Avaya Aura™ Communication Manager 5.2.1 SP# 3 Release Notes

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Preventing toll fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there can be a risk of toll fraud associated with your system and that, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya fraud intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. For additional support telephone numbers, see the Avaya Support Web site:

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Providing Telecommunications Security

Telecommunications security (of voice, data, and/or video communications) is the prevention of any type of intrusion to (that is, either unauthorized or malicious access to or use of) your company's telecommunications equipment by some party.

Your company's "telecommunications equipment" includes both this Avaya product and any other voice/data/video equipment that can be accessed by this Avaya product (that is, "networked equipment").

An "outside party" is anyone who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf. Whereas, a "malicious party" is anyone (including someone who might be otherwise authorized) who accesses your telecommunications equipment with either malicious or mischievous intent.

Such intrusions might be either to/through synchronous (time-multiplexed and/or circuit-based), or asynchronous (character-, message-, or packet-based) equipment, or interfaces for reasons of:

- Utilization (of capabilities special to the accessed equipment)
- Theft (such as, of intellectual property, financial assets, or toll facility access)
- Eavesdropping (privacy invasions to humans)
- Mischief (troubling, but apparently innocuous, tampering)
- Harm (such as harmful tampering, data loss or alteration, regardless of motive or intent)

Be aware that there might be a risk of unauthorized intrusions associated with your system and/or its networked equipment. Also realize that, if such an intrusion should occur, it might result in a variety of losses to your company (including but not limited to, human/data privacy, intellectual property, material assets, financial resources, labor costs, and/or legal costs).

Responsibility for Your Company's Telecommunications Security

The final responsibility for securing both this system and its networked equipment rests with you — Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:

- Installation documents
- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

To prevent intrusions to your telecommunications equipment, you and your peers must carefully program and configure:

- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products

TCP/IP Facilities

Customers might experience differences in product performance, reliability and security depending upon network configurations/design and topologies, even when the product performs as warranted.

Standards Compliance

Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment is the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. might void the user's authority to operate this equipment.

Federal Communications Commission Statement

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

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Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformity Europeénne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC).

Copies of these Declarations of Conformity (DoCs) can be obtained by contacting your local sales representative and are available on the Avaya Support Web site:

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Contents

Changes delivered to Communication Manager 5.2.1 SP#3

Communication Manager 5.2.1 SP#3 Release Notes

The **Communication Manager** service packs are cumulative and changes in **Communication Manager** 5.2.1 SP#0, SP#1 and SP# 2 are included in **Communication Manager** 5.2.1 SP#3. The changes delivered to **Communication Manager** 5.2.1 SP#3 are grouped as follows:

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

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 **ASG** keys.

Product Support Notices

Some problems are also documented as Product Support Notices (PSN). The PSN number defines the related document and appears in the Problem column in the tables.

To read the PSN description online:

1. Go to the Avaya support site at http://support.avaya.com.
2. Under **Product Notices**, click **Product Support Notices**.

   The alphabetical list of documentation is displayed.
3. Click letter **P** in that list. All documents starting with letter **P** are displayed.
4. Click **Product Support Notices (All Avaya Products)**.

   The **Product Support Notices (All Avaya Products)** page is displayed.
5. In the web browser's **Find in Page** function, type the last four digits of the PSN number to search a link to the PSN on the page.
6. Click the PSN title link to open the PSN.

Communication Manager Messaging

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

1. Go to the Avaya support site at http://support.avaya.com.
2. Click **Products**. The **Enter Product Name** box is displayed.
3. Click **A-Z list**. The alphabetical list of documentation is displayed.
4. Click letter **C** in that list. All documents starting with letter **I** are displayed.
5. Click **Communication Manager Messaging**.

   The overview of **Communication Manager Messaging** is displayed.
6. Under **Product Information**, click **Downloads**.
7. Choose the appropriate release from the drop-down list and click the link to the **Communication Manager Messaging - Release x.y.z**.

Enhancements

This release includes the following changes that are new to **Communication Manager**.

Table 1: Enhancements delivered to Communication Manager 5.2.1 SP#1

Enhancement	Keywords	Workaround
The Enhanced System directory feature enables user to view and search station names that contains Katakana characters.	092538	
Message Tracer version 6.3.9.2 delivery.	093458 093663	

Table 2: Enhancements delivered to Communication Manager 5.2.1 SP#2

Enhancement	Keywords	Workaround
The Alerting Tone for Outgoing Trunk Calls feature provides the capability to apply an alerting tone to an outgoing trunk call after an administrable amount of time. The alerting tone is then repeated on the call at a specified interval until the call ends.	092065	
There was no major alarm when the software duplication link was not 1Gb/s.	093923	

Table 3: Enhancements delivered to Communication Manager 5.2.1 SP#3

Enhancement	Keywords	Workaround
H.248 Media Gateways, under certain conditions, cause Services nagging Minor alarms that are usually unworkable. A change is necessary so that alarm levels can be manipulated for these alarm conditions. A user will have the ability to allow for Major on a H.248 Media Gateway when the gateway has had its link down from the main server for longer than the link loss delay timer (LLDT). Also, a Minor can be allowed on a link bounce. These capabilities will be added to the "set options" form on page 2 of the SAT. The default will be that Major alarms (off-board) are downgraded to a Minor alarm and Minor alarms (off-board) are downgraded to Warning:set options Page 2 of 22: ALARM REPORTING OPTIONS.Major Minor Off-board Firmware Download Alarms: w. Off-board Signaling Group Alarms: m.Remote Max Alarms: w.CLAN Ping Alarms: w.H.248 Media Gateway Alarms: m w.	100443	
New field, "Callr-info Display Timer (sec)" on the "system-paramters features" form. The field gives the ability to administer a timer value between 3-60 seconds with a default of 10 that controls how long callr-info information is displayed on stations before the display goes back to "a=calling party info".	100771	

Problems fixed in Communication Manager 5.2.1 SP#0

This release includes the following fixes delivered to **Communication Manager**.

Table 4: Fixes delivered to Communication Manager 5.2.1 SP#0

Problem	Keywords	Workaround
Message Trace Analysis (**MTA**) did not work on servers running **Communication Manager** 5.2.1. The System Log web page returned the following message when the interpreted Message Tracer (**MTA**)" text box was selected: `User not authorized to execute mta, contact AVAYA. The reason may be that ACM is not running or the Trace Analyzer may be disabled in ACM admin.`	093416 093541 093662	
In a Avaya Aura™ **Communication Manager**-Feature Server environment, administered a core Modular Messaging (**MM**) unit. Administered two **IMS** extensions A and B. Extension B had coverage all to **MM** unit. A called B. Occasionally **Communication Manager**-Feature Server performed a warm restart.	093535	
You could lose calls or hang port networks when doing an upgrade with software duplicated servers.	093537	
On a software upgrade or system reboot, the previously active **IPSI** may not be remembered. This could cause another **IPSI** interchange when the server re-discovered the fault that caused the original **IPSI** interchange.	093538	
Incoming ip-direct shuffled calls to Avaya Aura™ **Communication Manager** may occasionally drop.	093539	
Some **SNMP** traps would not be reported.	093664	
Issues associated with the following keywords were also corrected in **Communication Manager** 5.2.1 SP #0: 093536, 093544		

Problems fixed in Communication Manager 5.2.1 SP#1

This release includes the following fixes delivered to **Communication Manager**.

Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***1 of 15***

Problem	Keywords	Workaround
When inter network region bandwidth was exhausted station was unable to transfer the remote station to another station in the same network region.	081880	
When a **VDN** (**VDN** A) with "Allow **VDN** Override?" set to "yes" was called over a trunk, and the call was routed to another **VDN** (**VDN** B) during the course of execution of the relevant vector; the called number field of **ASAI** Alerting event displays number of called **VDN** (**VDN** A) instead of active **VDN** (**VDN** B).	082587	
Selective drop for Meeting Exchange conference bridges over a **SIP** trunk was not handled correctly.	090050	
The Don't Answer Criteria For Logged Off **IP**/ **PSA**/**TTI** Stations?" field on page 3 of the system features form allowed calls to unregistered **IP** phones and out-of-service **DCP** phones. However, when an EC500 phone was administered to the station the feature was not working. Now it works with EC500 present at the station, regardless of whether EC500 is enabled or disabled.	091329	
In certain scenarios using Idle Appearance Select, Held Appearance Select Feature Name Extensions, Shared Voice Connection feature did not work correctly.	092038	
When a trunk call was transferred from a **SIP** station to another **SIP** station, the transferred-to station did not update the **CPN** prefix.	092474	
When a station-A made a priority call to station-B over a Qsig value added trunk, display on station-B did not show last five digits of the Calling Party Number.	092509	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *2 of 15*

Problem	Keywords	Workaround
Changing a media gateways' network region could lead to undesired results if the 'Force Phones and Gateways to Active **LSP**s' field was enabled in the "system-parameters ip-options" form.	092517	
When an outbound call was placed by the Adjunct user over an **ISDN** trunk, using Vector Directory Number (**VDN**) as the calling number the destination used to display only **VDN** number, no **VDN** name. Instead of VDN name either 'UNKNOWN NAME' or 'EXT' was displayed.	092565	
When an agent pushed an autodial button, the drop button and then a second autodial button in quick succession, both calls were reported to IQ/**CMS** as if they were the same call (that is, identified by the same **UCID**). It appeared as if the call dropped and then became active again.	092621	
When Natl/Intl **CPN** prefix was administered on the system-parameters features form, analog caller ID sets did not show Calling Party Number.	092633	
If a contact center agent lost connectivity to **Communication Manager** and that agent was available to a queue of customer calls, then it was possible for all the calls in queue to be delivered to this agent in a short period of time. All of these calls would fail. For this scenario to occur it required that the Redirect on **IP** Failure (**ROIF**) feature be disabled.	092654	
Integrated Services Digital Network trunk members may not get released if treatment for dialing unassigned numbers was set as announcement instead of tone.	092691	
Digits C and D were not tandemed on **IP**-**IP** trunk calls.	092736	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *3 of 15*

Problem	Keywords	Workaround
If a customer had an **IP** phone that did not support the advanced capabilities of the 96xx series, known as Spice capabilities, (such as a 4620 **IP** phone) and the user registered it to an extension that was administered as a 9650, then the button labels were not downloaded to the phone from **Communication Manager**. This resulted in mislabeled buttons on the phone, or in the case of the **VPN** (virtual private network) phone the entire screen was blank.	092762	
80 character display sets like CallMaster (606A1), 8434D, 7444D showed date/time when they registered and faced network/power outage.	092849	
Agent dial "0" calls to attendant in night service were not tracked by IQ.	092903	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***4 of 15***

Problem	Keywords	Workaround
When an agent's IP (H323) telephone had its headset enabled in manual-answer mode, and a network outage occurred, a Computer Telephone Integration (**CTI**) application incorrectly appeared to answer the call for that agent. As a result, the caller heard their ringback stop and then heard silence. The agent also heard silence since their network connection to the phone was down. But the **Communication Manager** incorrectly responded to the **CTI** application that the call was answered, with the implication that there was *talkpath* between the caller and the agent.	092907	Do not use the phone with the headset enabled. Then the station will not incorrectly appear as being answered and Redirect on No Answer will occur. OR Use the Redirect On Ip Failure feature and configure the agent as auto-answer. In this case, when there is a network outage for an agent's **IP** (H323) telephone, then a switch hook query will take place but no response will be received from the phone. The agent will then be moved into an unavailable (AUX) state and the call will be redirected back into the hunt group or skill of the agent who was supposed to receive the call.
If "Automatic Exclusion by COS?" is active for an endpoint used as an Enterprise Mobility User visitor that endpoint could freeze up.	092937	
The last button was not downloaded to the phone when the user used the **EMU** (Enterprise Mobility User) feature to login as a visitor.	092938	
For a call made to hunt group, a missed called entry was created on busy member station of type 96xx.	092945	
Auto Callback upon calling a busy station was not working on **IP** stations when reset shift timer was administered to some non-zero value.	092961	Administer the reset shift timer on the system parameters form to zero.
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *5 of 15*

Problem	**Keywords**	**Workaround**
Communication Manager incorrectly interpreted Call Management System (**CMS**) "vector contents" administration of a vector "set" step with "none" as a vector "set" step with '?'. For example: CMS sends: "set A = none add 1" CM records: "set A =? add 1".	093001	Use **Communication Manager SAT** administration instead of **CMS** to administer this "none" in the "set" vector step.
When a call was redirected to a remote server by **BSR** (Best Service Routing), both the polling call and the redirected call on the polled/ receiving server had the same **UCID**. This duplication of the **UCID** for the two calls adversely affected the IQ call reporting adjunct tracking of the calls. As a typical scenario, an incoming IQ/**CMS** measured trunk call went to **VDN** 2100. **VDN** 2100 executed Vector 100. Vector 100 contained three "consider" location steps. The Trunk groups associated with the poll **VDN** and redirect **VDN** on the **BSR** plan both had "Send **UCID**? y".	093003	
In a system with 5.2.1 sp01 or earlier and 5.2 sp04 or earlier **SNMP** Load Agent stopped responding even after the connection state was initialized.	093010 101094	
INADS modem alarming was not working after a migration from **Communication Manager** 2.x to **Communication Manager** 5.2.	093024	An additional backup and restore of the OS system can be performed after the migration is complete and the original 2.x data is restored. Once the second restore is complete the user will have to disable alarming via the almenable -d n or -s n command and then re-enable it via the almenable -d f|b or -s y command.
An extension with Call-Forwarding activated to an X-ported bridged-extension could not establish a *talkpath*, when the extension tried to answer the call from the bridged-appearance button.	093092	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *6 of 15*

Problem	Keywords	Workaround
National or international prefixes were not prepended when the call was forwarded and the answering party did not send the connected digits.	093106	
CMS versions prior to R15 incorrectly reported an increased number of abandoned calls when connected to **Communication Manager** Release 5.2.	093107	
A **BCMS** report showed that an agent's staffed time was more than 60 minutes in a one hour interval.	093134	
The "converse-on" vector step failed to pass digits in a **VDN** Return Destination vector if the incoming trunk call had been conferenced with a single step conferenced invisible observer. This single step conferenced invisible observer must be part of the call when **VDN** Return Destination vector processing starts, but must also drop from the call prior to the time that the converse-on step began passing digits.	093155	Keep the single step conference invisible observer on the call while executing the converse-on step. OR Drop the single step conference invisible observer from the call prior to executing the **VDN** Return Destination vector.
Setup calltype analysis table and administered an entry for **AAR** and routed it over Qsig trunk. Administered far end to re-route the call. Re-route failed.	093202	
2410/2420 type of **DCP** stations with firmware version-5 were not able to make calls using SpDial buttons.	093203	Make the call using the dialpad.
The caller did not see the hunt group name during alerting while calling a hunt group over a **QSIG** trunk.	093229	
This fix requires SA8887 - Hotline for IP Telephones. When SA8887 was turned on along with A/D Grp/Sys List Dialing Start at 01 field on system parameters customer options form page 3, the value of dial code field on 4th page of station form was decremented by 10 instead of the actual value entered in the field.	093240	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *7 of 15*

