Calltype Analysis) is enabled and an Application Enablement Server (AES) associated Application initiates a Redirection of the call.	8337	
An active call would get dropped if Communication Manager sent a 'sips' UPDATE message with a sip contact header.	8393	
Call Logs would not get stored on the hard phones when the originator in Shared Control Mode on a SoftPhone placed a call using the Softphone by dialing the ARS Feature Access Code (FAC) followed by the Direct Inward Dialing (DID) Extension.	8399	
When a stub region connected to a core region which was in turn connected to a third region failed the Network Region connectivity test, Communication Manager was left with alarms that could not be resolved.	8400	
When any Avaya Aura Messaging (AAM) or Communication Manager Messaging (CMM) system had one or more Trap Receiver Destination(s) configured, one or more GAM “Border Process Registration Failed” traps would be erroneously generated on the system.	8438	
Occasionally, busying out an IP station and then releasing it did not force the station to unregister when multiple stations were being accessed simultaneously by one of the Communication Manager components.	8496	
No customer visible impact. SIP trunks and DCP agents remained on their respective Port Networks (PN)/Gateways rather than both entities being terminated on a DCP agent's PN when the Agent was in auto-answer mode and received an incoming call.	8514	
Under specific conditions, MOH was heard by both the agent and the calling party when the call was transferred to an agent over a SIP trunk and MOH was enabled on the system.	8525	
Use of the Enhanced Call Pickup Alerting feature sometimes caused a system reset with very large groups under heavy call traffic loads.	8533	
Under specific configurations for 96X1SIP Stations, a wrong number was displayed in the "Call-Limit" field on the 'off-pbx-telephone station-mapping' form.	8537	
When "Criteria for Logged Off/PSA/TTI Stations" was enabled on the form "system-parameters coverage-forwarding" on Communication Manager and the first point of coverage was not registered calls did not cover to the second point of coverage.	8564	
When different values for Session Refresh Timer were administered at the near-end and far-end of a SIP trunk, then in very rare circumstances users experienced loss of talkpath after a transfer took place.	8666	
A long duration SIP trunk call being hosted on a Scopia Multi Conferencing Unit (MCU) was being dropped. Calls which had been active for two session refresh intervals would get dropped.	8752	
When there were greater than 1653 entries in the "ARS DIGIT CONVERSON TABLE" for a single location, this form could not be changed and the message "Error Encountered, Can't Complete Request" was displayed on the screen.	8887	
Call Logs on MDA (Multiple Device Access) were different if the incoming ISDN trunk and the called station were in different locations.	8907	
If initially there were two or more calls being alerted on the station which had	9044	

Problem	Keywords	Workaround
been administered with a team button for the terminating station, then when this monitored station answered a call, the corresponding alert was not cleared at the monitoring station's display.		
A SIP endpoint registered on a One-X Communicator is administered with a team button which is used for monitoring another SIP endpoint which has Enhanced Call Forward activated for external calls. When an internal call was received at the monitored station, the visual toast alert on the monitoring station failed to discontinue once the call was answered by the monitored station.	9259	

Problems fixed in Communication Manager 6.3.113.0

Table 29: Fixes delivered to Communication Manager 6.3.113.0

Problem	Keywords	Workaround
"The Mask Calling Party Number (CPN)" feature did not work when the call originated from a bridged appearance of a SIP Endpoint.	2415	
The 'status mst' SAT command displayed a misleading status when a trace was disabled by overload control.	2837	
Occasionally, a user or an agent in Telecommuter mode over a PSTN permanent SIP service link would experience issues with talkpath.	3355	
When two SIP signaling groups with the same near-end/far-end IP address, the same near-end/far-end listen ports, but different far-end domain were configured they would be put into a bypass state for a short period of time.	3496	
Communication Manager underwent a software reset when a call to a SIP extension with Multiple Device Access (MDA) configured was answered by two or more devices simultaneously under traffic conditions.	3510	
On the Communication Manager System Access terminal (SAT), the command 'list calltype route-chosen <dialed string>' failed when the dialed string was longer than 17 numeric characters.	3700	
One way talkpath was experienced by one of the parties in a conference when the Avaya Media Server (AMS) was used as the media resource and encryption was enabled on the system.	4152	
Communication Manager generated two ALERT messages for a call made to a SIP extension capable of dual registration as an H.323 as well as a SIP station when the call was being monitored by an Avaya Enablement Server (AES) Application.	5093	
During conditions when the system is in ISDN b channel overload state for more than 8 minutes, ISDN trunks would go out of service.	5384	
Communication Manager (CM) could experience one extra restart after Duplicate Processor Ethernet (DUP PE) server interchange if non-TTS (Time-To-Service) AES (Avaya Aura Application Enablement Services) phones were registered.	6007	
When "Enforce SIPS URI for SRTP?" was disabled on the Signaling Group of the SIP trunk a video call made over this trunk to an H.323 video Call Center Agent was completed with an unencrypted audio stream when it should have been completed with an encrypted audio stream. In some cases "static noise" was heard on the call.	6047	
When the caller, agent and observer were on a single port network, and the call was transferred by the agent to a VDN with a collect step, the caller was unable to enter digits.	6471	
A call to a hunt group with members as 1x-CES users in 'trigger' mode with Call Forward enabled would result in the caller hearing continuous ringback.	6568	
If the SDP offer in an incoming SIP call contained more than two crypto lines the call would fail.	6576	

Problem	Keywords	Workaround
The Network Region used is now displayed when running the "list trace" SAT command on a SIP station.	6603	
In a system consisting of primary and secondary Session Managers (SM) when a customer uses the secondary SM for handling SIP phones, the primary signaling links go into bypass mode when a large amount of SIP phones register to it even though the primary SM is up.	6703	
For an incoming call over SIP trunk, the calling party name on the called party's display changed to display the trunk name when the following conditions are met: 1) The incoming UPDATE message contains display information in the 'From' header 2) The 'P-Asserted Identity' (PAI) header is absent 3) The 'Contact' header contains no display information.	6720	
An incoming call made over a trunk with Direct Media disabled landed on a SIPCC agent after being connected to a VDN Origin Announcement (VOA) and was placed on hold by the agent. The agent would be unable to unhold the call if it was connected to the Communication manager over a trunk with Direct Media enabled.	6911	Disable Direct Media on the SIPCC Agent
Occasionally, when a DCP station came into service from a Personal Station Access (PSA) or Terminal Translation Initialization (TTI) state, its labels and buttons would not get downloaded.	7025	
Communication Manager underwent a WARM reset when the SIP message transaction count went beyond 10,000 under SIP traffic.	7049	
When an Avaya One-X Communicator in roadwarrior mode was configured with multiple button modules, the button labels would not be downloaded.	7075	
Occasionally, an unrelated trunk call would be dropped unexpectedly if another call was dropped due to codec mismatch.	7079	
Calls made using the "PIN Check for Private Calls" Feature would fail once the caller dialed the Feature Access Code for "PIN Check for Private Calls", followed by the PIN.	7116	
Communication Manager allowed agents logging in through Application Enablement Services (AES) associated Applications to login overriding Tenant Permissions.	7142	
When a One-X CES Application has the options "Ring Phone" and "Call Back Phone" configured on two different Mobile Numbers then the mobile phone which was already busy on a call would get notification for a second incoming call overriding the One-X CES configuration.	7251	
Large number of Log files would be generated as a result of some unwanted Proc Errors being logged under high traffic conditions of SIP to SIP station and trunk calls.	7253	
When a call made over a SIP trunk with 'Network Call Redirection' (NCR) enabled was merged into conference by the answering station, the calling party experienced one-way talkpath.	7327	
Under very specific conditions, the Media Gateway was prevented from registering to the Primary Communication Manager (CM)	7328	

Problem	Keywords	Workaround
which had become available again after a Local Survivable Processor(LSP)/ Enterprise Survivable Server (ESS) failover.		
Occasionally, a trunk call made using an EC500 mapped extension would get disconnected after a few seconds after answering under the following configuration: 1) Two station users belonged to different Tenant Partitions with Calling Restrictions between the partitions 2) An EC500 mapping on one of the users used a trunk group which was placed under a third Tenant Partition which did not have any calling restrictions 3) The call was made using the external EC500 mapped extension to the other station user.	7430	
For an X-ported station, the Call Detail Recording (CDR) report displayed null characters for the "country-to" field.	7461	
Calls to the Audix hunt group failed and were dropped when Audix answered the calls.	7511	
While making a One-X callback call to a destination requiring an Authorization Code, the customer was not able to enter the Authorization code.	7569	
An incoming SIP call being routed through a 'goto,if ani' vector step would not be routed correctly due to ANI mismatch if the incoming ANI began with a '+'.	7577	Set "Remove '+' from incoming Called/Calling/ Alerting/ Diverting/ Connected Numbers?" to "Yes" on the SIP Signaling Group form for the incoming call.
Under conditions of SIP trunk call traffic and congestion in the IP Network Communication Manager underwent WARM resets.	7608	
Customer could not access entries that were not defined in the Dial Plan Analysis Table form when executing "change public-unknown-numbering" or "change private-numbering" command to remove or modify entries. Customer would see the "Ext code inconsistent with dialplan" error.	7644	
Large number of Log files would be generated as a result of some unwanted Proc Errors being logged under high traffic conditions of SIP to SIP station calls.	7696	
When a SIP station originated a 911 call over a SIP trunk with Direct Media disabled being used as the PSTN trunk, a call back from Public Safety Answering Point (PSAP) failed to terminate on the 911 caller.	7717	
No ACK/NACK message was being sent to the Computer Telephony Integration (CTI) application for a third party call control (3PCC) call if a VDN Origin Announcement (VOA) was being	7724	

Problem	Keywords	Workaround
played on the user's station. This caused CTI call control, for example, transfers, to fail.		
The mobile extension was being displayed on the called station instead of the extension of the EC500 station to which the mobile station was mapped when the field "Location to Route Incoming Overlap Calls" on 'off-pbx-station mapping configuration-set' form was set to "trunk".	7747	
Users experienced loss of video when a point to point video Call was initiated between two Avaya Communicator for Windows (ACW) clients with video enabled and one of the clients was monitored by an Avaya Call Recorder (ACR).	7806	
When an incoming ISDN call made to a local station covered to an attendant who used the "Transfer to Voice Mail" Feature Access Code to transfer this call to the Avaya Aura Voice Mail (AAM) system connected to the Communication Manager via a SIP trunk, the Attendant would be incorrectly identified as the originator of the call instead of the ISDN Calling number who left the voice mail.	7815	
When a call was originated using Computer Telephony Integration (CTI) by an ASAI application integrated with the Communication Manager with 3rd Party Call Control (3PCC) enabled, '#' was outpulsed over the trunk once the call was answered.	7905	
After installing a patch that contained the over-writeable patch 22038, Communication Manager prevented unpacking a patch or service pack. For more details see PSN020171.	7991	
An administered cabinet would not be removed unless the ip-interface PROCR was removed first.	7992	
Occasionally, users experienced no talkpath for SIP incoming calls terminating to DCP agents on a Port network when the DCP agents were in auto-answer mode in a Call Center setup with multiple Port networks and Gateways.	8093	
Call Suppression did not operate correctly if the called SIP endpoint's extension number was modified through the use of inserting digits using a route pattern entry.	8102	
Third Party Endpoints E.g. CISCO endpoints capable of Binary Floor Control Protocol would not be able to join a call.	8136	
When a call involving a remote worker logged into Aura using Avaya Aura Session Border Controller for Enterprise (ASBCE) was made then under a specific SIP messaging sequence the signaling connection between Communication Manager and Avaya Session Manager (ASM) disconnected.	8146	
IP Agent 6.0 soft client couldn't register to Communication Manager (CM) after the CM was upgraded to 6.3.8.0 or higher.	8169	
In a very rare scenario where internal CM data ended up in a mismatched state, hunt groups were not being monitored when they were configured to be monitored by BCMS/CMS or other similar applications that utilize monitoring events.	8184	
When path replacement was used and an announcement had finished playing in a vector, it would propose a new path replacement, which could requeue the call thus affecting the oldest	8221	

Problem	Keywords	Workaround
call waiting statistics.		
Occasionally, an ISDN trunk call made using the OneX Communicator Redial feature caused the Communication Manager to undergo a WARM reset.	8236	
Users experienced no talkpath when a direct media call was auto-answered by an agent administered with an auto-answer Automatic Call Distribution (ACD) group and logged in on a SIPCC station that was not administered as auto-answer.	8247	
On rare occasions, a system interchange would fail and lead to a system reload.	8253	
Under very specific SIP messaging sequences erroneous Proc Errors were generated resulting in a large number of log files.	8280	
Call Centers where a large number of registrations and unregistrations occurred frequently and a large number of stations were added or removed consistently experienced translation corruption on the Communication Manager preventing further administration change relating to stations.	8302	
A message containing the incorrect extension number sent by Communication Manager caused applications like Proactive Outreach Manager (POM) to drop the entire on-going call instead of only the Audix Recorder which had been recording the call.	8303	
Under specific conditions misleading Proc Errors would be logged generating large number of Log Files when IP Terminal Translation Initialization (TTI) and Personal Station Access (PSA) were enabled on the system.	8307	
CM failed to provide the call identifier for a monitored station in response to a value query from Application Enablement Services (AES). This resulted in unexpected behavior from AES integrated applications utilizing this information, depending on how the information was being used.	8317	
Occasionally, Communication Manager (CM) fails to reroute a call correctly when SA8904 (Location Based Calltype Analysis) is enabled and an Application Enablement Server (AES) associated Application initiates a Redirection of the call.	8337	
An active call would get dropped if Communication Manager sent a 'sips' UPDATE message with a sip contact header.	8393	
Call Logs would not get stored on the hard phones when the originator in Shared Control Mode on a SoftPhone placed a call using the Softphone by dialing the ARS Feature Access Code (FAC) followed by the Direct Inward Dialing (DID) Extension.	8399	
When a stub region connected to a core region which was in turn connected to a third region failed the Network Region connectivity test, Communication Manager was left with alarms that could not be resolved.	8400	
When any Avaya Aura Messaging (AAM) or Communication Manager Messaging (CMM) system had one or more Trap Receiver Destination(s) configured, one or more GAM “Border Process Registration Failed” traps would be erroneously generated on the system.	8438	

Problem	Keywords	Workaround
Occasionally, busying out an IP station and then releasing it did not force the station to unregister when multiple stations were being accessed simultaneously by one of the Communication Manager components.	8496	
No customer visible impact. SIP trunks and DCP agents remained on their respective Port Networks (PN)/Gateways rather than both entities being terminated on a DCP agent's PN when the Agent was in auto-answer mode and received an incoming call.	8514	
Under specific conditions, MOH was heard by both the agent and the calling party when the call was transferred to an agent over a SIP trunk and MOH was enabled on the system.	8525	
Use of the Enhanced Call Pickup Alerting feature sometimes caused a system reset with very large groups under heavy call traffic loads.	8533	
Under specific configurations for 96X1SIP Stations, a wrong number was displayed in the "Call-Limit" field on the 'off-pbx-telephone station-mapping' form.	8537	
When "Criteria for Logged Off/PSA/TTI Stations" was enabled on the form "system-parameters coverage-forwarding" on Communication Manager and the first point of coverage was not registered calls did not cover to the second point of coverage.	8564	
When different values for Session Refresh Timer were administered at the near-end and far-end of a SIP trunk, then in very rare circumstances users experienced loss of talkpath after a transfer took place.	8666	
A long duration SIP trunk call being hosted on a Scopia Multi Conferencing Unit (MCU) was being dropped. Calls which had been active for two session refresh intervals would get dropped.	8752	
When there were greater than 1653 entries in the "ARS DIGIT CONVERSON TABLE" for a single location, this form could not be changed and the message "Error Encountered, Can't Complete Request" was displayed on the screen.	8887	
Call Logs on MDA (Multiple Device Access) were different if the incoming ISDN trunk and the called station were in different locations.	8907	
If initially there were two or more calls being alerted on the station which had been administered with a team button for the terminating station, then when this monitored station answered a call, the corresponding alert was not cleared at the monitoring station's display.	9044	
A SIP endpoint registered on a One-X Communicator is administered with a team button which is used for monitoring another SIP endpoint which has Enhanced Call Forward activated for external calls. When an internal call was received at the monitored station, the visual toast alert on the monitoring station failed to discontinue once the call was answered by the monitored station.	9259	

Problems fixed in Communication Manager 6.3.14.0

Table 27: Fixes delivered to Communication Manager 6.3.14.0

Problem	Keywords	Workaround
Communication Manager was unable to match the correct entry in the Tandem Calling Party Number form in some cases, especially when the options "any length" and "any CPN" were chosen.	4164	
While using the Avaya Aura Messaging (AAM) Call Language Preservation feature on an AAM which had been configured to use Basic or Unattended transfers, customer language preference needed to be re-entered when the call had been transferred from the AAM to another station and covered back to AAM.	4268	
The Caller was unable to enter digits when prompted when an Avaya Aura Experience Portal(AAEP) Consultative Transfer call CM landed on a VDN (Vector Directory Number) over a SIP trunk when a Media Gateway was being used as a resource.	4656	
A dual registered SIP station was unable to enter any DTMF digits after dialing the EC500 FNE (Feature Name Extension) when the following conditions were met: 1) Direct Media was enabled for SIP signaling 2) Incoming SIP trunk had "DTMF over IP" set to 'rtp-payload' 3) Outgoing SIP trunk had "DTMF over IP" set to 'out-of-band'.	4821	
Upon expiry of the "Time Reminder on Hold" timer, the display on a OneX Attendant changed from the caller's identity to the trunk name when the incoming call to CM landed on a VDN which routed the call to an attendant with 'cov' set to 'y' on the vector 'route-to' step.	5012	
When an EC500 user, mapped to an enterprise user A, placed a call to another enterprise user B, the extension of the EC500 user instead of that of Station A was displayed on Station B.	5016	
In a Call Center setup integrated with a Call Management System (CMS) and an Application Enablement Server (AES), the CMS would receive two AUX work events when a call answered by an agent in Automatic Call Distribution (ACD) mode was placed on hold by the AES application and merged with a second call to complete a conference.	5063	
When an entry for a SIP extension existed in the Uniform Dial Plan Analysis table as AAR but there was no mapping for the extension in the AAR table, the SIP station would not be able to perform any button pushes.	5341	
When a user answered a call using a bridge appearance, no talkpath was experienced under the following conditions: 1) The bridged user and the principal user were in different network regions 2) The calling party was in the same network region as that of the principal 3) IGAR (Inter Gateway Alternate Routing) was enabled and invoked between the two network-regions.	5509	
Occasionally, in a Call Center Elite environment, where the stations were configured with message waiting lamps, updates to the lamps would be delayed.	5570	
Automatic Call Back (ACB) did not work for calls to a SIP station that had Call Forward activated for all calls. ACB would be activated, but the callback attempt would always fail returning busy tone to the originator even though the called party was idle.	6768	
An unexpected ISDN cause value 18 (CV_NUR) was returned when an incoming ISDN call termed on an unregistered SIP station causing undesired behavior.	6808	

Problem	Keywords	Workaround
The 'logv', 'logc', and 'logw' log evaluation commands had a race condition that occasionally caused unwanted prompts when examining log data on busy systems. This was observed on systems where the CM logs were writing more than once per second.	7297	
CDR (Call Detail Recording) for the second call leg was not generated, when a tandemed call, made from a station on CM1, to a Non-Optim SIP Station registered to an SM (Session Manager) was blind transferred to an H.323 station on CM2 under the following conditions: 1. H.323 trunk administered between CM1 and CM2. 2. SIP trunk administered between CM2 and SM (Session Manager).	7329	
Under a very specific SIP messaging sequence, a login attempt by a SIP agent would cause a segmentation fault.	7533	
For an incoming call over an R2MFC trunk which was answered by a SIP desk phone using the Call Pickup feature, no Caller ID (CID) was displayed on the SIP desk phones.	8190	
The "Simultaneous Active Adjunct Controlled Calls" count on the display capacity form kept growing, never decreasing back to zero.	8434	
In rare instances, executing a "list trace station" command on CM SAT (System access Terminal) for an extension that had a large number (hundreds) of bridged stations caused a system reset.	8513	
When an Avaya OneX-Attendant transferred an external incoming call to an external extension over any trunk, the far end did not receive the calling party's identity.	8578	
Team button interactions with calls involving service links caused CM resets.	8593	
Under rare circumstances, using ISDN or H.323 trunks caused CM to reboot.	8675	
With (1) "Client Room" enabled on the COS-group of the Calling Party and (2) Coverage Path set on the Called Party, when the covered call was answered and then dropped at the Called Party's coverage point, the call logs showed the Caller's identity as "unavailable".	8678	
A call routing to a Vector Directory Number (VDN) with "Allow VDN Override" set to "no" (disabled) generated two call records in applications that utilize monitoring events such as Basic Call Management System (BCMS) or Call Management System (CMS).	8679	
CM administration denied inserting the wildcard character '*' within the number string for Call Forward destinations.	8695	
Under a very specific SIP messaging sequence, a registration attempt followed by a message summary event by a SIP station caused a segmentation fault.	8732	
When an unattended transfer was initiated by a DECT station then upon expiry of the Transfer Recall timer, the returned call on DECT phone was not shown as a Priority Call.	8746	
Central Office (CO) trunk members remained active for failed transfers with "No Disconnect Supervision" set, even after all parties disconnected from the call.	8749	
In a Call Center Elite environment configured with Softphones, when the command "display capacity" was active on the SAT (System Access Terminal) while a trace was being collected, a large number of log statements would get generated which proved difficult to interpret.	8784	
In a Call Center Elite environment with Coverage Paths defined for agents, Calls failed to route to any coverage path after the first coverage path, for agents which	8791	

Problem	Keywords	Workaround
were logged out.		
Occasionally, attempts to change the "Mapping Mode" for previously defined station mappings on the "off-pbx-telephone station-mapping" through SAT failed and the message "Error Encountered, Can't Complete Request" was displayed on the screen.	8828	
When a SIP trunk call from an Avaya Experience Portal was transferred back to the Vector Directory Number (VDN) which was configured with a "collect" step without a "wait" step before it, the calling user was unable to enter any DTMF digits when prompted.	8840	
An incorrect Calling Party Number was being displayed for a 3PCC call made over a trunk when "SA8481-Replace Calling Party Number with ASAI ANI" was enabled in a Communication Manager with 3PCC configured and integrated with an Avaya Aura Application Enablement Services (AES) application.	8867	
In a Call Center setup integrated with an Avaya Aura Application Enablement Services (AES) application and Call Monitoring Applications when a device information query was launched through the AES application for various devices, other call monitoring applications ran into delays due to extra bytes being sent by CM.	8875	
Calls to a one-X Client Enablement Services (CES) station, with EC500 enabled and all call appearances busy, routed to coverage instead of returning busy tone when "Busy" coverage criteria was disabled.	9011	
An unregistered station with 2 call appearances and one of them busy, returned ring back instead of busy tone when "Restrict Last Appearance" was enabled on the station.	9012	
In a Call Center setup with IP agents configured, Zip tone was being heard by the calling party as well as the observers on the call which was being observed when it should have been heard only by the agent when different network region resources were preferred by the incoming call, agent and the service observer.	9014	
Computer Telephony Integration (CTI) applications were unable to pass DTMF tones to CM during digit collection steps of vector processing.	9022	
Agents that logged in with "Forced Agent Logout Time" configured did not get logged out after the logout time interval.	9059	
While executing the "change station" command for a SIP extension, the user was unable to tab through the "SIP trunk" field and would see the message "Field cannot be blank" even when the field was correctly populated.	9062	
When the (SAT) System Access Terminal command "status station" was executed on a station administered without a Network Address Translation (NAT) IP Address, the "Native NAT Address" field displayed an IP address instead of "not applicable".	9108	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	9191	
Occasionally, sockets would get stranded after a server interchange on Duplex CM systems with a large number of IP endpoints.	9194	
When the called party who was a member of a Call Pickup group answered a call made to it using the "Active Appearance Select" Feature Name Extension (FNE), the other members of the pickup group continued to be alerted endlessly for the same call.	9207	
With SA9124 "AACC Connected Information Enhancement" enabled on a CM integrated with an Application Enablement Server (AES), when a call to a Domain Controlled station was placed on hold at the station, the generated 'HELD' event	9209	

Problem	Keywords	Workaround
did not contain the extension of the station.		
No Call Initiated Event message was being sent to the CTI application for a third party call control (3PCC) originated call. Call recording failed for such calls.	9215	
The "status station" System Access Terminal (SAT) command showed a truncated call forward number for an Emergency Location Extension (ELE) station.	9217	
Calls to a Coverage Answer Group (CAG) would fail under the following conditions: 1) One SIP member was in ONEX_TRIGGER MODE and another SIP member was in a logged off state OR 2) One H.323 or DCP member was available and another SIP member was in logged off state	9233	
The System Access Terminal (SAT) commands "list trace station" and "list trace tac" failed to display all SIP messages associated with the traced call.	9235	
For an incoming call if the URI was an exact match with a dial plan configured on the CM but the called party was not an actual extension on the CM and the Special Application "SA8904 - Location Based Call Type Analysis" was enabled, then the administration on the field "DID/ Tie/ISDN/SIP Intercept Treatment" was overridden and unexpected tones were heard.	9244	
When "Chained Call Forwarding" was enabled on the system, and a call failed to forward because all the trunks were busy, the caller would hear silence instead of hearing reorder tone.	9255	
When Communication Manager received an incoming SIP trunk call which had very low Max-Forwards header value, the call failed to route to the EC500 extension.	9302	
On rare occasions, calls to a group page caused the call appearance of a paged digital station to be left in a busy state for a few minutes until an audit cleared the call appearance.	9327	
With "SA8481 - Replace Calling Party Number with ASAI ANI" enabled on CM and 3PCC configured and integrated with an Application Enablement Server (AES), a 3PCC call made over a trunk resulted in a modified Calling Party Number (CPN) being sent in the "P-Asserted-Identity", "Contact", and "From" headers in the SIP message.	9344	
In a Call Center Elite system integrated with an AES application and a reporting adjunct such as CMS, when digits outside of the range of digits (0-9) were sent to reporting for the calling party number (ANI), the message was ignored by reporting.	9345	
When a call was originated using Computer Telephony Integration (CTI) by an ASAI application integrated with the Communication Manager with 3rd Party Call Control (3PCC) enabled, '#' was outpulsed over the trunk once the call was answered.	9357	
When there were more than 500 IP stations registered through AES which were sharing the same IP address, the "reset ip-station" command did not complete.	9371	
When a call terming on a virtual station with its two points of coverage as Coverage Answer group and Hunt group for Messaging Server respectively, landed on the second point of coverage, a wrong greeting was played to the caller.	9399	
No Team Button lamp update was seen if the monitoring station which was administered as 96x1 set type and registered using Avaya OneX Communicator attempted to initiate a call.	9413	
A call landing on a VDN, when redirected to a SIP adjunct, such as a Voice Mail	9428	

Problem	Keywords	Workaround
Server, over a SIP trunk, resulted in an error and the caller was unable to leave a voice mail.		
When an announcement was in the process of being connected to a call, and the caller disconnected before the commencement of the announcement, the announcement media (e.g., VAL board) displayed errors on the system	9429	
Under certain conditions where the cabinet had been removed first, Avaya services were unable not remove X.25 data-module translations.	9477	
A call from a SIP station that dialed "0" to reach an attendant failed.	9489	
The "statapp" command did not accurately report Messaging "Up/Down" status.	9497	
Computer Telephony Integrated (CTI) stations were unable to send DTMF tones to CM if Service Observing (SO) warning tones were enabled on CM.	9524	
When Direct Media was enabled for Sip calls and an unregistered SIP station had EC500 activated, then calls to the EC500 extension failed if LAR (Look Ahead Routing) was used to route the call.	9546	
Calls from Computer Telephony Integration (CTI) applications with Third Party Call Control (3PCC) failed if the Call Detail Recording (CDR) account access code was 4 digits and the access code was sent in the private data of the CSTA MakeCall event.	9566	
Occasionally, incorrect Location IDs for measured PRI trunks involving Media Gateways would be sent to reporting adjuncts, such as Call Management System (CMS).	9597	
When the field "Caller ANI during pickup alert" was disabled on the Calling Party's COR, then for a call terminating on the SIP members of a Call Pickup group, the Caller's ANI was incorrectly being displayed.	9627	
Under a very specific SIP messaging sequence for a call involving a Transfer, reporting adjuncts, such as CMS or IQ, were unable to accurately track the call if the transferred leg was redirected to a number that failed to route successfully.	9653	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	9725	
When a call routed to a One-X Attendant via a Vector Directory Number (VDN) with Coverage enabled in the route-to step of the vector, a wrong Caller ID was displayed on the Attendant.	9782	
Occasionally, there was no talkpath on a SIP conference call when the field "Use reINVITE for display update" was enabled on CM.	10760	
Occasionally, there was no talkpath on a SIP conference call when the field "Use reINVITE for display update" was enabled on CM.	10992	