Problem	Keywords	Workaround
An attendant in telecommuter mode made a call to station that was not answered. The call covered to another station and there was no *talkpath* when answered.	093251	
Application Enablement Server (**AES**) endpoints configured with the Time-to-Service (**TTS**) feature active and registered on duplicated **Communication Manager** through the Processor Ethernet interface was "force" unregistered after two consecutive server interchanges.	093259	
Call center agent was locked listening to reorder tone when the incoming **ISDN** trunk disconnected because of a network failure.	093267	
SIP trunk members were locked up when improper REFER message was sent from the far-end **PBX**.	093269	
When a service observer dropped from an Audix recorded call (using Audix One Touch Recording), the call was dropped.	093272	
Customer could not add direct connect fiber using a DS1C configuration from the **SAT** (System Access Terminal) "Fiber Link Administration" form because more than one " ipserver-interface" was administered in the system. The customer saw the following error message: `Configuration not allowed with more than one IP Server Interface.`	093273	
When a station-A was in conference with **PSTN** stations, and then after the last party was dropped by station-A from conference, the conference display was not refreshed.	093282	
If a call was made over a **SIP** trunk, the originator did not hear ringback if 183 Session Progress was received after 180 Ringing and the far end had already sent back **SDP** information either in 180 Ringing or 183 Session Progress.	093283	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***8 of 15***

Problem	Keywords	Workaround
The system administration terminal "**`list trace vdn/vector`**" command did not show the "ani", "digits" or "ii-digits" value.	093300	Create an "ani", or "collect" vector variable and insert a no-operation vector step that uses the new vector variable for "ani" or "digits". The "**`list trace vdn/vector`**" will display value of that "ani" or "collect" vector variable. ii-digits does not have an associated vector variable so there is no workaround for that case.
If either the **VDN** service observer, the **VDN** or the agent connected to a call was associated with a class of restriction (**COR**) greater than 255, the **COR** restriction was not properly checked (a different **COR** was used).	093301	
When the user was in Manage Software and selected to copy **CMM** over, but later in the install step the user decided NOT to install **CMM**, then **CMM** would still be installed.	093313	
H.248 media gateways may not be able to register after a server interchange.	093316	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***9 of 15***

Problem	Keywords	Workaround
With a two-party call over a **CMS** measured trunk that queued to multiple skills and had an integrated-music announcement playing, if the trunk caller abandoned the call, **Communication Manager** sent unexpected events to **CMS**, causing the MIS link to drop. For example, an incoming call arrives at server-1 on **CMS** Measured Trunk Group 1 and goes to **VDN** 2033, executing Vector 33. The call then routes to server-2 on **CMS** Measured Trunk Group 5. On server-2, the call goes to **VDN** 2034 and executes Vector 34. The call routes back to server-1 on **CMS** Measured Trunk Group 5. On server-1, the call goes to **VDN** 2035 and executes Vector 35. The call queues to multiple skills, Path Replacement drops the inter-server trunks, and the caller hears an integrated-music announcement. The caller then abandons the call.	093322	
Avaya **Communication Manager** (**ACM**) did not send **TCP** keep alive message on **TCP** connection from PROCR interface and causes stale **TCP** connection. **ACM** sends INVITE messages on the stale connection and the message never get to the far end. This caused **ACM** to drop the call after the 32 **SIP** transaction timer expired.	093323	
When a video enabled 1XC endpoint called into the MX conference bridge over a video enabled **SIP** trunk and a wideband audio codec was being used in a direct-ip configuration, the 1XC user was unable to pass **DTMF** entered PIN codes to the conference bridge.	093327	
An **AAS** (Auto Available Split/Skill) agent could not be added to a 7434ND station configured with an ip-softphone, through the hunt-group form.	093334	
Call failures and degraded system performance could occur if **SIP** call hold times were longer than the session refresh value administered on the trunk group forms.	093338	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***10 of 15***

Problem	Keywords	Workaround
The call-appr of Toshiba SIP Phone(TSP)was stuck when TSP tried to park the second call on its bridge-appr, while the earlier call on its first brdg-appr was parked.	093348	
After a reset or interchange occurred, a 1X-Mobile would no longer be able to place calls and may no longer receive calls on the cell phone. Also, the deskphone would alert silently with no display.	093378	
The service-link mode was changed from permanent to as-needed if an **IP** Agent in telecommuter mode was logged into a previously registered ip station with the same extension.	093385	
QSIG users could not cover to **SIP** integrated messaging adjuncts in an integrated fashion if special development SA8904, Location Based Call Type Analysis, was enabled.	093386	
You could lose calls or hang port networks when performing an upgrade with software duplicated servers.	093387	
AD system labels were not displayed for display language set to 'unicode2', 'unicode3' and 'unicode4' even if SA8942 was enabled.	093394	
When fewer connections than CLAN ethernet links were active during an upgrade, then the upgrade was not connection-preserving.	093396	
In a **Communication Manager**-Feature Server environment **IMS** user A and **IMS** user B, located in the same **Communication Manager**-Feature Server, were in conversation state. User A called B. User B had a One Touch Recording (**OTR**) soft key. During the conversation, B pressed the **OTR** button. The connection to the remote messaging system was established, but another call appearance was activated on the extension of user A. The same call scenario as above, but user A was an unknown remote station either in a branch or a **PSTN** caller. In this case, the **OTR** key was rejected.	093397	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***11 of 15***

Problem	Keywords	Workaround
When user selected the License File page and selects the step to "Install the license file specified below" and there is a "role" change (server mode changes from **ESS**/**LSP** to main for example), then the "Restart CM" button was shown. When the user clicked on this "Restart CM" button, an error was displayed and **Communication Manager** was not restarted.	093400	Use the “Install the license file I previously downloaded option” instead and the “Reset CM” button works.
On a software upgrade or system reboot, the previously active **IPSI** may not be remembered. This could cause another **IPSI** interchange when the server re-discovers the fault that caused the original **IPSI** interchange.	093402	
Audio conference was not established between audio endpoints and Avaya Meeting Exchange when sip trunk was video enabled.	093405	
After a trunk call was transferred to a station, caller's number was seen along with the trunk name instead of Trunk Access Code (**TAC**). This problem was reproducible even when Special Application SA8851 was enabled and in station's COR form " Remove Incoming CallerID from Set Display" field was enabled.	093417	
When the "Force gateways and phones to active **LSP**s" field is 'y' in the "system-parameters ip-options" form, there would be occasions when **MG**s which have a time-day window recovery rule would not be forced back to the **LSP** at the end of the hour when not all **MG**s have registered with the main server. In addition, there would be occasions when running the '**`disable nr-registration`**" command would elicit the bogus warning message: `WARNING: this region is currently in a Time-Of-Day return period. Disabling this region could cause other regions to be automatically disabled at the end of the hour."`	093418	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***12 of 15***

Problem	Keywords	Workaround
In a **Communication Manager**-Feature Server environment, administered a core Modular Messaging unit. Administered two **IMS** extensions A and B. Extension B had coverage all to **MM** unit. A called B. Occasionally **Communication Manager**-Feature Server performed a warm restart.	093421	
When S8510/S8500B/S8500C was configured as Local Survivable Processor (**LSP**) or Enterprise Survivable Server (**ESS**), its Processor Ethernet (**PE**) interface could only supported 3500 sockets. Now, it will be able to support 24000. This fix must be applied to the S8510/S8500B/S8500C **ESS/LSP** before it can take effect. If it is only applied on the main server the current operation is unchanged.	093422	
Under certain conditions, **Communication Manager** with bridged stations in the setup experienced system restart.	093425	
When an Application Enablement Service (**AES**) Device Media and Call Control (**DMCC**) endpoint registered in shared control mode was requesting media and received a second call, **Communication Manager** sent a new audio channel to the **DMCC** endpoint before the endpoint (physical set or **DMCC** endpoint) answered the call. When this happened the **DMCC** (called) endpoint got a "ring back" tone on this new audio channel until the called endpoint answered the call. During this time **DMCC** endpoint may not listen to the original call.	093434	
No talkpath was observed when switching rapidly between two active call appearances on H.323 phones.	093438	
Adhoc Conference failed with MX and H323 One-X Communicator when SRTP was enabled.	093444	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 *13 of 15*