Problems fixed in Communication Manager 6.3.114.0

Table 29: Fixes delivered to Communication Manager 6.3.114.0

Problem	Keywords	Workaround
Communication Manager was unable to match the correct entry in the Tandem Calling Party Number form in some cases, especially when the options "any length" and "any CPN" were chosen.	4164	
While using the Avaya Aura Messaging (AAM) Call Language Preservation feature on an AAM which had been configured to use Basic or Unattended transfers, customer language preference needed to be re-entered when the call had been transferred from the AAM to another station and covered back to AAM.	4268	
The Caller was unable to enter digits when prompted when an Avaya Aura Experience Portal(AAEP) Consultative Transfer call CM landed on a VDN (Vector Directory Number) over a SIP trunk when a Media Gateway was being used as a resource.	4656	
A dual registered SIP station was unable to enter any DTMF digits after dialing the EC500 FNE (Feature Name Extension) when the following conditions were met: 1) Direct Media was enabled for SIP signaling 2) Incoming SIP trunk had "DTMF over IP" set to 'rtp-payload' 3) Outgoing SIP trunk had "DTMF over IP" set to 'out-of-band'.	4821	
Upon expiry of the "Time Reminder on Hold" timer, the display on a OneX Attendant changed from the caller's identity to the trunk name when the incoming call to CM landed on a VDN which routed the call to an attendant with 'cov' set to 'y' on the vector 'route-to' step.	5012	
When an EC500 user, mapped to an enterprise user A, placed a call to another enterprise user B, the extension of the EC500 user instead of that of Station A was displayed on Station B.	5016	
In a Call Center setup integrated with a Call Management System (CMS) and an Application Enablement Server (AES), the CMS would receive two AUX work events when a call answered by an agent in Automatic Call Distribution (ACD) mode was placed on hold by the AES application and merged with a second call to complete a conference.	5063	
When an entry for a SIP extension existed in the Uniform Dial Plan Analysis table as AAR but there was no mapping for the extension in the AAR table, the SIP station would not be able to perform any button pushes.	5341	
When a user answered a call using a bridge appearance, no talkpath was experienced under the following conditions: 1) The bridged user and the principal user were in different network regions 2) The calling party was in the same network region as that of the principal 3) IGAR (Inter Gateway Alternate Routing) was enabled and invoked between the two network-regions.	5509	
Occasionally, in a Call Center Elite environment, where the stations were configured with message waiting lamps, updates to the lamps would be delayed.	5570	

Problem	Keywords	Workaround
Automatic Call Back (ACB) did not work for calls to a SIP station that had Call Forward activated for all calls. ACB would be activated, but the callback attempt would always fail returning busy tone to the originator even though the called party was idle.	6768	
An unexpected ISDN cause value 18 (CV_NUR) was returned when an incoming ISDN call termed on an unregistered SIP station causing undesired behavior.	6808	
The 'logv', 'logc', and 'logw' log evaluation commands had a race condition that occasionally caused unwanted prompts when examining log data on busy systems. This was observed on systems where the CM logs were writing more than once per second.	7297	
CDR (Call Detail Recording) for the second call leg was not generated, when a tandemed call, made from a station on CM1, to a Non-Optim SIP Station registered to an SM (Session Manager) was blind transferred to an H.323 station on CM2 under the following conditions: 1. H.323 trunk administered between CM1 and CM2. 2. SIP trunk administered between CM2 and SM (Session Manager) .	7329	
Under a very specific SIP messaging sequence, a login attempt by a SIP agent would cause a segmentation fault.	7533	
For an incoming call over an R2MFC trunk which was answered by a SIP desk phone using the Call Pickup feature, no Caller ID (CID) was displayed on the SIP desk phones.	8190	
The "Simultaneous Active Adjunct Controlled Calls" count on the display capacity form kept growing, never decreasing back to zero.	8434	
In rare instances, executing a "list trace station" command on CM SAT (System access Terminal) for an extension that had a large number (hundreds) of bridged stations caused a system reset.	8513	
When an Avaya OneX-Attendant transferred an external incoming call to an external extension over any trunk, the far end did not receive the calling party's identity.	8578	
Team button interactions with calls involving service links caused CM resets.	8593	
Under rare circumstances, using ISDN or H.323 trunks caused CM to reboot.	8675	
A call routing to a Vector Directory Number (VDN) with "Allow VDN Override" set to "no" (disabled) generated two call records in applications that utilize monitoring events such as Basic Call Management System (BCMS) or Call Management System (CMS).	8679	
CM administration denied inserting the wildcard character '*' within the number string for Call Forward destinations.	8695	
Under a very specific SIP messaging sequence, a registration attempt followed by a message summary event by a SIP station caused a segmentation fault.	8732	
When an unattended transfer was initiated by a DECT station then upon expiry of the Transfer Recall timer, the returned call on DECT phone was not shown as a Priority Call.	8746	

Problem	Keywords	Workaround
Central Office (CO) trunk members remained active for failed transfers with "No Disconnect Supervision" set, even after all parties disconnected from the call.	8749	
In a Call Center Elite environment configured with Softphones, when the command "display capacity" was active on the SAT (System Access Terminal) while a trace was being collected, a large number of log statements would get generated which proved difficult to interpret.	8784	
In a Call Center Elite environment with Coverage Paths defined for agents, Calls failed to route to any coverage path after the first coverage path, for agents which were logged out.	8791	
Occasionally, attempts to change the "Mapping Mode" for previously defined station mappings on the "off-pbx-telephone station-mapping" through SAT failed and the message "Error Encountered, Can't Complete Request" was displayed on the screen.	8828	
When a SIP trunk call from an Avaya Experience Portal was transferred back to the Vector Directory Number (VDN) which was configured with a "collect" step without a "wait" step before it, the calling user was unable to enter any DTMF digits when prompted.	8840	
An incorrect Calling Party Number was being displayed for a 3PCC call made over a trunk when "SA8481-Replace Calling Party Number with ASAI ANI" was enabled in a Communication Manager with 3PCC configured and integrated with an Avaya Aura Application Enablement Services (AES) application.	8867	
In a Call Center setup integrated with an Avaya Aura Application Enablement Services (AES) application and Call Monitoring Applications when a device information query was launched through the AES application for various devices, other call monitoring applications ran into delays due to extra bytes being sent by CM.	8875	
Calls to a one-X Client Enablement Services (CES) station, with EC500 enabled and all call appearances busy, routed to coverage instead of returning busy tone when "Busy" coverage criteria was disabled.	9011	
An unregistered station with 2 call appearances and one of them busy, returned ring back instead of busy tone when "Restrict Last Appearance" was enabled on the station.	9012	
In a Call Center setup with IP agents configured, Zip tone was being heard by the calling party as well as the observers on the call which was being observed when it should have been heard only by the agent when different network region resources were preferred by the incoming call, agent and the service observer.	9014	
Computer Telephony Integration (CTI) applications were unable to pass DTMF tones to CM during digit collection steps of vector processing.	9022	
Agents that logged in with "Forced Agent Logout Time" configured did not get logged out after the logout time interval.	9059	
While executing the "change station" command for a SIP extension, the user was unable to tab through the "SIP trunk" field and would	9062	

Problem	Keywords	Workaround
see the message "Field cannot be blank" even when the field was correctly populated.		
When the (SAT) System Access Terminal command "status station" was executed on a station administered without a Network Address Translation (NAT) IP Address, the "Native NAT Address" field displayed an IP address instead of "not applicable".	9108	
The "snmpget" command failed on a Local Survivable Processor (LSP) or an Enterprise Survivable Server (ESS) first time after a CM reboot if the 'acpsnmp' login not logged in after the system reboot or someone reset the SAT using the 'reset login-ID'.	9190	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	9191	
Occasionally, sockets would get stranded after a server interchange on Duplex CM systems with a large number of IP endpoints.	9194	
When the called party who was a member of a Call Pickup group answered a call made to it using the "Active Appearance Select" Feature Name Extension (FNE), the other members of the pickup group continued to be alerted endlessly for the same call.	9207	
With SA9124 "AACC Connected Information Enhancement" enabled on a CM integrated with an Application Enablement Server (AES), when a call to a Domain Controlled station was placed on hold at the station, the generated 'HELD' event did not contain the extension of the station.	9209	
No Call Initiated Event message was being sent to the CTI application for a third party call control (3PCC) originated call. Call recording failed for such calls.	9215	
The "status station" System Access Terminal (SAT) command showed a truncated call forward number for an Emergency Location Extension (ELE) station.	9217	
In a setup involving a Standalone Messaging server, SNMP SMI Agent Status Page incorrectly displayed connected subagents to the MasterAgent as not connected.	9220	
Calls to a Coverage Answer Group (CAG) would fail under the following conditions: 1) One SIP member was in ONEX_TRIGGER MODE and another SIP member was in a logged off state OR 2) One H.323 or DCP member was available and another SIP member was in logged off state	9233	
The System Access Terminal (SAT) commands "list trace station" and "list trace tac" failed to display all SIP messages associated with the traced call.	9235	
For an incoming call if the URI was an exact match with a dial plan configured on the CM but the called party was not an actual extension on the CM and the Special Application "SA8904 - Location Based Call Type Analysis" was enabled, then the administration on the field "DID/ Tie/ISDN/SIP Intercept Treatment" was overridden and unexpected tones were heard.	9244	
When "Chained Call Forwarding" was enabled on the system, and a	9255	

Problem	Keywords	Workaround
call failed to forward because all the trunks were busy, the caller would hear silence instead of hearing reorder tone.		
When Communication Manager received an incoming SIP trunk call which had very low Max-Forwards header value, the call failed to route to the EC500 extension.	9302	
On rare occasions, calls to a group page caused the call appearance of a paged digital station to be left in a busy state for a few minutes until an audit cleared the call appearance.	9327	
With "SA8481 - Replace Calling Party Number with ASAI ANI" enabled on CM and 3PCC configured and integrated with an Application Enablement Server (AES), a 3PCC call made over a trunk resulted in a modified Calling Party Number (CPN) being sent in the "P-Asserted-Identity", "Contact", and "From" headers in the SIP message.	9344	
In a Call Center Elite system integrated with an AES application and a reporting adjunct such as CMS, when digits outside of the range of digits (0-9) were sent to reporting for the calling party number (ANI), the message was ignored by reporting.	9345	
When a call was originated using Computer Telephony Integration (CTI) by an ASAI application integrated with the Communication Manager with 3rd Party Call Control (3PCC) enabled, '#' was outpulsed over the trunk once the call was answered.	9357	
When there were more than 500 IP stations registered through AES which were sharing the same IP address, the "reset ip-station" command did not complete.	9371	
When a call terming on a virtual station with its two points of coverage as Coverage Answer group and Hunt group for Messaging Server respectively, landed on the second point of coverage, a wrong greeting was played to the caller.	9399	
No Team Button lamp update was seen if the monitoring station which was administered as 96x1 set type and registered using Avaya OneX Communicator attempted to initiate a call.	9413	
A call landing on a VDN, when redirected to a SIP adjunct, such as a Voice Mail Server, over a SIP trunk, resulted in an error and the caller was unable to leave a voice mail.	9428	
When an announcement was in the process of being connected to a call, and the caller disconnected before the commencement of the announcement, the announcement media (e.g., VAL board) displayed errors on the system	9429	
Under certain conditions where the cabinet had been removed first, Avaya services were unable not remove X.25 data-module translations.	9477	
A call from a SIP station that dialed "0" to reach an attendant failed.	9489	
The "statapp" command did not accurately report Messaging "Up/Down" status.	9497	
Computer Telephony Integrated (CTI) stations were unable to send DTMF tones to CM if Service Observing (SO) warning tones were enabled on CM.	9524	

Problem	Keywords	Workaround
When Direct Media was enabled for Sip calls and an unregistered SIP station had EC500 activated, then calls to the EC500 extension failed if LAR (Look Ahead Routing) was used to route the call.	9546	
Calls from Computer Telephony Integration (CTI) applications with Third Party Call Control (3PCC) failed if the Call Detail Recording (CDR) account access code was 4 digits and the access code was sent in the private data of the CSTA MakeCall event.	9566	
Occasionally, incorrect Location IDs for measured PRI trunks involving Media Gateways would be sent to reporting adjuncts, such as Call Management System (CMS).	9597	
When the field "Caller ANI during pickup alert" was disabled on the Calling Party's COR, then for a call terminating on the SIP members of a Call Pickup group, the Caller's ANI was incorrectly being displayed.	9627	
Under a very specific SIP messaging sequence for a call involving a Transfer, reporting adjuncts, such as CMS or IQ, were unable to accurately track the call if the transferred leg was redirected to a number that failed to route successfully.	9653	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	9725	
When a call routed to a One-X Attendant via a Vector Directory Number (VDN) with Coverage enabled in the route-to step of the vector, a wrong Caller ID was displayed on the Attendant.	9782	
Occasionally, there was no talkpath on a SIP conference call when the field "Use reINVITE for display update" was enabled on CM.	10760	
Occasionally, there was no talkpath on a SIP conference call when the field "Use reINVITE for display update" was enabled on CM.	10992	

Problems fixed in Communication Manager 6.3.115.0

Table 28: Fixes delivered to Communication Manager 6.3.115.0

Problem	Keywords	Workaround
In a Communication Manager configuration that uses Media Gateways, with Ephemeral Caching disabled, or AMS (Avaya Media Server) as media resources, Endpoints experienced loss of talkpath when a call that utilized the "Direct Media" feature initially, reverted to TDM.	3717	
Inter Communication Manager call over H.323 trunk with shuffling enabled, the calling endpoint was being service observed (SO) when the called endpoint answered the call manually, the endpoints experienced loss of talkpath. The issue occurred when AMS (Avaya Media Server) was being used as the media resource.	5515	
On rare occasions, Avaya Aura Communication Manager underwent a reset while sending messages to H.323 stations or trunks.	5789	
In a Call Center setup with VOA (VDN of Origin. Announcement) configured, VOA was not played for stations that supported Shuffling and used the "voa-repeat" button configured on them, when Ephemeral Caching was disabled on media-gateways that were used as media resources.	5989	
Under rare circumstances, owing to corruption in a SIP station's internal data structure, the station's call appearances became unresponsive which made the station unable to make or receive calls.	6803	
Under a very specific SIP message sequence, the History-Info was omitted for inbound calls from SBC (Session Border Controller).	8536	
In a Call Center setup with feature "VDN Return Destination" enabled, a SIP inbound VDN (Vector Directory Number) call had no talk path after it was been redirected by a One-X Agent in Telecommuter mode using a SIP trunk as the service link.	8722	
The "list trace button" SAT (System Access Terminal) command did not allow two endpoints to be traced simultaneously.	9025	
Occasionally, when there were media gateways present in the system, Communication Manager underwent a system reset due to corrupted data structures.	9105	
A conference call initiated by an agent using the Bridged Appearance of another station that received the initial call, resulted in the call appearance of the Principal station to become unresponsive when the Agent dropped off from the conference.	9234	
When a SIP endpoint transferred a call using a Feature Access Code, that contained a special character, to a Voice Mail, the transfer failed.	9318	
If an H.323 IP phone location-parameter used an E.164 international number format, and call forward was active on the phone, the phone call log did not include the "+" character or the country code.	9396	
In case of SIP or H.323 Endpoints in Dual Registration mode and using Extend Call functionality, such as EC500, when the H.323 station had the first call appearance active on a call, and had another incoming call ringing on the second call appearance, then in some cases when the second call would be extended to a mobile phone, the two calls would incorrectly merge into a conference call.	9408	
Under rare circumstances, Communication Manager experienced a segmentation fault when an H.323 call was received on a shared signaling group.	9442	
The "list trace vdn" and "list trace vector" System Access Terminal (SAT) commands failed to output the active VDN and VDN Return Destination numbers.	9496	
When a One-X Communicator soft phone sent an RRQ message with the Network Region Number through the "login.xml" file, incorrect information caused media	9547	

Problem	Keywords	Workaround
resources to be selected from incorrect Media Gateways.		
When multiple emergency calls were made from the same station, the PSAP (Public Safety Answering Point) call back call was not always treated as a priority call.	9567	
The "list trace tac" System Access Terminal (SAT) command was enhanced to add a "c" option to support a calling number.	9740	
Under very rare circumstances, in a Duplex system, after a server interchange, the new standby server did not relinquish the Processor Ethernet's alias address causing all IP applications, such as stations, gateways to operate abnormally.	9744	
Occasionally, when the field "Remove '+' from Incoming Called/Calling/Alerting/ Diverting/Connected Numbers?" was set to "n" on the SIP signaling group CM SAT (System Access Terminal) form that was being used, Communication Manager would experience a system restart when the far end SIP client sent an invite with an exceptionally long user string in the request URI.	9767	
Under very rare circumstances in a call center configuration with work at home agents (i.e., agents using service links) where agent calls were being recorded, calls for these agents using service links (telecommuter/another telephone number/another phone mode) were not recorded.	9778	
In a Call Center setup including a CMS (Call Management System) and integrated with SIP adjuncts, such as IVR (Interactive Voice Response), ICR (Intelligent Customer Routing), or AAEP (Avaya Aura Enterprise Portal), under a very specific SIP messaging sequence, CMS stopped tracking internal calls after routing out to a SIP adjunct.	9825	
In a call center configuration with tandem Communication Managers using the VDN Return Destination feature, an outgoing trunk call that was transferred to a Vector Directory Number (VDN) and answered by an agent, failed to route to the VDN return destination if the call routed to another trunk due to no disconnect supervision being set on the original outgoing trunk.	9828	
In a Call Center Elite system, where the CM (Communication Manager) was integrated and configured with a CTI (Computer Telephony Integration) Adjunct, when a call with a service observer active used CTI to outpulse DTMF and attempted to transfer this call within five seconds, the transfer failed.	9829	
When the field "V6 Node Names" was administered on the "survivable-processor" SAT (System Access terminal) form on CM, removal of the "Service Type" entries for CDR (Call Detail Recording) on the "ip-services" SAT form failed and the message "Error Encountered, Can't Complete Request" was displayed on the screen.	9837	
In a configuration with SIP phones with auto-dial buttons programmed with a code which is a concatenation of the "Call Park" Feature Access Code (FAC) and the "Answer Back" FAC, Users were unable to park a call using an auto-dial button on a SIP phone.	9917	
Occasionally, in a Call Center Elite system, an agent logging into an AWOH station (Administration Without Hardware) and then logging out caused the Communication Manager to restart.	9921	
Communication Manager experienced a reset when it received SIP response message that contained more media lines in SDP than what was sent in the outgoing INVITE.	9924	
On rare occasions with Shared Control Call Recording active, when a Forced Unregistration request was sent to the X-ported station on which a One-X agent was logged in, the agent continued to be logged in instead of being logged out.	9960	

Problem	Keywords	Workaround
On a system with the following configuration: 1. "Display Information With Bridged Call" set to "y" on the CM SAT (System Access Terminal) form "system-parameters features" 2. A station Station-B administered with three bridged appearances of another station Station-A in place of the default three call-appearances. When Station-A went off-hook on its first call-appearance, the second off-hook attempt by its bridged station Station-B resulted in no dial tone being heard by Station-B.	10006	
Occasionally, when a call landed on a principal station, that had its Bridged Appearances administered on both a SIP station and an H.323 Station, and the subsequent dialog event state Publish message contained the H.323 station in the Request URI, the Bridged Appearance on the SIP Station did not alert.	10010	
Occasionally, VOA (VDN of Origin Announcement) played on a call which was being recorded by a Call Recorder, resulted in loss of talkpath when: 1) MOH (Music on Hold) was enabled on the system 2) SIP Stations sent media to a Media Gateway.	10024	
Occasionally, Avaya Communication Manager experienced a system reboot under SIP call traffic.	10038	
Communication Manager experienced a reset when it received SIP response message that contained more media lines in SDP than what was sent in the outgoing INVITE.	10082	
Call involving H.323 Stations dropped when the main Communication Manager took over from an ESS (Enterprise Survivable Server).	10095	
Communication Manager added an extra "0" to odd length SIP User-to-User Information (UUI), which caused SIP UUI to be incorrect.	10104	
In a Call Center Elite system, when an agent received a call for a skill with 'Timed ACW' active and went into 'pending ACW' mode during the call then after the call dropped, the agent did not go into timed ACW mode as expected. If the next call received by the agent was for a skill that did not have 'Timed ACW' active, the information from the prior call was used and the agent went into 'Timed ACW' mode after the second call.	10126	
An incorrect ISDN Cause Value generated by Communication Manager for a call to a station, that was busy on another call, triggered a routing loop that caused the calling party to listen to silence for 15 seconds before being dropped.	10136	
For Auto Call Back calls initiated by a SIP Station administered with an Auto Call Back button, the "call-back" call, would get dropped.	10138	
In a Call Center setup that has H.323 Agents and VOA (VDN of Origin. Announcement) configured, under conditions of high traffic with media gateways running at full capacity, system logs reported MEMPOOL errors.	10255	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	10408	
Users were neither able to deactivate the station lock feature using the feature button, nor use the on-hook dialing feature, nor answer calls using team button when: 1. "(SA8967) - Mask CLI and Station Name for QSIG/ISDN Calls?" was enabled (set to "y") on the "system-parameters special-applications" SAT (System Access Terminal) form and, 2. The "send-nn" feature button was set as "permanent" on an H.323 extension.	10426	
Communication Manager underwent a system reset under conditions of SIP traffic	10428	

Problem	Keywords	Workaround
comprising of tandem calls for call scenarios, such as transfer, that generate display update messages.		
Occasionally, announcements recorded on media-gateways or TN2501 VAL boards were played for a very short duration.	10446	
When incoming calls over SIP trunks that did not support REFER, landed on a vector that contained a "route-to" step with "~r", the calls failed to progress to the next step if the "route-to" step with " ~r" failed to complete.	10451	
Occasionally, calls that were transferred to an extension with bridged appearances/stations, that covered to voice mail, failed to drop the ringing bridged stations.	10462	
The "send-nn" feature button lamp did not get updated when 1. "(SA8967) - Mask CLI and Station Name for QSIG/ISDN Calls?" was enabled (set to "y") on the "system-parameters special-applications" SAT (System Access Terminal) form and 2. The "send-nn" feature button was administered on a button module and set as "permanent".	10463	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	10465	
A Non-Avaya H.323 Call Recording port did not receive any audio for SIP Direct Media Calls to Avaya H.323 Agents.	10474	
Occasionally, announcements could not be recorded on Media-gateways or VAL boards.	10491	
In a CM (Communication Manager) configuration consisting of more than one CTI (Computer Telephony Integration) applications performing call control via Application Enablement Services (AES) and Adjunct Switch Application Interface (ASAI), a call made by a CTI Application to a monitored station, that is busy on another call made using with the second CTI application, failed with the message "Out Of Service".	10502	
Occasionally, in a Communication Manager system with Computer Telephony Integration (CTI) configured and integrated with a CTI adjunct supported by AES (Avaya Aura Application Enablement Services), when a soft-phone in AES shared control mode re-registered, it caused the base set's TCP socket to close causing an active call to be dropped and the base set to unregister.	10522	
Outgoing Computer Telephony Integrated (CTI) calls that received a busy tone or were not answered did not get recorded by an Application Enablement Services (AES) integrated call recorder.	10530	
In a Call Center setup that includes a CMS (Call Management System) and is integrated with SIP adjuncts, such as IVR (Interactive Voice Response), or AAEP (Avaya Aura Enterprise Portal), under a very specific SIP messaging sequence, CMS showed inaccurate "hold" and "acd" durations for an Agent on an ACD (Automatic Call Distribution) call.	10532	
In a Call Center Elite System, under high traffic conditions, a large number of innocuous but unwanted Proc Errors were generated when agents resumed the calls that they had previously kept on hold.	10578	
The same port was occasionally assigned to multiple stations when transfers were performed with one-X Mobile/OPTIM stations.	10591	
In a call center configuration with multiple VDNs (Vector Directory Numbers), VDN1 and VDN2, when a call that termed to VDN1 and finally to an agent was transferred successfully to a second agent via VDN2, incorrect information was being	10621	

Problem	Keywords	Workaround
displayed on the second agent's display.		
The Voice mail greeting was terminated midway when calls that were established using 3PCC (Third Party Call Control) covered to a SIP integrated voice mail.	10677	
An unrecoverable corruption was encountered while entering a new line whose trunk group column overlapped with an existing line in the table on the CM SAT (System Access Terminal) "tandem-calling-party-number" (Tandem CPN) form.	10703	
Occasionally, under the following conditions: 1. Communication Manager with CTI (Computer Telephony Integration) configured and integrated with a CTI adjunct supported by Avaya AES (Application Enablement Server) 2. A third party Call recorder being used to record calls. 3. An incoming call over an IP Trunk Caller to an IP Agent. 4. Shuffling enabled on the system, when a third party device used Single Step Conference to join a call between an IP trunk caller and an IP Agent, it did not receive any audio. In such cases, call Recording was affected.	10726	
When SIP trunks were involved in calls, Communication Manager did not tandem unknown headers in 4xx/5xx Response Messages.	10750	
Under rare circumstances, when H.323 stations registered and unregistered consistently, the H.323 registration count audit could not be completed which resulted in inaccurate data being recorded.	10814	
In a Call Center system integrated with a CMS (Call Management System) which is used to administer Agent skills on the system, Multi-agent skill changes took many seconds to complete, increasing the chances of encountering contention errors when multiple administrators attempted simultaneous changes.	10815	
In a Call Center system which was configured with: 1. SIPCC Agents receiving calls through a VDN (Vector Directory Number) that plays an announcement 2. Multiple Communication Managers (CMs) or AAEP (Avaya Aura Enterprise Portal), the active VDN identity was not displayed on the SIP station that finally received a call that re-entered the CM system with a specific SIP message, which was generated during a transfer, and terminated to a VDN that played an announcement prior to queuing the call to the agent's skill or routing to the station.	10846	
Incorrect Voice Mail greeting was heard by the caller when a call over a SIP trunk covered to an Endpoint user who then transferred the call to a Voice Mail application, such as AAM (Avaya Aura Messaging) over a direct SIP trunk.	10870	
The CM-SAT (Communication Manager - System Access Terminal) command "list trace hunt-group" displayed incorrect information for calls attempting to terminate to the hunt-group agents when network region blockages were reported in the system.	10884	
With the "Multi-National Locations" feature and Shuffling enabled, an incoming call over a SIP trunk to a DCP agent did not have talkpath if the SIP trunk and the agent used different Location Parameters.	10948	
A caller did not receive caller information (SIP Call-Info header information) when making a conference call to Scopia or Avaya Aura Conferencing.	11001	
Owing to incorrect data in the history-info header, Avaya Communicator iPad dropped calls that involved SIP trunks.	11004	
Occasionally, for a brief period of time, when the SAT (System Access Terminal) command "status media-processor board" was executed for the duplicated TN2602	11049	

Problem	Keywords	Workaround
board, the "Standby Refreshed" field showed a blank value.		
An incorrect ANI was sent when One-X CES (Client Enablement Services) was used to make national calls, and the field "Expand ISDN Numbers to International for 1XCES" enabled on the "system-parameters features" SAT (System Access Terminal) form.	11060	
The team button on the monitoring station kept flashing even after the call was dropped. This occurred only for calls that covered to monitored stations with the following configuration on "system-parameters coverage-forwarding" CM-SAT (Communication Manager- System Access Terminal) form: 1) coverage criteria set to "All calls" 2) "Criteria for Logged Off/PSA/TTI" set to "y"	11071	
Under rare circumstances, Communication Manager experienced a segmentation fault during an H.323 trunk call when the field "H.323 Station Outgoing Direct Media" was enabled on the associated Signaling Group.	11117	
Incoming trunk calls to an Attendant became unresponsive when 1. The Attendant was in Night Service 2. "(SA8904) - Location Based Call Type Analysis" enabled on Communication Manager 3. Call routed to an external number via UDP (Uniform Dial Plan).	11119	
Occasionally, Communication Manager underwent a system reset under conditions of heavy SIP traffic.	11145	
Under extremely rare circumstances, when media gateways and H.323 or SIP stations were used, Communication Manager experienced resets.	11148	
For an incoming call over a SIP trunk terming on an extension that covers, or has "call forward no-answer" configured to another extension over a SIP trunk, when Communication Manager received the initial INVITE without the SDP, the call dropped once coverage or "call-forward no-answer" was initiated.	11149	
Forwarded calls between SIP endpoints caused the CM-SAT (Communication Manager- System Access Terminal) commands "Monitor Traffic Trunk" and "status trunk" to display conflicting information.	11238	
When "Extend-Call" was initiated by Avaya Communicator for Android/iOS to a cellular endpoint, the Call Appearance on the Avaya Communicator which was used to answer the incoming call continued to display the active call indication even though the call had been disconnected.	11251	
Occasionally, in a Communication Manager system that was integrated with an Avaya Aura Application Enablement Services (AES) application, such as Avaya DMCC (Device, Media and Call Control) H.323 Agents, network congestion caused incorrect lamp updates for ACD (Automatic Call Distribution) buttons on DMCC H.323 stations which resulted in dropped calls until the Hunt group queue was drained out.	11262	
Under rare circumstances, the creation and abandonment of SAT connections made via the TN799 CLAN board caused resource exhaustion leading to a system reset.	11284	
Under a very specific SIP messaging sequence arising out of tandem calls, Communication Manager underwent a system reset.	11288	
Occasionally, Communication Manager experienced a system restart if the TCP connections to H.323 endpoints fluctuated.	11306	
In a Call Center Elite system with "ACW Agents Considered Idle" set to "n" on the	11327	