Problem	Keywords	Workaround
In a system with more than 127 combined **CLAN** and **VAL** boards, any of the following **SAT** commands could cause a system COLD restart of **Communication Manager** if executed on a particular board in the system. **`status ip-board`** **`status clan-ip`** **`status val-ip`** **`status link`** **`status clan-port`** **`netstat ip-route`** **`netstat arp`** **`refresh ip-route`**	093447	
Communication Manager could experience a system restart when making an **IP** trunk call when the Multilocation feature was turned on.	093449	
User must be a oneX Mobile user. The user must have had his oneX Mobile administration removed while one of his oneX destinations was on a call. A reset or interchange occurred. Afterwards, a oneX Mobile user would no longer be able to place calls using oneX Mobile and may no longer receive calls on his cell phone.	093451	Don't remove the oneX Mobile administration for a user while on a call.
When service observing a station that had an analog bridged appearance, service observers were not connected to calls through that bridged appearance even though the special application "Service Observe Physical Set" was enabled.	093495	
If an incoming R2MFC trunk call was forwarded unconditionally over **ISDN PRI** trunk, the call failed.	093509	
Incoming ip-direct shuffled calls to **Communication Manager** may occasionally drop.	093534	
After update of S8800 IMM firmware no alarms were generated in case of power supply or disk drive failures.	093555	
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***14 of 15***

Problem	Keywords	Workaround
Some default Midsize Business Template license right-to-use were modified to meet the new market direction for this offer.	093557	
When there were oneXMobile users on a **Communication Manager**, EC500 and PBFMC usage counts may be improperly decremented, so that they could not reflect actual usage.	093586	To avoid the problem, don't remove users more than once.
A supervisor agent used the vu-stats button to generate display information, including a non-zero Aux Reason Code. The supervisor agent Service Observed an extension and then disconnected the call. After selecting the vu-stats button again, the Aux Reason Code was incorrectly displayed as 0.	093610	
System may run out of memory when **SIP** calls enter connection preservation on failed session refresh.	093648	
Some **SNMP** traps were not reported.	093658	
When integrated with Avaya One-X Mobile, callback feature did not work in some scenarios.	093759	
If a user entered an almenable -s or -d command when no "secure-services n" entry was in the alarm_oss.conf file the file was populated with the string "secure-services" without a value. Without an 'n' value **INADS** modem alarming will not work.	093789	The string "n" will have to be manually added to the "secure-services" string. Once the file is changed the user needs to disable alarming via the almeanble -d n or -s n command and then re-enable it via the almenable -d f|b or -s y command.
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Table 5: Fixes delivered to Communication Manager 5.2.1 SP#1 ***15 of 15***

Problem	**Keywords**	**Workaround**
A user was set up to use oneX Mobile. The user's station had per button ring control set to y. The user logged a softphone (oneX Communicator) into the station in telecommuter mode. Calls to the station alerted the softphone but could no t ring the telecommuter phone until after the user answered at the softphone.	093835	
An agent with an observer was on a call and transferred the call to a **VDN** with a converse vector step. The converse step answered the call and the observer was connected. Then the agent pressed transfer and the observer remained on the call.	094039	
This change fixed T.38 fax failures when using TN2302 and TN2602 media processor circuit packs in the connection.	100370	
Issues associated with the following keywords were also corrected in **Communication Manager** 5.2.1 SP#1: 091165, 092252, 093030, 093097, 093129, 093221, 093481, 093520		
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Problems fixed in Communication Manager 5.2.1 SP# 2

This release includes the following fixes delivered to **Communication Manager**.

Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***1 of 20***

Problem	Keywords	Workaround
When a caller with Enterprise Mobility User (**EMU**) active on a visitor station pushed an autodial button whose target was an extension on the visiting switch, the call failed and the **EMU** caller received reorder tone.	074064	
The **QSIG** redirecting name was truncated incorrectly when the call was rerouted to the originating **Communication Manager**.	081834	
Remote party called into system over **SIP** trunk and got reorder tone. The **SIP** trunk must be using the **TLS** transport mechanism and the incoming SIP INVITE must arrive just as the **TLS** handshake completed. This resulted in delayed delivery of the INVITE message, which usually caused the first call to fail, but all subsequent calls proceeded normally.	082897	
When an extension with the Send All Calls (**SAC**) feature activated had an incoming call using Inter Gateway Alternate Route (**IGAR**) then the coverage point was not alerted instantly.	091148	
Displays may show an incorrect incoming phone number after **QSIG** path replacement operations.	091672	
Under certain conditions a filesync could fail, resulting in a MINOR alarm. The next filesync that passed insured all files were up-to-date.	091694	
EC500 calls failed if the trunk to the cellular service provider was an R2MFC trunk.	091755	
COR did not apply for a outbound proxy call. For a call originated from a **SIP** station administered on the **Communication Manager** for a user who was in an external domain, not administered on the system, the **COR** restrictions would not apply on the call.	092226	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***2 of 20***

Problem	**Keywords**	**Workaround**
Abbreviated dialing containing ~w failed on Media Gateway where dialing string contained remote access extension and authorization codes in it. When user pressed autodial button and dialed access code, dial tone continued without confirmation tone for service observing.	092256	
When R6 **IP** softphone registered in shared control mode with the unicode capable base station, **Communication Manager** would download button labels multiple times. As a result, it could cause the **TCP** socket congestion on the softphone.	092351	
Agent occupancy data could falsely display as being greater than 100%.	092475	
Unanswered calls to H.323 native endpoints were not properly disconnected.	092612	
Calls failed with intercept tone which otherwise should have been completed by Look Ahead Routing.	092693	
When a station was dialing out a call and another call came in to that extension but was picked up by another station, the display on the called station showed the call-appearance (a=) instead of showing the partly dialed string.	092729	
When ever shuffling was initiated due to master group change and the shuffling got rejected due to reInvite glare, then shuffling should be reinitiated once the glare scenario was over to avoid *talkpath* issue.	092845	
`Error encountered, can't complete request; check errors before retrying` message was displayed when ran the **SAT** command **`display/change display-messages button-labels`**.	093005	
In some cases EC500 did not work for XMOBILE stations. EC500 station failed to ring on incoming calls.	093029	
After Ayava Communication Manager reboot, third party call forward could not be disabled on Toshiba SIP Phone.	093161	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***3 of 20***

Problem	Keywords	Workaround
An unexpected Re-Invite was received from **Communication Manager** after the call to **VDN** was held by remote end.	093067	
SIP stations that used the public numbering format have to prepend a "+" to their extension number. When a public number did not start with a "+" calls from a **SIP** station did not succeed. Thus a "+" has to be prepended to the dial prefix of a station's "off-pbx station-mapping" table entry, if a corresponding match was contained in the public numbering table. Additionally it might have happened, that the field "dial prefix" of the "off-pbx station-mapping" table got overwritten with a false value after a reset of the system.	093112	
Far End sent ReInvite to **Communication Manager** and in counter offer changes the codec list from initial offer with only the last negotiated codec, for srtp calls **Communication Manager** failed to answer to this ReInvite.	093176	
Under extreme call volume conditions directed towards an unattended hunt group, the first call in the hunt group queue may be automatically dropped after not being answered for a period of time.	093178	
Infrequently, software duplicated **Communication Manager** main servers with H.248 Media Gateways and call traffic could interchange and reset.	093190	
An agent with an observer was on a call and transferred the call to a **VDN** with a converse vector step. The converse step answered the call and the observer was connected. Then the agent pressed transfer and the observer remained on the call.	093220	
After multiple link bounces, **ISDN** trunks and **DCP** stations sometime did not recover correctly to the point they were not working.	093244 093560 093581 091942	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***4 of 20***

Problem	Keywords	Workaround
2410/2420 type **DCP** stations with firmware version 5 were not accepting authorization code when the calls were made using internal speed dial buttons.	093250	Make the call using the dialpad.
Successful Two B-Channel transfer marked as abandoned due to parse error in parsing SetCallTag operation.	093305	
If there was a service observer on a **PCOL** (Personal CO Line) call and the **PCOL** station dropped, the call was not dropped.	093342	
Incoming ISDN Public Switch Telephone Network(PSTN) undergoing Q-Signaling path replacement are dropped by Public Switched telephone network.	093346	
When an **ESS** or **LSP** was added to or removed from translations, a MINOR alarm could occur on the main that could not be cleared.	093395	
Sometimes undesirable "ANSWERED BY TAC" display updates were seen when calls over **SIP** trunk were held by far end.	093414	
Customers would see the following error when doing seven simultaneous **`list history`** commands at the **SAT** (System Access Terminal): `Error Encountered can't complete request; check errors before retrying.` If customers did not do seven simultaneous **`list history`** commands, they will not see the problem.	093445	
When a service observer was on a call, **ECT** (Explicit Call Transfer) did not work. For example, an external call arrived through a trunk and was delivered to a station. The station transferred the call via the trunk. If the station was being service observed, the **ECT** did not work. The trunk channel did not get released.	093450	
The ports on the Processor Ethernet (**PE**) were blocked after enabling the **VLAN** 802.1q priority tagging.	093462	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *5 of 20*

Problem	**Keywords**	**Workaround**
Before, the count from the field "Number of features activated:" in the system-parameters special application form did not reflect the total number of activated features in the form.	093482	
Previously, in the case of flaky **WAN** link, video calls between **IP** softphones would fail intermittently. Now, they do not.	093490	
In an Avaya Aura environment which used **Session Managers**, it was possible that **SIP** stations did not always receive a correct update for their bridged appearances. This happened most likely if more than one **Session Manager** were controlling a single **Communication Manager**.	093497	
Changing many "Codec Set" or "Direct Wan" fields during a transaction on the **IP** Network Region form could result in a system reset due to high occupancy.	093514	
Dropping a **SIP** call failed if intial INVITE was received without Route header and without transport parameter in Contact header.	093529	
When migrating from a G3r to a Linux platform the tenant partitioning translation could be corrupted.	093542	
When the customer attempted to add an **IPSI** with **IP** Control disabled and blank **IP** Addresses, the transaction would be blocked and the cursor would be placed on a field that was not displayed.	093561	
ISDN PRI calls on H.248 gateways may intermittently not establish correctly.	093562	
Parked call was dropped if park originator dropped the call termed to a Q-Signaling trunk that had advice of charge feature enabled.	093567	
When an **IP** Softphone logged in telecommuter mode had call-forward-busy-don't-answer or enhanced call-forward no-reply active, any incoming call to the Softphone did not alert the telecommuter extension.	093577	Keep the service link up by making the 'Service Link Mode:' to permanent on the station form, page-2.
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***6 of 20***

Problem	Keywords	Workaround
Proc Errors are displayed for each call made on a **Communication Manager** Feature Server with **SIP** Sgrp set as **IMS**=yes.	093582	
When doing a `list history` command from the **SAT** (System Access Terminal) using a 513 terminal emulation, customers would occasionally see blank lines between data entries. Customers could use another **SAT** emulation to avoid this issue.	093585	
Denial events were generated and displayed on **DSP** measurement reports as Touch-Tone receivers were not getting handled properly because of Lookahead Routing functionality.	093587	
It took around 15 minutes or so for the Processor Ethernet interface (**PE**) to detect the **IP** phone's socket down if the phone's cable was unplugged. Now it would only take around three minutes.	093589	
CDR did not record the PIN Code for DECT telephone type originated calls.	093592	
With **CMS** R15 and later, when an agent in the Auto-In work mode with a Multiple Call Handling (**MCH**) skill placed a non-**ACD** (for example, extension-in or extension-out) call on hold, this action was not reported correctly to **CMS** in the periodic audit of measured agents and trunks. It may also have caused the **CMS** link to drop and re-establish.	093605	
When H323 station make a call to another station over **SIP** trunk and **IP**-hairpin was enabled, **DTMF** digits were not heard on called station.	093629	
The customer complained that **Communication Manager** was not sending events for some phone calls where the far-end off-switch user dropped first and the agent removed the resulting call trmt with a DropLast action and originating a new call.	093636	
The Total column of the BW Limit field on page 20 of ip-network-region form was displaying blank.	093643	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***7 of 20***

Problem	Keywords	Workaround
Custom labels on phone display were damaged after executing **`change station`** command on System Access Terminal (**SAT**) form on **Communication Manager**.	093653	
When SA8942 was enabled from the **SAT**, necessary changes to the web interface for “CM Phone Message File” were not enabled. To trigger the problem: 1. Do not install SA8942 license file. 2. Enable SA8942 from **SAT**.	093661	
List trace station and **TAC** commands now output **ROIF** (Redirection On **IP** connectivity Failure) events for the traced call.	093667	
When the **IVR** (Interactive Voice Response) channel status was monitored from the **IVR** side, the **IVR** channel randomly went down and then came up again in 1 second. This was caused by the station type being initialized as DS1FD and not DS1SA.	093668	
If a user on a Siemen's server calls a **Communication Manager** user over a **QSIG** trunk and the **Communication Manager** user was call forwarded, then the redirection display presented on the Siemen's user's display was incorrect.	093672	
When an Application Enablement Server (**AES**) endpoint, logged in Shared Control mode with a telecommuter **IP**-Agent or **IP** Softphone, unregistered the *talkpath* in the ongoing **IP**-Softphone/**IP**-Agent call would get lost.	093680	
An incoming call on Distributed Communication System Trunk (**DCS** Trunk) followed external coverage criteria if Time Of the Day coverage table was used as called station's coverage point.	093686	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***8 of 20***

Problem	Keywords	Workaround
A one-X Communicator, version 5.2 or later, was installed in a **Communication Manager** 5.1 test system with the station type 96XX. On the one-X Communicator a team button was assigned that pointed to a station that was allowed as monitoring station. When the monitored station was called, it was indicated on the team button of the monitoring one-X Communicator as blinking green lamp. Answering the ringing call on the monitored station erased the green lamp instead of displaying the active state of the monitored station. When the monitored station went into active state by establishing a call an update of the team button on the monitoring one-X Communicator did not take place.	093687	
For **SIP** calls, if the call was made using Avaya Switch Adjunct Interface (**ASAI**), under certain internal conditions of redirection the call was dropped.	093699	
Wrong display on transferring party when transferred call over Q-sig/**SIP**/**DCS** trunk and after transfer recall timer expired.	093701	
Prior to this fix, only 1 oneX Mobile server could connect to a **Communication Manager** through session manager.	093702	
False TTR-LEV alarms could be seen on **MBT** systems if the "ping" parameters on the "system-parameters ip-options" form was submitted.	093710	
Previously, **Communication Manager** would respond to an inactive fax re-INVITE with a null **IP** address and port. This caused the far-end to not generate a subsequent active fax offer, so the fax would fail. Now, **Communication Manager** responded with an **IP** address and a non-zero port. Since this was not interpreted as a rejection by the far-end, an active fax offer was subsequently sent.	093715	
Coverage answer group member could not answer the call if last member had call forward active.	093729	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***9 of 20***