Problem	Keywords	Workaround
"system-parameters features" CM SAT (System Access Terminal) form, when a non-SIP agent transferred an ACD (Automatic Call Distribution) call and then entered into the ACW (After Call Work) state, either "manual-in" or "pending ACW", the agent was incorrectly considered idle.		
Occasionally, in a Call Center setup with a VDN (Vector Directory number) and an associated Hunt group that has the field "ISDN/SIP Caller Display" field set to "mbr-name", Communication Manager experienced a segmentation fault when a call made to the VDN termed to an agent.	11332	
Unwanted denial event 2040 was logged on Communication Manager while registering H.323 stations in a Stub Network Region with TTS disabled.	11363	
When an attendant called over a SIP trunk and far end attempted to transfer the call over to the same Communication Manager where the attendant was registered, the display on the attendant was incorrect.	11383	
When the Communication Manager Hospitality feature was being used and a call landed on a called station that was busy on another call, the call log showed the caller extension number instead of the room number.	11394	
Under rare circumstances, the Communication Manager system experienced a restart.	11405	
Occasionally, in a CM (Communication Manager) system that was integrated with an Avaya Aura Application Enablement Services (AES) application, such as Avaya DMCC (Device, Media and Call Control) Client or recorder, call recording failed when an H.323 Endpoint answered a call using the bridged appearance button.	11430	
In a Call Center configuration where a Remote Worker is connected via SBC (Session Border Controller), calls between a Remote Worker and an Agent call did not get recorded.	11464	
With the "Multiple Locations" feature enabled, the "list ars route-chosen" SAT (System Access Terminal) command displayed incorrect output for outpulsed 7 digit numbers.	11563	
On rare occasions, after a system interchange on Duplex systems, the message "Translation Corruption" would be displayed on the CM SAT (Communication Manager System Access Terminal) when logging in, and the "save translations" SAT command would be blocked.	11574	
Under a specific SIP messaging sequence, one-way audio path was experienced for a call that used a SIP trunk configured between CM (Communication Manager) and CS1K when: 1) "Initial IP-IP Direct Media" was configured to "y" on the associated Signaling Group and 2) SIP messages sent by CM were compliant with RFC2833.	11592	
BSR (Best Services Routing) calls interflowing over SIP trunks without Initial Direct Media failed to play local treatment when administered to do so.	11641	
Occasionally, in a Communication Manager system that was integrated with an Avaya Aura Application Enablement Services (AES) application, such as Avaya DMCC (Device, Media and Call Control) Client or recorder, when the network connection between the CM and the recorder was lost, and the recorder re-registered with a new IP address and port, there would be no further recording of that call or any subsequent calls.	11658	
Occasionally, when the Session Border Controller was mis-configured, Communication Manager underwent a system reset when transmission of SIP messages were delayed because of underlying network issues.	11724	

Problem	Keywords	Workaround
In a Communication Manager system with greater than 500 DMCC (Device, Media and Call Control) endpoints, a few endpoints continued to be registered on the main server after the media-gateways correctly migrated to a survivable server, ESS (Enterprise Survivable Server) or LSP (Local Survivable Processor), when the SAT (System Access Terminal) command "disable nr-registration" was executed for the Network Regions to which the media-gateways belonged.	11734	
The Emergency Extension Forwarding feature did not work (disconnected calls were not returned to the agent) for agents with a Single Step Conference (SSC) monitoring/recording party on the call.	11763	
When a SIP endpoint A, used the bridged appearance of another SIP extension B, to answer a call made to B, intercept tone was heard by A if the field "Mask CPN/NAME for Internal Calls" was enabled on the CM-SAT (Communication Manager- System Access Terminal) COR (Class-of-Restriction) form.	11822	
The Emergency call, made over a SIP trunk, would fail to complete if the calling party's name contained a special character.	11838	
The "Timed Outgoing Trunk Call Disconnect" feature configured by administering the field "Outgoing Trunk Disconnect Timer" on the SAT (System Access Terminal) COR (Class or Restriction) form to a timer value, did not disconnect outgoing WATS trunk calls.	11840	
Under rare circumstances, calls made to AAAD (Avaya Aura Agent Desktop) over a SIP trunk would become unresponsive.	11864	
When the cable connecting the H.323 Endpoint to the network was unplugged, and the SAT (System Access terminal) command, "list registered-ip-stations" was executed, the endpoint continued to appear as registered.	11874	
When a specific SIP INVITE message was received over a SIP trunk while a call was in the vector processing state, reporting adjuncts, such as CMS (Call Management System) was unable to correctly report the call in the summary report for the skill or split involved.	11875	
When the SIP incoming INVITE message received by Communication Manager contained no m-line in the SDP, CM was unable to initiate new SIP trunk calls.	11877	
Loss of connectivity between Communication Manager and the far end of the SIP signaling channel that was being used during heavy SIP call traffic caused invocation of LAR (Look Ahead Routing) to be delayed which caused several calls to fail.	11912	
While transferring a call over a trunk, the attendant was prompted for an Authorization Code when it was not required to do so.	11915	
The "list trace station digits" and "list trace tac calling number" SAT (System Access Terminal) commands did not capture digit strings of variable length.	11919	
When an incoming SIP trunk call had "anonymous" as part of the URI, then this call could not be forwarded.	11950	
Occasionally, announcements on TN2501 VAL boards were played for a very short duration under the presence of heavy load on the announcement boards.	11992	
A One-X CES (Client Enablement Services) station did not display the calling party's extension if the call arrived from an attendant.	12013	
Under rare circumstances, when an EC500 device answered a call, ringback was heard by both parties instead of talkpath being established.	12036	
Under a very specific SIP messaging sequence, for a call over a non-OPTIM SIP trunk, the caller was unable to hear the other end.	12068	

Problem	Keywords	Workaround
Using the "cpn-blk" feature button on a One-X Communicator to invoke a call to an extension that could not be routed, resulted in the call terminating on the calling station.	12124	
After network outage, a few time-to-service H.323 stations remained unresponsive.	12127	
Incoming calls that routed through a Vector Directory Number (VDN) route-to step did not display Caller ID (CID) numbers.	12141	
Under a very specific SIP messaging sequence, long-duration FAX calls over a SIP trunk failed.	12231	
In a Call Center setup integrated with an Avaya Aura Application Enablement Services (AES) application, such as Avaya DMCC (Device, Media and Call Control), a 3PCC (Third Party Call Control) call by an agent in telecommuter mode failed when the call was generated within 5 seconds of the agent login.	12262	
Announcements could not be recorded on Media-Gateways or TN2501 VAL Boards owing to busy channels.	12265	
In a Call Center environment, under a very specific SIP messaging sequence, reporting adjuncts, such as CMS or IQ, were unable to accurately track the call if SIP connected adjuncts, such as Avaya Aura Enterprise Portal redirected the calls over a trunk to a different system.	12268	
In a Call Center Elite environment, when a skill was added to an agent who was active on a call and after the call entered into a Timed ACW state, the agent could receive a call on the new skill while still in ACW. Occasionally, this would cause CMS (Call Management System) to reset because data related to the agent's new skill was not updated on the CMS in time.	12300	
Frequent internal process resets would result in a WARM restart of the system.	12337	
Under rare circumstances, when Shuffling is enabled on the Signaling Group used for an incoming call that is eventually answered over a service link, there would either be no talkpath on the established call or the call would be abruptly dropped.	12530	
Under rare circumstances, Communication Manager would undergo a reset when using H.323 stations or H.323 trunks.	12565	
In a Call Center Elite system integrated with an Avaya AES (Application Enablement Server) Application along with Reporting Adjunct, such as CMS (Call Management Server), 3PCC (Third Party Call Control) calls would generate an incorrect sequence of messages causing the CMS link to disconnect.	12701	
In a Call Center environment that includes CMS (Call Management System), the CMS lost track of a call after an incoming call to a VDN (Vector Directory Number) went through a vector that immediately tandemed out over a trunk that failed the look-ahead and then tried a different trunk. This issue was reported when the vector's first step was not set to 'wait 0 hearing ringback', as is typically recommended.	12733	
In a Call Center environment, when the workmode of an agent is changed from AUX to Auto-in using a FAC, the active call would be dropped.	12875	

Problems fixed in Communication Manager 6.3.115.1

Table 29: Fixes delivered to Communication Manager 6.3.115.1

Problem	Keywords	Workaround
When an agent uses a feature access code to move from ACW (After Call Work) to Auto-in (Available), the agent did not receive any calls, even though there were calls in queue.	13150	Instead of Feature Access Code (FAC) use "auto-in" feature button.
Announcements stopped playing.	13677	

Problems fixed in Communication Manager 6.3.116.0

Table 30: Fixes delivered to Communication Manager 6.3.116.0

Problem	Keywords	Workaround
In a call between SIP station on Communication Manager and CS1K, SIP station was not connected to announcement played by CS1K when CS1K ACD had all agents busy.	5809	
"t.38-G.711-fallback" fax call failed when the order of codecs G.711A and G.711MU on the remote end was different from what was configured on Communication Manager.	6438	
An incoming call over SIP trunk to a Vector Directory Number vector that attempted to play an announcement on AMS (Avaya Media Server), either failed to play the announcement or experienced a delay. This issue occurred with the following configuration: 1. Receiving side of the SIP signaling group had "Initial IP-IP Direct Media" disabled and "Direct IP-IP Audio connections" enabled. 2. The first step of vector was "wait 0 hearing ringback" with the next step set to play announcement. 3. The announcement was in a different IP-network region from that of the SIP trunk.	6896	
Incorrect calling party information was logged in the Call Detail Record (CDR), when a station dialed into the Voice Mail.	10991	
The service observer observed a loss in voice path if service observing station supported only codec Set G.711 and the IP-codec-set used for the call did not support G.711. This happened only when Agent and Service Observing station belonged to different Network regions.	11054	
Transferring firmware to the IPSI (IP Server Interface) boards on CM (Communication Manager) failed if the firmware was being fetched from a file server that expected a password and the loadipsi command was used with the "-w" option	11489	
On CM (Communication Manager), enabling ASAI (Adjust Switch Application Interface) message tracing in the MST (Message sequence tracer) caused warm resets.	11591	
Occasionally, calls did not complete when H.323 trunks were involved.	11831	
The IP-DECT (Digital Enhanced Cordless Telecommunications) station incorrectly displayed the name of the trunk-group instead of the number of the caller. This occurred when IP-DECT station was configured in a call-pickup group and Incoming call was made over a trunk with "Send Name" disabled and "Send Calling Number" enabled on the "change trunk-group" form on SAT (System Access Terminal).	11897	
Secondary dialtone was not provided if an R2-MFC trunk was accessed via TAC dialing.	11949	
Under rare circumstances, administering or modifying button labels on endpoints caused Communication Manager to reset.	12067	
CM (Communication Manager) sent "sdp-anat" in "Supported" headers in outgoing SIP messages when the field "SIP ANAT Supported" was set to 'n' on the SAT (System Access Terminal) Trunk Group form.	12085	
When a call covered to a station that was active on another call, the room number of the active call was over-written by the room number of the new incoming call. This resulted in the same room number being displayed on both call appearances.	12116	

Problem	Keywords	Workaround
This occurred for calls that were covered to a coverage answer group on Communication Manager configured with Hospitality feature.		
Answer back of a chime call resulted in one-way talkpath when the answering user was on a different Port Network than that of the chime port.	12201	
The calling SIP station showed "UNKNOWN NAME" on display for calls placed to a virtual extension which then covered to a remote coverage point.	12251	
There was no talk-path on an incoming call coming from a Cisco device after the call covered to AAM (Avaya Aura Messaging) voice mail and Communication Manager tried to shuffle the call to direct IP.	12323	
In a Call Center Elite system that was integrated with a Reporting Application, such as CMS (Call Management System), while making changes relating to skill/vdn/vector the CMS received the error "Cannot Perform the requested Operation. Administration contention. Try again later."	12341	
CM (Communication Manager) sent "sdp-anat" in "Supported" headers in outgoing SIP messages when the field "SIP ANAT Supported" was set to 'n' on the SAT (System Access Terminal) Trunk Group form.	12422	
Avaya Client Enablement Services based callback Call failed when "Incoming Call Handling Treatment" was configured for the PSTN trunk involved in the call.	12431	
One-X CES (Customer Enablement Services) mobile application had incorrect call log, when it dialed to another internal station over trunk. This occurred under the following configurations - 1. ARS FAC (Feature Access Code) was added to the called number through "inc-call-handling-trmt trunk-group" or through Trunk Group form on SAT (System Access Terminal). 2. Incoming call, over SIP/H.323 trunk, routed to an internal station via ARS digit conversion. 3. Multiple Locations were configured on CM.	12433	
When a virtual station used a coverage path that covered to a Voice Mail and had the coverage criteria "All" enabled on the "coverage path" SAT (System Access Terminal) form, then when an incoming trunk call to an attendant was transferred to the said virtual station, the generic greeting was heard on the voicemail server.	12450	
For incoming trunk calls, missed call log entries were not logged on SIP stations if the call-forward feature was activated on the station.	12482	
Occasionally, H.323 Agents were not able to make or receive calls.	12490	
The top line on a 96x1 deskphone did not display caller information for the call pickup feature when "Enhanced Call Pickup Alerting" and "Enhanced Redirection Notification (ERN)" features were enabled.	12505	
Occasionally, when an H.323 hardphone, that was registered in ANNEXH mode, moved from "named" to "unnamed" then this phone would be logged out by the Communication Manager.	12540	
Under a very specific SIP messaging sequence, one way talk path was encountered on a call that used an external SIP trunk. This call would eventually be dropped.	12564	
Occasionally, there was no output when the "list measurement announcement yesterday" SAT (System Access Terminal) command was executed on a system that was administered with announcements with recordings that generated statistics over days while the announcements were being administered.	12566	
For a SIP station with Call-forward activated, the call-forward destination number	12575	

Problem	Keywords	Workaround
got stored as the Last Dialed Number (LDN) instead of the actual last called number. Therefore, invocation of LDN feature, caused the call forward destination number to be dialed.		
A denial event was not logged causing it to be absent from the "list trace station" SAT (System Access Terminal) command output when a traced station to station call was put on hold, and the held station had Data Restriction and Music on Hold enabled.	12625	
CM (Communication Manager) running SIP traffic occasionally experienced system resets leading to a service outage.	12634	
No announcement was played to a telecommuter agent that was registered on AES (Avaya Enablement Service) shared control phone that was configured to "auto-answer" mode.	12636	
Execution of "status ip-network-region" SAT (System Access Terminal) command resulted in an incorrect output.	12644	
Occasionally, in a Call Center Elite system that was integrated with a Reporting Application, such as CMS (Call Management System), the timestamp in messages that went into the reporting application, appeared to move backwards for some message sequences.	12675	
The displayed information on bridge appearance administered on DCP station, as a result of an incoming call transferred to the principal station, did not clear even after the call between principal and caller was dropped. This occurred only when "Display Information with Bridged Call" was disabled on the "change system-parameters features" SAT (System Access Terminal) form.	12710	
Occasionally Communication Manager underwent a System Reset.	12729	
For an incoming ISDN call on a SIP station, the station displayed "Anonymous" in the call log instead of "info restricted" or no number, if the incoming ISDN call had restriction in the presentation and had no name and no number in the SETUP message.	12732	
The "list trace station" and "list trace tac" SAT (System Access Terminal) commands have been enhanced to display whether private or public numbering is in use.	12734	
Occasionally, a SIP call over a Non Optim SIP trunk would eventually be dropped if the far end SIP trunk sent a 200OK response with inactive SDP (Session Description Protocol) to CM (Communication Manager).	12749	
During administration of multiple announcements with audio-groups, the announcements could not be added, changed, or removed because of translation corruption.	12764	
No call log was registered on DCP station for any outgoing calls made over SIP trunk via ARS.	12806	
When CM (Communication Manager) SIP endpoints that were being monitored by an application on the AES (Application enablement services) went offhook, the "Call Initiated" event was not sent by CM to the AES application because of which Avaya Call Recorder (ACR) did not record the call.	12831	
When the field "Agent/Caller Disconnect Tones" was enabled on the "system-parameters features" SAT (System Access Terminal) form, and the agents attending SIP trunk calls were being monitored by ASAI (Adjunct Switch Application Interface) application and service observed, the ASAI application was unable to disconnect the agent's call.	12847	
When Communication Manager (CM) was configured with CTI (Computer	12854	