Problem	Keywords	Workaround
If a member of a terminating extension group (**TEG**) attempted to service observe another member of the same **TEG** that was on the **TEG** call, service observing might not work.	093730	
For **SIP** calls, under certain internal conditions of redirection the call was dropped.	093732	
An incoming call to an unstaffed agent did not cover immediately to the coverage point, resulting in the call being dropped after a period of time.	093737	
When an incoming **ISDN** call was transferred locally, the called number in the EVENT CONNECT message was not reported correctly.	093753	
Call was getting dropped on **SIP** trunk due to Port missing in contact header.	093755	
If an incoming **TSC** call was received during SOSM incoming trunk monitoring, then an incoming trunk event report was generated that could not be parsed by **AES** and caused a library error message.	093768	
Call transfer was failing when a station that was was on an active call with an agent tried to transfer the call to a station of which Class of Restriction (**COR**) was different from that of the station where the agent has logged-in.	093778	Change the **COR** of the physical phone where the agent logged-in to the **COR** of the Agent/other stations.
For a customized legacy **CDR** format, if the **INS** field was administered with a data length of three, only 1-digit **INS** value was recorded in the **CDR** record, instead of the 3-digit **INS** value.	093781	
If a Trunk-Group, Hunt-Group, or **VDN** was changed from Measured:none to Measured:external, or Measured:both the associated name was not sent to IQ and so the name was not available for reports. For example, the **VDN** extension was displayed in Routing Point reports, but not the **VDN** name.	093783	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *10 of 20*

Problem	Keywords	Workaround
The customer saw the incorrect number of logins when executing the "**`status health`**" or "**`status media-gateways`**" commands at the **SAT** (System Access Terminal) when more than nine logins were active.	093800	
Upon calling a **XDID** (X-ported Direct Inward Dialable) number the "Hunt-to Station" endpoint did not ring.	093811	
Under some conditions, the processing of a system WARM restart was delayed up to two minutes. During the two minute delay, no service was being provided.	093825	
Agents were unable to retreive calls that were held. In addition, the agents display showed CONFERENCE on calls when there was no conference.	093828	
An **SNMP** Walk on the g3trunksta **MIB** group timed out.	093829	
IGAR (Inter Gateway Alternated Routing) call was dropped when call record audit was run.	093837	
Incorrect error handling of abnormal native H.323 station disconnects could lead to system resets.	093843	
When an Agent X who is logged on station A, calls station B that has EC500 feature enabled to station C over an **ISDN** trunk, station C always displayed agent information irrespective of the value of the field 'LoginID for **ISDN**/**SIP** Display?' on the agent form.	093849	
IGAR (Inter Gateway Alternate Routing) call was dropped when call record audit was run.	093855	
Previously, if call came in on **IP**-Trunk in Port Network 1 and then went out to a EC500 extension using a trunk in Media Gateway 1. Then there could be possibility of the call having no *talkpath* if the ip-codec-set between the Network Regions on Port Network and Media Gateway were using a wideband codec.	093863	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***11 of 20***

Problem	Keywords	Workaround
An agent in an Intelligent Customer Routing (**ICR**) solution was presented with the incorrect **UCID** information. Call flow: 1. Customer call arrived at the **ICR**. 2. Customer was prompted for agent information. 3. Customer entered a **VDN** number. 4. **ICR** generated a **SIP** Refer with Replaces request with the specified **VDN**. 5. Agent received the call and inspected the **UCID** using the ucid-info button. 6. The incorrect **UCID** was displayed. The **UCID** for the original customer call was expected.	093871	
A very small percentage of calls may fail under very high call traffic due to an internal software error.	093873	
Incoming R2MFC trunk call to a station which had CFU/CFB (Call Forward Unconditional/ Busy) active over another R2MFC trunk failed if the option 'Collect All Digits before Seizure' was enabled on the multifrequency-signaling form.	093879	
A server reset occurred due to a seg fault while making sip tandem calls under certain conditions.	093880	
The timestamp field in the g3alarmsAlarmNumber OID in resolved **Communication Manager** traps was not reporting correct information.	093902	
Unanswered extension to cellular (EC500) calls would erroneously cover to the cell phone provider's voicemail system instead of covering to **Communication Manager**'s voicemail system.	093911	
Call was getting dropped on **SIP** trunk due to transport type missing in contact header.	093924	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *12 of 20*

Problem	Keywords	Workaround
After an upgrade from G3SI to **Communication Manager** 5.2, Group Do not Disturb feature on attendent console did not set the time correctly.	093944	
At very high **SIP** traffic under certain conditions Avaya **Communication Manager** would dump core.	093951	
For calls transferred by **SIP IVR** to the **PSTN**, the called party display showed the dialed number instead of calling party number.	093957	
When a supervised transfer was made and transfer to party did not answer the call and call went to cover and pickup member answered the call, after the transfer was completed, the call got dropped. The problem was only reproducible when "Temporary Bridged Appearance on Call Pickup?" was disabled on "change system-parameters features" form and "Maintain SBA At Principal? " was enabled on "change system-parameters coverage-forwarding" form.	093965	
During a blind transfer for a call from **ISDN** to a **SIP** trunk, the calling party would get dropped from the call but call would terminate to agent.	093976	
When making an incoming **SIP** trunk call to an unregistered **IP** station, **SIP** response was not consistent and changed depending on the type of trunk group service.	093978	
The bash and **SAT** (system access terminal) interface was slow when **SNMP** trunk-group MIB (OID 32) would run because processor occupancy was high. This would also occur when **MSA** (Multi Site Administration) was doing a database synchronization. This would make **MSA** synchronization take longer also.	093992	
Previously, if the system did not undergo a **Communication Manager** restart of any kind, other than a single-process restart, for 994.2 days, the tmr_mgr process would trap and cause a WARM restart of **Communication Manager**. This should no longer happen.	093993	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *13 of 20*

Problem	**Keywords**	**Workaround**
Call reporting adjunct IQ could not adequately record calls processed by **SIP** merge with replaced for **ICR**.	094000	
When there was a large turnover of agents, such as during a shift change when many agents log off and log in, data corruption could prevent any **EAS** agent from logging in.	094012	
An **IP** Agent was active on two calls, one of which was a call to a vector involving a collect step with a repetitive announcement. When this **IP** Agent toggled between these call appearances multiple times, one of the two active calls dropped.	094049	
In an Intelligent Customer Routing (**ICR**) configuration with a **SIP** queueing trunk measured by IQ/**CMS** and a **SIP** redirection trunk NOT measured by IQ/**CMS**, an abandoned event was reported when the queueing trunk dropped from the call when the **SIP** Merge with Replaces took place.	094059	
IP stations capable of TTY operation can now activate TTY communication with an appropriate TTY device.	094064	
If a Visitor Enterprise Mobility User (**VEMU**) had all of its call appearances busy, under certain internal conditions, the system encountered a reset system 2.	094067	
Path Replacement not working for calls using call type analysis table.	094072	
When trying to remove a TN2602 board on the circuit pack form of a system that had numerous ip-interfaces enabled, the system hung and trunk groups went out of service.	094075	
If an Application Enablement Services (**AES**) Device Media and Call Control (**DMCC**) endpoint is registered using **TTS** feature and the associated **TCP** socket bounces (goes down and comes back up), the registered endpoint will stop receiving **DTMF** digits for the existing call and future calls until it re-registers.	094076	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *14 of 20*