Problem	Keywords	Workaround
telephony Integration) interface and ACD (Automatic Call distribution), calls that involved CTI transfer occasionally experienced talkpath disturbances if the agents were using SIP softphones and shuffling was enabled on SIP trunks.		
The calling phone continued to see the called destination identity even when the far end set the Privacy ID header in the response message.	12888	
Administration of a large number of “busy-indicator” buttons failed even when there was sufficient capacity on the system to accommodate such administration. Administration of a large number of “busy-indicator” buttons failed even when there was sufficient capacity on the system to accommodate such administration. Administration of a large number of “busy-indicator” buttons failed even when there was sufficient capacity on the system to accommodate such Administration of a large number of “busy-indicator” buttons failed even when there was sufficient capacity on the system to accommodate such administration.	12892	
Occasionally, in an environment where a Call Center System was configured with reporting adjuncts, such as CMS (Call Management System) or IQ/APC, and integrated with SIP connected adjuncts, such as AAEP (Avaya Aura Enterprise Portal), for a scenario that involves ICR (Avaya Intelligent Customer Routing), a trunk was idled prematurely, thus preventing the CMS from further tracking the call.	12912	
CM (Communication Manger) configured with Port networks with DS1 boards (TN767, TN 2464, etc.) underwent warm restarts caused by the DS1 board continuously generating and clearing a "yellow" alarm.	12972	
The 'list bcms trunk' SAT (System Access Terminal) command for a trunk group that was administered with 255 members, displayed a blank for the 'Number of Trunks' field on the form displayed.	13010	
When a SIP station initiated a call with the Auto Callback feature enabled, the display on the calling station did not get cleared if the called station was busy on another call.	13037	
When the field "IP Softphone" on the "station" SAT (System Access Terminal) form was configured as 'y', then while executing the "display capacity" CM SAT command, the values for the fields "Limit" and "Available" for "Administered IP Softphones", on page 8, and "Softphone Enabled on Station Form", on page 11, were blank.	13095	
When an incoming trunk call was transferred to another station which further forwarded the call, CDR (Call Detail Recording) showed incorrect number as the called party.	13096	
The ACB (Auto Call Back) softkey on the calling station continued to be displayed on the station screen, instead of being cleared, after the call was answered by a member of the pickup group that included the called party.	13122	
When Special Application "(SA8481) - Replace Calling Party Number with ASAI ANI?" was enabled on the "system-parameters special-applications" SAT (System Access Terminal) form, Device, Media and Call Control (DMCC) stations/ applications used the Calling Line IDentification (CLID) of the DMCC station/application instead of the CLID of the desk phone.	13126	
Occasionally, for calls involving SIP trunks, under a very specific SIP messaging sequence that cause the SIP UPDATE message to loop, Communication Manager experienced segmentation faults or warm restarts.	13132	
In a Call Center Elite environment, when the "Auto-In" FAC (Feature Access Code) was used to change the work-mode of an agent from "After Call Work" to "Auto-In" (available), the agent did not receive any calls when there were calls in queue.	13150	
In a Call Center environment, with "SA8569 - No service observing tone heard by	13174	

Problem	Keywords	Workaround
agent" enabled on Communication Manager, sometimes agent heard the service observing tones.		
Occasionally, IP Agent registrations logged the same denial event multiple times in the trace.	13307	
A call to VDN (Vector Directory Number) dropped unexpectedly after the execution of route-to "~r" step in the associated vector when the field 'Network Call Redirection' was enabled on the SIP trunk that was being used.	13327	
When a call received as a result of ACB (Auto call back) was not answered, call appearance displays on calling as well as called stations did not clear.	13393	
Ocassionally, CM (Communication Manager) agents that were controlled by an AES (Application Enablement Services) application and were being service observed experienced talkpath disturbances if the agents were One-X H.323 softphones configured as "other phone" (on the softphone) and the other phone was being accessed by a SIP trunk on which shuffling was enabled.	13419	
An Incoming trunk call that was answered, unexpectedly dropped after 30 seconds under the following conditions: 1. "(SA8965) - SIP Shuffling with SDP" was disabled on "system-parameters special-applications" CM SAT (Communication Manager - System Access Terminal) form. 2. The field 'Shuffling with SDP' was enabled on the trunk group that is being used.	13446	
CM (Communication Manger) agents trying to conference two calls experienced call drops when the agents were One-X H.323 softphones configured as "other phone" (on the softphone) and the other phone was being accessed by a SIP trunk on which shuffling was enabled.	13455	
Occasionally, H.323 Agents unexpectedly unregistered from the Communication Manager.	13470	
The "list trace" SAT (System Access Terminal) command was enhanced to display when CM overrides existing routing and sends a SIP INVITE message to Avaya Aura System Manager.	13491	
The Time Of Day Routing feature, when used with the Special Application "(SA9050) Increased TOD Routing Tables/Partition Grp Num" failed to route the call correctly after the system underwent a reboot or an upgrade and required the Time of Day Routing Plan to be edited in order to be effective.	13531	
In a Call Center Elite environment, incoming ACD (Automatic Call Distribution) calls routed via a VDN (Vector Directory Number) were dropped upon receipt of a 380 Alternate Service message from a station.	13535	
In a Call Center System configured with reporting adjuncts, such as CMS (Call Management System) or IQ/APC and integrated with an SIP connected adjuncts, such as AAEP (Avaya Aura Enterprise Portal), when an agent, after conferencing a caller over a SIP trunk that resulted in the call being queued to a hunt group or an Agent (Direct Agent Call), dropped out of the call and changed its mode to "After Call Work" , the CMS stopped tracking the call. Eventually, the link between CMS and CM was reset.	13536	
A station with "Type" administered as "9608" on the station form, failed to register and the denial event "1927 IP RRJ-Invld station type" was generated.	13565	
A call made to an extension that had EC500 (Extension to Cellular) enabled, dropped unexpectedly when the EC500 SIP trunk received a 603 (Decline) SIP Response via the Session Manager.	13566	

Problem	Keywords	Workaround
Occasionally, when a minimum of 15 agents were logged in, each configured with 40 skills or more, a Forced Agent Logout either by the configured Location Access Code or Skill Access Code, caused the system to reset.	13604	
In a Call Center environment integrated with a CMS (Call Management System) using SIP trunks and an adjunct such as Experience Portal redirecting calls between multiple CM (Communication Manager) systems, the CMS stopped tracking a call that routed out through the SM (Session Manager) to the Experience Portal that performed a consultative transfer. The error message, "ERROR illegal transfer/conference", was logged in the CMS.	13655	
CM (Communication Manager) sometimes failed to respond to other products connecting to CM through SIP trunks.	13665	
Occasionally, an unattended (blind) transfer involving SIP stations caused the Communication Manager to restart.	13676	
For an outgoing call that required an Authorization Code, an AAAD (Avaya Aura Agent Desktop) client was unable to enter the authorization code causing the call to fail.	13689	
In a configuration with 9611SIPCC deskphones/agents, DTMF (Dual Tone Multi frequency) in a call involving a 9611SIPCC was not translated to a tone during an established call causing applications requiring DTMF tones to fail.	13690	
When SIP endpoints whose calls were being recorded by ACR (Avaya Call Recorder) and were being monitored by the CTI (Computer Telephony Integration) applications transferred a call to another SIP application, the ACR did not stop recording the call for the transferring SIP endpoint even after the transfer was complete.	13706	
Calls to the Voice mail system, over a direct SIP trunk, dropped unexpectedly, making it impossible to retrieve voice mail messages.	13785	
The called H.323 station display did not show "Anonymous" for a call originated by a SIP extension when 1. "Mask CPN/NAME for Internal Calls" was set to 'y' on the COR (Class of Restriction) form and 2. "Per Station CPN - Send Calling Number" was configured as 'n' for	13788	
The "list usage ip-address" SAT (System Access Terminal) command failed to display IP addresses of the media-gateway configured in the system.	13832	
Occasionally, when the "Calling Party Number Conversion For Tandem Calls" SAT (System Access Terminal) form was administered, the "list usage trunk" command would render the session unresponsive without printing any data on the screen.	13835	
In one instance (very rare circumstances), when calls over a SIP trunk failed to complete a warm interchange occurred.	13912	
In a Call Center environment with the "Multiple Call Handling" feature enabled on skills and splits, the agents were disallowed to handle multiple calls as was expected.	13924	
In one instance (very rare circumstances), after falling back from Enterprise Survivable Servers (ESS) to the primary server pair, H.323 IP trunks became stuck in a "pending-busyout" state.	13951	
In a configuration where a CM (Communication Manager) was integrated with Avaya AES (Application Enablement Server) and integrated with a CTI (Computer Telephony Integration) adjunct that conferenced in an SSC (Single-Step Conference) party, in invisible mode, onto an existing call, then when the calling party pressed the "drop" button, the called party was dropped from the call.	13968	

Problem	Keywords	Workaround
However, the call remained active between the SSC party and the calling party. In addition, when an agent was dropped from such a call, the agent did not move into Timed ACW (After Call Work) mode as expected.		
Occasionally, calls over Tandem H.323 trunks, that had encryption enabled, used more media resources than were assigned. This was displayed in usage reports.	14022	
The field "Override ip-codec-set for SIP direct-media connections" on the "System-parameters ip-options" SAT (System Access Terminal), when set to "y", failed to override media from being encrypted on a call, as it was configured to do.	14023	
When administering an announcement that used an audio-group as the announcement source, the message "Error Encountered, Can't Complete Request" was displayed on the screen when an AAMS (Avaya Aura Media Server) was configured as a location for the audio-group.	14080	
In a Call Center configuration utilizing ACD (Automatic Call Distribution), agents were not available to receive calls after transferring an ACD call.	14089	
Blind transfer failed when SEMT (SIP Endpoint Managed Transfer) was enabled on Communication Manager and Reliable Provisional Responses (100rel header option) was supported or required.	14110	
DMCC (Device, Media and Call Control) shared control H.323 station could not record the registered SIP station due to a missing Facility message from Communication Manager (CM) if the administered set type of SIP station was 96x1SIPCC.	14125	
The Called Party displayed the CPN (Calling Party Number) in an incorrect format when it received a call from an agent that belonged to another location.	14151	
In a Call Center environment that included a CMS (Call Management System), when a One-X Communicator or a non-Avaya SIP phone transferred a measured call by sending CM (Communication Manager) a blind REFER, the CMS reports incorrectly showed this call to be in queue along with other inaccuracies.	14176	
In a configuration in which the CM (Communication Manager) was integerated with a CTI (Computer Telephony Interface) application, when endpoints that were being monitored by the CTI application transferred or conferenced calls using SST (Single Step Transfer) or SSC (Single Step Conference) features, the transfer or conference actions failed.	14233	
A call answered by an EC500 (Extension to cellular) extension experienced loss in talkpath when: 1. The called SIP station used Direct Media (DM) 2. The call was extended over a trunk that did not support DM 3. The call involved multiple media resources.	14257	
Occasionally, in a stub and core network topology the network bandwidth for any one or more network regions would be exhausted.	14260	
DTMF tones were not received by the far-end when "Out-of-band" DTMF mode was in use.	14262	
The "list ars route-chosen" SAT (System Access Terminal) command, when executed for a digit string not defined in the AAR/ARS tables, showed incorrect location information.	14265	
Occasionally, under a very specific SIP messaging sequence that involved Codec Set manipulation, CM (Communication Manager) experienced a restart.	14322	
Under rare circumstances, the system would undergo a reset when calls were made from H.323 endpoints to SIP endpoints.	14335	

Problem	Keywords	Workaround
In one instance (very rare circumstances), execution of the "list trace tac" SAT (System Access Terminal) command with the "c" option caused a warm interchange.	14388	
A trunk call between two CMs (Communication Managers) failed when attempted to be answered using a bridge-appearance, when 1. The field "Allow Bridge DM answer" was enabled on the CMs 2. Shuffling was enabled on the CMs 3. "SRTP" along with "none" (no encryption) was used as an option on the ip-codec-set on one CM.	14436	
In one instance (very rare circumstances), in CM 6.3.9.0 and later Service Packs/Releases, a configuration with SIPCC deskphones/agents experienced a reset.	14747	
CM (Communication Manager) sent two call initiation events to the CTI (Computer Telephony Integration) applications if a SIP endpoint originated the call because of which Avaya Call Recorder (ACR) did not record calls.	14966	
Occasionally, segmentation faults on Communication Manager led to server interchange when an AES (Application Enablement Service) shared control phone was involved in a call.	14998	
Announcement boards required to be busied-out prior to adding or changing integrated announcements or audio-groups that used these announcement boards.	15001	
In a Call Center environment where the "Timed After Call Work (TACW)" feature was administered and being used for SIP agents, the agents failed to move into TACW mode after transferring a call that had not yet been answered. Eventually, this resulted in the agent being available but unable to take further calls.	15158	
Occasionally, when the field "Initial INVITE with SDP for secure calls" was enabled on the "system-parameters features" SAT (System Access Terminal) form, on calls that involved SRTP, several talkpath issues were encountered.	15303	
Occasionally, in a Communication Manager configuration with either two MGs (Media Gateways) or one MG and one PN (Port Network), exhaustion of resources caused new calls to fail.	15413	

Problems fixed in Communication Manager 6.3.117.0

Table 31: Fixes delivered to Communication Manager 6.3.117.0

Problem	Keywords	Workaround
No talk-path was observed for an unattended transferred call over SIP trunk while SIP Direct Media was enabled.	3578	
There was one way talk path and subsequent call drop after 11xx/12xx SIP station did a 3-party ad-hoc conference.	4491	
Adjunct/Switch Application Interface (ASAI) initiated calls via hunt groups to agents were recorded as "Incoming Call" instead of "Adjunct Call" in CDR (Call Detail Record).	5493	
CMS ignored a call under the following condition; a call to an agent over a SIP trunk is transferred back to the same VDN via a SIP trunk, the agent completes the transfer when music is playing in the vector, and the call is again delivered to the same agent.	8498	
SIP principal phone was dropped automatically from a bridged-on call that was already answered by its bridge appearance when "Enforce sips URI for SRTP" was enabled on the "change signaling group form" on System Access Terminal (SAT).	9043	
When the TCM command "extu <ext>" was executed for a SIP extension the network region erroneously changed to NR0 after a call was made to this extension.	9570	
Sometimes, if shuffling was enabled for SIP trunk, no talk-path or call drop issue might be observed.	12423	
In a configurations with variable numbering plans and optimised digit timeouts (by adding many repeated ARS Analysis entries that differ only in the min/max column), administrator saw "error encountered, cannot complete request" or "entry is bad" on SAT screen while configuring the ARS Analysis form.	12632	
When an attendant user transferred the external call to a SIP station, caller number displayed incorrectly on SIP stations. When an attendant group transferred the internal call to a SIP station, SIP station displayed attendant name instead of "OPERATOR".	13087	
"Transfer-to" feature activation used incorrect calling party information when it was activated from a covering station or a bridge appearance of a station that received a call from a SIP trunk and was transferring a call to a SIP trunk to reach voicemail.	13537	
Shared Control NICE recording station failed to record the full call.	13666	
When an endpoint controlled via Application Enablement Services (AES) made a call to a station over SIP trunk and routed the call to an agent via hunt-group, the agent display showed it's own number instead of the caller's number.	14121	
When the X-ported endpoint was removed from CM, the application on the AES did not get a notification to stop monitoring the endpoint.	14285	
List trace station on the SAT failed to provide agent state and list activity for an agent extension	14403	
Appropriate denial events were not logged when service observing failed with Data Restriction set to 'y' on a call object, e.g., trunk-group, etc.	14416	
If the dialed number had # followed by + over an H.323 trunk having "overlap/ overlap" configured on the "Digit handling (in/out) field on the "change trunk group form", the system underwent a restart.	14531	
Call to the VDN (Vector Directory Number) failed on Communication Manager as Feature Server. The problem occurred with the minimum configuration:	14572	

Problem	Keywords	Workaround
1. IMS Enabled on the "signaling group" form. 2. VDN number not entered in the public/ private numbering tables.		
Call logging for intercom calls failed on DCP phones.	14581	
An incoming call was not recorded in the call logs if the called number began with a pound/hash sign (#).	14588	
When a station that transferred an incoming call over a SIP trunk (unattended transfer) to another station, logged-off before the "Transfer recall" triggered, the call got stuck during transfer recall when "Music (or Silence) on Transferred trunk calls" on "change system-parameters features" form was set to 'yes'.	14608	
Caller dials a VDN and reaches an agent in a skill X by means of a 'converse-on skill X' step in a vector. This skill has Timed After Work Call (TACW) configured. If the Caller hangs up before the 'Agent' the Agent does not go into 'Timed After Call Work'. This is incorrect behaviour. Note: If the Agent hangs up first they do go into TACW [Correct behaviour].	14705	
ISDN trunks were in an Out-of-service-NE state after a connection preserving upgrade or patch activation was performed. At the time of Release Notes publishing this issue had occurred only once.	14731	Perform a system reset to recover the IDSN trunks.
CM dropped calls while executing the 'adjunct route' vector step. The problem occurred with the following minimum configuration: Communication Manager (CM) configured with Call vectoring and H.248 media gateways, Application Enablement Services (AES) server, Computer Telephony Integration (CTI) applications.	14771	
When an incoming call was covered after being forwarded, the first coverage point on the coverage path was ignored.	14816	
When stations were monitored by a CTI application and an incoming call was transferred to another station which was being service observed, disconnect event was not sent to CTI application causing recording of call to continue for 9 hours.	14822	
Under a very specific SIP messaging sequence involving SIP UPDATE message, call did not tear down properly.	14860	
Call was routed out to an incorrect extension if the dialed number contains a routable extension and "Location-Based Call Type Analysis?" is set to "y" on "change dial plan parameters" form.	14870	
Periodically, an Attendant consoles went out of service.	14918	
The Calling SIP station displayed the trunk group name instead of the called party number while the outgoing call was ringing.	14939	
Outbound calls over ISDN trunks initiated by Customer Interaction Express (CIE) clients were stuck in an unanswered state when the calls that were transferred to agents on CM were answered.	15042	
The extension numbers did not get recorded in the call log when H.323 IP phones originated a directory call.	15056	
When the attendant transferred an incoming trunk call to a virtual station whose coverage was set to 'All', the caller over the trunk received a generic greeting from the SIP Modular Messaging.	15093	
When an H.323 station made an autodial pause call (~p), the last digit was truncated on the outgoing display.	15121	
Team button reroute notification was wrongly activated when the call-fwd destination was changed from monitoring station to external user.	15157	

Problem	Keywords	Workaround
Occasionally, after an incoming ISDN trunk call was answered on the SIP station, if this SIP station or its MDA (Multiple Device Access) device tried to re-subscribe the dialog state event, CM (Communication manager) omitted sending the national/international Call party number prefix.	15178	
CM incorrectly sent a user not responding message to Application Enablement Services (AES) applications for SIP trunk calls which failed to reach the user because of network problems.	15188	
The customer was able to enter and submit a value out of the permissible range(1-2000) for the "change route-pattern" form on the System Access Terminal(SAT).	15189	
Occasionally, during administration of an audio-group, SAT would be locked and translations would fail to complete	15199	
When multiple location short dialing is configured and a local station having short dial code same as initial dialing digits of EC500 destination, EC500 call termed to a local station when the call met with a glare on first attempt.	15242	
TCM variable NumSipRingingCalls on Avaya Communication Manager was showing count as number of transactions instead of number of SIP calls in ringing state	15319	
On shuffled (Direct IP-IP) calls involving an H.323 IP AnnexLP station/phone, the DTMF tone for the first digit dialed from the station after the call was established was longer than the DTMF tone provided for subsequent dialed digits.	15388	
One-x Attendant couldn't register to Communication manager with Pin-Eke security profile.	15403	
When an ongoing call was redirected to a gateway for joining a conference, the call was dropped during a very specific capabality negotiation SIP signalling between CS1K and Gateway via CM.	15435	
Remote users/workers utilizing Telecommuter mode would have their trunk ports incorrectly displayed as "in-service active" with no connected ports.	15436	
Call was dropped by MSUM (Microsoft UM voicemail) with a 403 "Forbidden" due to invalid History-Info header in INVITE coming from Avaya Communication Manager (ACM).	15469	
Occasionally, a One-X Agent in telecommuter mode was unable to answer the incoming call.	15490	
During a network outage between CM and CLAN boards, the 'status socket-usage' command on CM administration terminal 'sat' displayed incorrect values.	15521	
ACM (Avaya Communication Manager) experienced a restart owing to a memory leak situation.	15581	
CMS ignored a call under the following conditions: agent receives a call, puts the call on hold, makes another call, while the second call is queued, a service observer joins the call. The agent puts the second call on hold, returns to the first call, then conferences in the second call.	15619	
A warm interchange occurred on one occasion in a duplicated server pair where SIP stations with bridged appearances were being used.	15674	
The "Phone Number" field on the "off-pbx-telephone station-mapping" form would revert back to it's original value if changed using SMGR or ASA when the newly entered extension matched a previously administered extension for another station. This occurred only when a new SIP station was added using System Manager(SMGR)/Avaya Site Administration(ASA).	15676	
The SAT "Terminal Parameters" form, page 2 included DCP set types that do not support download of terminal parameters. Those were 9404, 9408, 1408 and 1416	15682	

Problem	Keywords	Workaround
set types.		
The call made over the H.323 trunk was dropped. The problem occurred with the following minimum configuration: 1) Communication Manager(CM 7.0.0.0.0 above) system with Avaya Media Server(AMS) connected to a CM(6.3.0.0.0 or above) system over a H.323 trunk. 2) Codec used on both CMs is G.726 administered on "change ip-codec-set" form.	15735	
Team-button did not ring for an incoming call on QSIG trunk forwarded to the monitored station which was unregsitered at that time	15746	
The display capacity did not show the correct "Logged-In IP Softphone Agents" and lot of 7171 10132 proc errors were seen in the logs.	15957	
Intermittently, no talkpath at one of the IP endpoints in the conference across SIP trunks. The problem occurred with the following minimum configuration: Field 'Prefer use of G.711 by IP Endpoints Listening to Music' and 'Prefer use of G.711 by IP Endpoints Listening to Announcements' set to 'y' on system-parameters ip-options form on ACM (Avaya Communication Manager) Multiple network regions Multilple gateways Conference.	15960	
The standby server from the duplex ESS server pair went into a reboot once a day when configuration files were pushed from the main server to the ESS servers.	15963	
Conference display on SIP phones was not consistent when any one of the Latin-script based unicode language was Used	16015	
MEMPOOL errors were observed on Communication Manager when "Multiple Level Precedence & Preemption?" field on" system-parameters customer-options" form was set to "Yes".	16016	
When a user with SIP extension logged in to Equinox with his/her EC500 feature enabled and connectivity to Equinox was lost, user's own extension number was displayed on his/her EC500 device upon receiving a call on EC500 device	16034	
CMS may stop tracking a call when an agent used a hard conference button to complete a conference after only dialing the AAR or ARS code and insufficient digits then completed the remaining digits after the conference.	16047	
Segmentation fault when trying to clear a list of server alarms and comma seperated list as follows. almclear -n 6, 5	16050	
Avaya Communication Manager did not tandem SIP response "480 SIPS not allowed when LAR (Look Ahead Routing) was used with SIP DM (Direct Media). This could result in call drop, instead of retry.	16052	
When an incoming call for Avaya One-X Agent was answered by telecommuter, the call could not be answered on the Avaya One-X Agent Client. The problem occurred with the following minimum configuration: Avaya one-X Agent with telecommuter mode enabled. Auto-answer enabled on "change station" form on System Access Terminal (SAT).	16332	
CM incorrectly sent a user not responding message to Computer Telephony Integration (CTI) applications for SIP trunk calls which failed to reach the user because of network problems.	16516	
Customer is blocked from adding 1000th announcement or more when doing "add announcement".	16861	

Problems fixed in Communication Manager 6.3.118.0

Table 32: Fixes delivered to Communication Manager 6.3.118.0

ID	Minimum Conditions	Visible Symptoms	Found In Release
CM-3296	SIP endpoints	No notification about simulated bridge appearance being maintained.	6.3.9.0
CM-11028	Send All Calls(SAC), Call-forwarding activated	The call went into a loop between two stations.	6.3.8.0
CM-11120	SIP phone on a communication Manager	Occasionally, a call appearance on the SIP phone stayed stuck with bridging icon even when there were no active calls.	6.3.9.1
CM-13156	A CM configuration with port networks	Occasionally it is possible to exhaust all timeslots in a port network. All calls involving that port network will fail.	6.2.7.0
CM-14213	CM fetching license from the WebLM whose response is delayed.	Temporarily statuslicense shows license timeout error. CM generates a major LIC-ERR alarm	6.3.14.0
CM-15501	Call Center using Timed After Call Work and the option "After Xfer or Held Call Drops" set to 'n'.	The option for Timed ACW after transferred call dropped worked differently depending on which direction the ACD call was transferred and whether the agent was on a SIP station or a non-SIP station.	6.3.16.0
CM-15629	Voice Mail greeting	Transfer to sip integrated Voice Mail (VM) behaves differently with number format issues causing the calling party to hear the wrong greeting.	6.3.14.0
CM-15745	R2MFC trunk between CM, AES-CTI-application	No Called Party Number was sent to cti-application if call went over R2MFC trunk	7.0.1.2.0
CM-15978	Administered with Agent/Caller Disconnect Tones and Music on Transferred trunk calls	A hung call could occur on a system with music on transferred trunk calls, when an incoming SIP ACD call answered by an agent is transferred to a SIP station that does not answer and has no coverage path	6.3.14.0
CM-16180	Voice Mail routing configuration through aar locations table	Direct call to Voice Mail did not work.	6.3.16.0
CM-16188	Two CMs directly connected with SIP Trunk, Service Observing, Initial IP-IP Direct Media? set to 'Y', outgoing direct media enabled	Call dropped	6.3.13.0
CM-16251	H.323 trunk group and CM configured to transmit a large number of DTMF digits when the far end answers.	A system reset could occur when CM attempted to transmit more than 32 digits via end-to-end DTMG signaling over an H.323 trunk group.	6.3.116.0

ID	Minimum Conditions	Visible Symptoms	Found In Release
CM-16275	SIP station had "rpxxx" configured on "SIP Trunk" field on stations form, and call forwarding was enabled on the SIP phone.	No call log was recorded on SIP phone when call forwarded.	7.0.1.2.0, 6.3.16.0
CM-16278	Call flow: SIP trunk -> VDN -> SIP trunk -> IVR -> SIP trunk -> VDN -> Agent and IVR does a supervised transfer of the call and SA9124 is enabled.	Connected number in ASAI transfer message incorrect.	6.3.16.0
CM-16501	H.323 stations with vu-display configuration	Topline display on H.323 phone got cleared when endpoint goes off-hook and on-hook.	7.0.1.2.0, 6.3.16.0
CM-16507	Call Center with CMS/IQ/Oceanalytics connected. Call Center agents defined with a mix of measured and unmeasured skills where their first logged-in skill is unmeasured.	When an agent's first skill is unmeasured and agent logs via CTI and CM did not send AUX23 message to CMS. CMS did not report the time of an agent spent in AUX accurately.	6.3.16.0
CM-16517	ASAI/SIP	Occasionally an ASAI event received the incorrect cause value resulting in a dropped call.	6.3.16.0
CM-16540	Incoming Call Handling Treatment (ICHT) is configured	ICHT on ISDN-PRI trunks did not work when called from a cell phone associated with a OneX Mobile station	7.0.1.2.0, 6.3.16.0
CM-16571	Call forward Incoming ISDN PRI Setup with No CPN (Calling Party Number)	Random digits displayed on station	6.3.116.0
CM-16577	New CM station addition	While adding a new station, an error displayed "Transient data conflict detected, please try again"	6.3.14.0
CM-16652	SIP Call	In some rare circumstances CM did reset	6.3.14.0
CM-16660	VDN, Dialer, Outbound Trunk call	Erroneous call pop-up at agent screen was seen.	6.3.15.1
CM-16724	Call Center using Multiple Call Handling (MCH)	When an agent was on a call and had ACW pending they could not receive a Multiple Call Handling (MCH) call	6.3.16.0
CM-16861	Customer must have more than 999 announcements administered.	Customer is blocked from adding 1000th announcement or more when doing "add announcement"	6.3.17.0
CM-16915	Coverage answer groups having 60 or more members and being team monitored by 10 users and ASAI monitored, CM attempted to transmit more than 32 digits via end-to-end DTMG signaling over an H.323 trunk group	CM did interchange and/or reset.	6.3.16.0