Problem	Keywords	Workaround
When a call over a **SIP** trunk was unanswered for 10 seconds, a **CDR** record was still generated. This change fully addresses the problem when new trunk groups are added. If customers experience this problem on existing trunk groups and want to eliminate these **CDR** records, with this change they can correct this problem for existing **SIP** trunk groups by performing a "change-submit" sequence on the **SAT** for each assigned **SIP** trunk group without changing any field values.	094088	
For incoming R2MFC calls to stations which had EC500 enabled, the numbering plan identification/type of address (NPI/TOA) encoding of the calling party number sent on the outgoing **ISDN PRI** trunk was incorrect.	094089	
During a software upgrade, when there were more PPP links active than actual stable calls, the upgrade escalated and stable connections were not preserved.	100010	
A skill with a low number of logged-in agents or low number of calls could experience a condition in which the Expected Wait Time (**EWT**) as understood by Avaya IQ was not updated for a potentially long period of time. This impacted reports related to the skill.	100012	
The bash and **SAT** (system access terminal) interface was slow when **SNMP** trunk-group MIB (OID 32) would run because processor occupancy was high. This would also occur when **MSA** (Multi Site Administration) was doing a database synchronization.	100013	
There were situations where after a domain controlled user hit the DropLast button on a phone, the new call did not receive **ASAI** events.	100014	
When an agent transferred a call to an auto answer station, agent service observers were not removed from the call.	100021	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *15 of 20*

Problem	Keywords	Workaround
Calls covered/forwarded from **AWOH** stations always used the ALL LOCATIONS routing table.	100044	
Control over active call was lost when there was an incoming call to a bridged-appearance. Pressing the call-appearance button dropped the call. This happens only when the field "Display Information with Bridged Call" on system-parameters features was set to 'n' and the field "Bridged Idle Line Preference" on station form was set to 'y' for the bridged station.	100060	Set the field "Display Information with Bridged Call" to 'y', in system-parameters features form.
A call transferred out of Modular Messaging routing to an auto-answer agent could result in a 1-way *talkpath* between the caller and the agent.	100064	
The settings for the ethernet interfaces could not be changed on S8800 servers.	100070	
Under certain circumstances and traffic levels, users were unable to view vector and vdn-related call traces on the **SAT** using the "**`list trace vector`** or "**`list trace vdn`**" commands.	100071	
SIP calls were failing when Session Refresh INVITEs crossed causing **Communication Manager** to resend this INVITE. The problem involved **SIP** media that had been deleted too soon, causing failures on message responses in **Communication Manager**.	100078	
Receiving a large **SIP** message caused a memory overrun which resulted in a process trap.	100082	
Under conditions where stations were muted, it was possible for an inter Port Network to H248 GW IP voice connection using an older version of TN2602 media processor to report that activity on the voice path had ended causing the connection to the terminated.	100108	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***16 of 20***

Problem	**Keywords**	**Workaround**
Applications that use packet links, **PRI**, **BRI**, CrossFire, **CLAN** sockets (H323, **SIP**, **CDR**, **CMS**, IQ, etc.), etc., experience link allocation failures resulting in service outages. The scope of the outage depends on which links experience the failure. For **BRI** links and H323 station sockets, the outage is limited to single users. For **PRI** links and H323/SIP trunk sockets, a whole signalling-group is out-of-service. For **CDR**, **CMS**, IQ, etc., the whole application is out-of-service. The condition is triggered by a shortage of **IPSI**/ PKTINT resources, resulting in resource exhaustion on some **IPSI** boards. This can occur due to the system being incorrectly engineered (insufficient **IPSI**s provisioned) or because multiple **IPSI**s have been taken out of service (hardware failure, network failure, maintenance actions (busy-out), etc).	100115	
On the "System-parameter Features" form, the title "Auto-answer IP Failure Aux Work Reason Code:" has been changed to remove "Auto-answer" to reflect the actual operation of the feature.	100122	
A busy tone was sent to a caller if One-X mobile triggered station was present in a coverage answer group.	100123	
When the user picked up the call through the "call-pkup", "team" or "dir-pkup" button, the picked up call would be shown in the outgoing call log instead of the incoming call log.	100127	
Blind transfer by the called party put the caller on hold but not unholding after the transfer is complete, causing one way *talkpath*.	100128	
TDM/**DSP** blockage may result when attempting inter IP-connected port network calls. This blockage may result when a duplicated Crossfire media processor is shared amongst other port network media processor boards in a network region.	100171	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***17 of 20***

Problem	Keywords	Workaround
If an agent is in AUX mode, a call to the agent's ID correctly covers to Modular Messaging and reaches the voice mailbox. If music on hold is administered and playing while the call is going to coverage, when call is answered at the coverage point, music does not stop playing. For example, Administer music on hold and enable it for the agent's COR. Music is not integrated music. Administer a vector such as: collect 1 digits after announcement 4111401 for none route-to number 1110001 with cov y if digit = 1 where 1110001 is the agent extension with a coverage point that is a station. Turn on DAA: (sys feat form, page 11) direct agent announcement extension is 4111401. Make a call to the vdn that executes the vector and dial '1' to route the call to the agent. Let the call cover, have the announcement complete, and answer the call. The music is still playing for the caller when the coverage point answers the call.	100188	If music is integrated-music, the problem does not happen.
If an incoming call on a measured trunk-group is reflected out and back on an unmeasured **SIP NCR** trunk-group, and was subsequently transferred off-switch, the subsequent transfer was not reported to IQ/**CMS**.	100190	
After the **IP** phone logged out, it could not register any more.	100208	
After an upgrade from G3SI to **Communication Manager** 5.2, the Group Do not Disturb feature on the attendent console did not set the time correctly.	100212	
A tandem data call between two DS1s on two different IP Connected Port Networks could fail.	100228	
If the name field in a **SIP** HistoryInfo header was missing, the system would restart.	100231	
This change fixes T.38 fax failures when using TN2302 and TN2602 media processor circuit packs in the connection.	100234	
SIP trunk calls failed if the RFC 3389 Comfort Noise field was enabled and the Media Gateway was not Comfort Noise Capable.	100239	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *18 of 20*

Problem	Keywords	Workaround
If a **SIP** station used the "call pickup" feature, the call could result in no *talkpath* if the originating party was a SIP user.	100240	
When an observed station transferred a call to an auto-answer station that was also observed, upon completing the transfer the observer of the auto-answer station was no longer considered an observer, but rather a regular party on the call. If the caller then dropped off the call, the call would not drop because there were still 2 regular parties on the call.	100245	
When a user pressed the Mst_debug button or dialed the Message Sequence Trace (**MST**) Disable FAC to report a problem with a call that call was recorded in the list trace buffer. If a list trace was active at the time of the event the trace automatically switched to tracing the problematic call and output the following message to indicate this switch. 11:56:44 MST debug button or FAC station 5100 cid 0x14 If a list trace command was not active at the time of the event, the call trace was recorded in the trace buffer and could be displayed later using the list trace previous command.	100247	
SIP Call pickup not working when using "Call Pickup" button	100271	
Calls that were queued by **ICR** on a measured **SIP** trunk and subsequently delivered to an agent on an unmeasured **SIP** trunk were not tracked properly by IQ/**CMS**.	100279	
Comfort Noise was signalled even when Media Gateway did not support Comfort Noise leading to failure of **SIP** trunk calls.	100283	
Under heavy traffic conditions or **IP** network outages, entries in the **Communication Manager** logs showed that H.248 media gateway links were experiencing large delays in sending messages, often greater than 300 seconds.	100285	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 *19 of 20*