ID	Minimum Conditions	Visible Symptoms	Found In Release
CM-16917	CM, LAR configurations, no-hold-conference feature	Look Ahead Routing (LAR) did not trigger while no-hold-conference was in-progress	6.3.14.0
CM-16943	Call forward, SIP & QSIG trunk	The call forwarded terminating station did not display any forwarding information	6.3.12.0
CM-16955	SIP Bridge	The principal station locked up after its first call appearance joined a bridge call and the bridge phone conferenced the first line to the second line.	6.3.16.0
CM-17013	SIP endpoint as Group Page member	Non-compliant SIP History-Info header caused parsing error	6.3.14.0
CM-17068	Communication Manager with at least 15 agents being logged out at the same time. These agents must have an average > 40 skills each	Performing a Forced Agent Logout by Clock Time causes system reset.	6.3.15.1
CM-17073	Call Center with a SIP-connected adjunct using REFER without Replaces to route calls out of the CM.	CMS Reporting ignored calls that went through a REFER without Replaces and were answered off-site prior to completing the transfer processing	6.3.16.0
CM-17258	SIP station A has a brdg-appr of button X of station B and the value of X is greater than the button number of SIP station A on which the last call-appr or brdg-appr is administered	Unable to add bridge appearance on SIP Station	6.3.14.0
CM-17382	Add more than one codec, in "ip-codec-set" form.	Calls dropped due to CM did not increment SDP version	6.3.0.0
CM-17388	CTI agents transferring calls with 'Station Tone Forward Disconnect' configured to 'busy'.	Agents were getting busy tone and remaining on calls when transferring via a CTI application if 'Station Tone Forward Disconnect' was configured to 'busy'	6.3.13.0
CM-17421	EC500 and Service Observe (SO) enabled for the station And, station is logged off	EC500 Call dropped for logged off station	6.3.12.0
CM-17613	AES CTI Application	Agents were getting CTI error messages due to TSAPI CSTA timeout	6.3.15.1
CM-17614	H323 Telecommuter	A call cannot be answered on a telecommuter phone if the answer came 60 seconds after ringing started.	7.0.1.3.0, 6.3.16.0
CM-17619	Call Center, using Add/Remove Agent Skills via FAC	Changing agent skills using the Add and Remove Skill FACs had occasional delay in sending updates to CMS. Thus CMS was showing the agent in the OTHER state for minutes after the state change.	7.0.1.3.0, 6.3.4.0

ID	Minimum Conditions	Visible Symptoms	Found In Release
CM-17731	NATed h323 user	The H323 station behind the Network Translated Device (NAT) couldn't get dial tone if the user tried to go offhhok the first time after registration.	6.3.8.0
CM-17773	CM, AES CTI Application	cti-application received the wrong npi_toa(Number Plan Identifier - Type Of Address) on incoming trunk call to Agent/station.	6.3.15.1
CM-17812	LSP or ESS with no C-LAN board.	An installation or upgrade on an LSP or ESS may fail.	6.3.17.0
CM-17847	SIP Trunk, Call Transfer	An outgoing call over a SIP trunk might drop for some blind transfer call scenarios	7.1.0.0.0, 7.0.1.3.0, 6.3.17.0
CM-17867	Call Center with measured SIP trunks using ASAI/CTI.	For some types of outgoing SIP trunk failures, an extraneous, delayed trunk IDLE message was sent to CMS causing CMS to stop tracking the next call on that same measured trunk	6.3.16.0
CM-18000	Signaling groups configured.	In one instance a system reset occurred when signaling groups were used.	6.3.16.0
CM-18001	A system logging a new error.	In one instance a system reset occurred due to a corrupted linked list when a new system error was logged.	6.3.15.0
CM-18009	more than 255 network regions	An H.323 phone registers to CM in a stub Network Region and gets an incorrect Alternate Gatekeeper List.	6.3.14.0
CM-18011	Communication Manager with G650 Media Gateway	In one instance a reset system 1 occurred on a system with a G650 Media Gateway.	6.3.10.0
CM-18195	Stations with more than 2 button modules	Stations with more than 48 custom button labels will lose the labels that exceed 48 buttons. Generally this happens only on stations with more than 2 button modules.	7.0.1.2.0
CM-18196	ISDN Call, CDR	Null characters in CDR record if Calling Party Number has a leading '+' in ISDN SETUP message.	6.3.15.0
CM-18206	CM 6.3 or later, vectoring with queue-to skill step and VDN return destination.	A 'queue-to skill' vector command in the VDN Return Vector step will result in the call being queued at Low Priority regardless of the priority (top, high, medium, low) specified in the command.	6.3.11.1

ID	Minimum Conditions	Visible Symptoms	Found In Release
CM-18244	Monitored station is cleared when it takes over another station and TTI is enabled.	CTI events/messages were not provided to CTI applications when a monitored station took over a station that was registered to another extension when Terminal Translation Initialization (TTI) was enabled.	7.0.1.3.0
CM-18428	SIP stations/agents configured as non-ACD group or skill members.	No 'list trace hunt-group' System Access Terminal (SAT) command output was provided for calls terminating to a SIP station that was a member of a non-ACD (Automatic Call Distribution) group or skill.	7.1.0.0.0
CM-18477	Vectors and vector variables configured, orphaned data records.	In one instance a system reset occurred when vectors and variables were configured in a system due to an audit that encountered orphaned data records.	6.3.11.0
CM-18896	SIP call-center agent with a vu-display button above 9	The system may reset if the site has a SIP call-center agent with a vu-display button above button number 9 or a format number above 9.	7.1.1.0.0, 7.1.0.0.0
CM-18941	"ip-network-map" form "Emergency Location" field extension	list usage extension shows messed up IP addresses for emergency extension	6.3.12.0
CM-19285	Incoming SIP trunk with malformed UUI with UUI Treatment is set to "service-provider" and tandem over to ISDN/H.323 trunk with UUI Treatment is set to "shared".	CM did system reset	6.3.16.0
CM-20882	Incoming H.323 trunk call with a CPN to a station that is forwarded to another station shows 'Unknown' instead of the CPN. CPN/ANI/ICLID Replacement for Unavailable Calls: Unknown on System Features page 9. Replace Unavailable Numbers? y on trunk group form page 3.	Forwarded call shows unknown instead of CLI of the original caller.	7.1.2.0.0

Known problems

Known problems in Communication Manager 6.3.9.1

Table 33: Known problems in Communication Manager 6.3.9.1

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	
An agent would get a display number instead of display name for an external call when a Look Ahead Interflow (LAI) request by Communication Manager failed and the call was delivered to the agent on the Communication Manager system that made the LAI request.	111047	
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**.
A 2004 IP phone on Communication Server 1000 calls an 1140 IP phone on a Business Communication Manager. If the 1140 IP	120170	

Problem	Keywords	Workaround
phone blind transfers the call to a 96xx SIP phone, there is no talk path.		
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 PSN100127 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.
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	
Communication Manager would 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	
When the Auto Call Back feature is administered on a station that is a part of the multiple device access (MDA) feature, any attempt to invoke this feature on a busy extension will fail if that extension is active on another call.	131448	
Devices configured as part of the MDA feature will not display detailed conference information on their call appearance.	131475	
Some operations performed from the Elite Multichannel application after Session Manager (ASM) fails over cause inconsistencies between the status displayed on the application and that of the physical phone. Calls dropped from the application still remain up on the phone and, calls placed on hold from the application would remain active on the physical phone.	131524, 131525	
The File Transfer Protocol (FTP) has now been disabled on Communication Manager.	NA	
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.	NA	

Problem	Keywords	Workaround
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.		
The active server of a server pair running the Duplex Communication Manager Main/Survivable Core Template can experience a service outage when System Platform is upgraded or updated on the standby server.	NA	Perform the pre-upgrade step on the active server. Busy out the standby server and upgrade/update the System Platform. Release the standby server and verify the duplication state. Activate the Communication Manager Software update (service pack) on the standby server and again verify the duplication state. Perform a non-forced interchange of the Communication Manager servers. Busy out the previously active server which is now the standby and upgrade/update the System Platform. Release the standby server and verify the duplication state. Activate the Communication Manager Software update (service pack) on the standby server and again verify the duplication state.
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 Communication Manager 6.3.11.0

Table 34: Known problems in Communication Manager 6.3.11.0

Problem	Keywords	Workaround
When a. An active call at a SIP CC station involving the caller, agent and a Service Observer was placed on hold, b. The call was removed from hold, c. The Service Observer was observing an agent or a station, the Service Observer would not be reconnected to the call.	7241	
When "direct media", "Network Call Redirection" and the "Prefer G711 for Music/Annc" field are turned "on", a call between two stations, A and B, over a SIP trunk could experience problems with talk path if the following sequence of events happens: a. Station A and Station B are on a call b. Station B places the call on hold c. Station A and Station B are no longer talking to each other d. Station A places the call on hold e. Station A resumes the call f. Station B resumes the call If Station B resumed the call first, the talk path is restored when Station A resumes the call.	7255	Any one of the below operations will either restore the talk path or prevent the problem from happening Disable: a. Network Call Redirection b. "Prefer G711 for Music/Annc" field c. Shuffling Perform a hold and resume operation

Known problems in Communication Manager 6.3.11.1

Table 35: Known problems in Communication Manager 6.3.11.1

Problem	Keywords	Workaround
If the Avaya mobile SIP client disconnects ungracefully, EC500 calls to a called party number that is modified on Communication Manager no longer works.	8178	Re-enable EC500 for the station even though it shows ‘enabled’.

Known problems in Avaya Video Conferencing Solutions

This release includes the following known issues in Communication Manager 6.3.9.1 for Avaya Video Conferencing Solutions.

Table 36: Known problems in Communication Manager 6.3.9.1 for Avaya Video Conferencing Solutions

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	
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.
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. Call flow results in an audio-only call.	A92	While creating terminals or endpoints on the iVIEW suite, be sure to properly assign the matching protocol type, SIP to SIP stations and H.323 to H.323 stations.
re 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 8.2 or later, 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 8.2 or later.
iVIEW does not strip the prefix digits for outbound calls from iVIEW to Communication Manager.	R13/ QC19493/ QC15404	Upgrade to iVIEW 8.2 or later. For iVIEW 7.7, follow the admin steps in the Quick Setup Guide.
There is intermittent audio quality when Siren audio codecs are used for calls between Avaya 1000-series endpoints and the SCOPIA Elite	R14/AGS-289	Ensure that the Siren codecs are not in the

Problem	Keywords	Workaround
MCU.		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 the 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.
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/ ADVD-10062	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.

Problem	Keywords	Workaround
RMX dial-out to AV1010/20 leads to one-way video (Connect with Problem).	AVA-1551	Use dial-in on RMX.
ADVD may show severely distorted video with XT5000 embedded MCU.	A87/ ADVD-9909	This interop is currently not supported with FP2 and FP3.
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.
ADVD video calls made to a Radvision Elite MCU via an IVR result in audio-only connections for the ADVDs.	ADVD-10012	ADVDs should dial directly into the virtual conference room instead of dialing in via the IVR.
On Multi-Communication Manager audio calls between ADVD and Avaya one-X® Communicator SIP, after performing the Hold operation twice on the ADVD, users have audio and video.	10078	
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**.
lare video escalations from an audio-only call to Radvision H.323 XT endpoints going over an H.323 trunk remain audio-only.	130320	
XT5000 calls made to a bridged appearance on ADVD leads to an audio-only call.	130434	Currently, ADVD does not support bridging another station that is another ADVD.
Video SRTP calls to TLS registered HDX fail to connect.	131375	Use TCP signaling on the HDX.
Multi-Communication Manager Avaya one-X® Communicator H.323 calls in an XT MCU conference loses audio in one direction when video is stopped.	131684/ ONEXC-9211	Move the H.323 Avaya one-X® Communicators to instead be SIP registered Avaya one-X® Communicators.
Multi-CM transfers of Flare via Avaya one-X® Communicator to an XT MCU may fail.	131689	
Mid-Call Features are not supported behind the DMA.	131696	
When a Polycom Gatekeeper is involved, all Polycom entities should be associated with the Polycom Gatekeeper (DMA/CMA).	AVA-1562	
In a Multi-Communication Manager XT hosted conference, the Avaya	QC23239	Stop the video and restart

Problem	Keywords	Workaround
one-X® Communicator H.323 cannot become the active speaker.		it.
Radvision XT H.323 to Radvision XT H.323 calls end up with audio-only connection when any SIP endpoint transfers the call from one Radvision XT H.323 to another Radvision XT H.323.	131741	
Consulted transfers using SIP endpoints and a Radvision XT H.323 endpoint result in one-way video.	131746	Press hold/unhold or video stop/start to bring up two-way video.
A video call between two Avaya Communicators sometimes get stuck in hold when the two endpoints are simultaneously put on hold or simultaneously resumed.	131901	Drop and reestablish the call.
There is no talkpath after transfer of a Multi-Communication Manager call involving a Polycom VVX endpoint and an H.323 endpoint.	131950	
There is one-way talkpath between a Polycom HDX and a 96xx SIP endpoint when H.239 is enabled on the Polycom HDX.	131951	Disable H.239 on the Polycom HDX.
Video calls that traverse multiple Communication Managers may drop video when mid-call features (hold/resume, transfer, conference) are performed.	140915	
Calls started as audio-only Multi-Communication Manager become audio and video calls after performing the Hold operation and then releasing them.	SCAE-3910	Set Music On Hold (MOH) to No on Communication Manager.
Flare clients cannot access MCU Meeting Room via IVR.	QC23513	Dial into the meeting room directly.
An Avaya Communicator dialing into a Scopia Elite MCU cannot escalate to video if the call was initially made as an audio-only call.	QC-27593	If video is desired, dial into the Scopia Elite MCU as a video call.
An Avaya Communicator dialing into a Polycom RMX MCU cannot escalate to video if the call was initially made as an audio-only call.	FW-2158	If video is desired, dial into the Polycom RMX as a video call.
When an Avaya one-X® Communicator SIP client calls into a Scopia Elite MCU conference via IVR, the local screen may go black when local video mute is enabled.	ONEXC-1043 4	Instead of using IVR, dial directly into the Scopia Elite MCU conference.
When a Scopia Elite MCU dials out to an Avaya Communicator Windows client, the call sometimes comes up with very low resolution video.	SCAE-6229	Instead of using MCU dialout, dial in to the Scopia Elite MCU.
With TLS and SRTP encryption enabled, video calls may sometimes lose video when the call is transferred or conferenced (CM-hosted conference). In rare cases, the call may drop upon transfer or conference.		
Video SRTP with OneX Communicator Release 6.2 has the following known issues: • SRTP video with H.323 endpoints is not supported. • When Communication Manager-based conferencing is used: ○ There is loss of video when a third audio-enabled or video-enabled endpoint is conferenced or bridged onto a point-to-point video call. ○ After the third endpoint drops from the conference, the video re-established between the other two endpoints will be RTP, not SRTP. **Note:**		

Problem	Keywords	Workaround
Direct Media must be enabled.		
Shared control mode with 96x1 endpoints is not supported.		

Technical Support

Support for Avaya Aura® 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: Abbreviations

3PCC	Third Party Call Control
AAC	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
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
CR	Call Recognition
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
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