Problem	Keywords	Workaround
Unable to activate Special Applications SA8662, SA8684 and SA8661 regardless of the status of the Hospitality (Basic) feature.	100300	
Incoming and outgoing R2MFC calls to CO failed.	100366	
Could not downgrade from SP1 or SP2 to SP0 based update.	100372	
Communication Manager may reset if a call was made to a port on a H.248 media gateway and the media gateway was administered as the 250th media gateway.	100389	
SIP, H.323, and **ISDN PRI** trunks became unusable when stations with a type of XMOBILE were assigned in **Communication Manager**.	100452	
Unable to make a trunk call if authentication code was dialled digit by digit in time interval of 6-7 seconds.	100465	
"DID/Tie/ISDN/SIP Intercept" to Announcement was failing with Multifrequency Compelled R2 Signaling System(R2MFC) trunks when the caller mis-dialed the number.	100508	
When call was routed out of Vector Directory Number (**VDN**) to phantom and terminated on Toshiba **SIP** Phone, line four of the terminating phone display was not updated with **VDN** information.	100522	
IP DECT (Digitally Enhanced Cordless Telephone) phones were not able to transfer calls.	100545	
Page two of the **`Status station xxxx`** command displayed "0" in the FW Version:" field and its "Part ID Number" and "Serial Number" both displayed "unavailable" when a 1408 or 1416 set type was used with **Communication Manager**.	100554	
A **SIP** phone should be able to do six party conference with proper *talkpath* and proper display.	100602	
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Table 6: Fixes delivered to Communication Manager 5.2.1 SP#3 ***20 of 20***

Problem	**Keywords**	**Workaround**
If a **SIP** signalling group was administered with "H.323 Station Outgoing Direct Media? y", an outgoing call from a H.323 station to the **SIP** trunk, would sometimes not hear ringback.	100901	
An incoming **SIP** direct ip trunk call may not provide audible ringback to the calling trunk.	100994	
In a system with 5.2.1 sp01 or earlier and 5.2 sp04 or earlier **SNMP** Load Agent stopped responding even after the connection state was initialized.	101094	
Issues associated with the following keywords were also corrected in 5.2.1 Service Pack # 2.	083445, 091559, 092942, 093281, 093638, 093916, 093961, 093986, 100026.	
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Problems fixed in Communication Manager 5.2.1 SP# 3

This release includes the following fixes delivered to **Communication Manager**.

Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***1 of 18***

Problem	Keywords	Workaround
When "**UDP**->**ARS** Calls Offnet" option was enabled, the following calls were treated as on-net and not blocked:* toll analysis * forced entry of CDR account code * authorization code prompting.	083736	
Systems with several large IP trunk groups have trouble getting all of their trunks into service.	091253	
An SNMP walk on the "g3 version" MIB group did not return any data and caused a segmentation fault.	093145	
The user would see many of the following software errors in the error logs.**CM5_proc_err: pro=7195,err=203,seq=136660.**	092715	
On systems with more than 7 **ESS** clusters, duplicated ESS pairs could undergo server interchanges.	092828	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***2 of 18***

Problem	Keywords	Workaround
Issues associated with the following keywords are also corrected in **Communication Manager** 5.2.1 SP #3.	093033 093823 093824, 100144, 100166, 100344, 100442, 100556, 100585, 100603, 100647, 100862, 100910, 101021 100987 092215	
There was a window in which Kernel Updates could be committed before the LINUX reboot. If committed before the reboot the Kernel replacement doesn't take place and the update ends up in the activating state requiring manual intervention to correct.	093050	
Two Bearer Channel Transfer failed because glare occured while setting up second leg of the call. Handling of glare condition is incomplete for Network Call Rediretion.	093108	
An **SNMP** walk on the "g3version" MIB group did not return any data and caused a segmentation fault.	093145	
When call is initiated using autodial button and answered on **EC500** station, **DTMF** digits are played to wrong party.	093148	
Long Hold Recalling does not work on **SIP** phones when Session Refresh Interval on **SIP** trunk and Long Hold Recall Timer for the held call on the SIP phone are fired at the same time.	093226	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***3 of 18***

Problem	Keywords	Workaround
During call forward/transfer to external number diverting/redirection number is send as presentation allowed even if it is administered as restricted on station form.	093271	
Calls transferred out of a trunk integrated voice mail adjunct may not generate correct call detail recording (**CDR**) records.	093311	
When communication Manager receives 200 ok with port change or codec change against session refresh reInvite or Display change reInvite, the talk path should not break.	093518	
Call to an xported station say X having bridge appearance on station say B, B answered thecall from bridge appearance and then transferred to station C which was not answering. The call then covered incorrectly to X's voice mail rather than C's voice mail.	093580	
Call was not going to correct voice mail if an attendant was tranfering the call to another station which was busy.	093607	
Avaya Communication Manager does not respond back to farend switch for INVITE messages with only **T38** codec in it, when **Avaya Communication Manager** is already on call which supports only audio. This may lead to memory leak issues and system reboot.	093612	
When a remote **QSIG** user calls the **QSIG** Message Waiting Indicator (**MWI**)hunt group in an effort to dial into voice mail the remote user does not see the hunt group name and extension on their phone display.	093616	
When **ARP** requests are sent to a main **Communication Manager** server and only for a particular ipaddress, only the ethernet port with that ip address will respond.	093703	
If the Survivable **CDR** feature was enabled and the Local Survivable Processor went active then the very first **CDR** generating call was supposed to create a new Survivable **CDR** file. In some scenarios this file did not get created and hence the **CDR** were lost.	093711	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *4 of 18*

Problem	Keywords	Workaround
Missing or out-of-date **UNICODE** language files on a standby server (duplicate main server, **LSP** or **ESS**), resulted in system restarts when the standby server becomes active.This problem occurred when the **UNICODE** language files were being installed on the active server while the standby server was down. Since the standby server was down, the file copy to that server failed.The problem can be avoided by making sure all standby servers (duplicate main server, **LSP** and **ESS**) are up before installing the **UNICODE** language files.	093712	
When a caller drops while an integrated announcement is playing, before being connected to an agent, and a **VDN** service observer is on the call, the call continues to the agent as a phantom call.	093767	
If the agent is being notified by the Interruptible Aux feature, then an incoming or originating a call does not clear the notification display on the phone.	093792	Clear the display with the Normal/Exit button
When ~p~p is used in abbreviated-dialing for **SIP** station to **SIP** station call, **DTMF** digits administered after ~p~p in abbreviated-dialing are not heard at called station.	093846	
"Status link xxxx" command put ip stations sockets count in the field "**IP SIG GRPS** & MEDIA GATEWAYS" instead of "**H.323 IP** PHONES".	093875	
The 'list measurements trunk-group hourly' command gave inaccurate data intermittently for the **ATB** (All Trunks Busy) field. Occasionally it started showing 100% **ATB** during low traffic times and stayed at 100% **ATB** for a long period, such as over night, then cleared itself.	093890	
If a call was made to a hunt group and **IGAR** was triggered, then the agent did not ring if it had **EC500** enabled.	093910	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***5 of 18***

Problem	Keywords	Workaround
When an **ASAI** call is made and the call is forwarded to another station, calling party's call log was showing called party's number as UNAVAILABLE.	093917	
Call from a Siemen's user to a Communication Manager user over a **QSIG** trunk won't properly cover to voice mail if the Communication Manager user is "call forwarded no response" to the **QSIG** messaging hunt group extension and the **QSIG** messaging hunt group allows diversion by re-routing.	093955	
When a delayed socket closure results Communication Manager does not send a **SIP** UPDATE message causing calls to drop.	094033	
Calling Party Number was missing on Cell Phone if Auto Callback was activated over a **QSIG** trunk to an extension that has **EC500** active.	094091	
The count of **IP_API_A** stations shown on the display capacities screen was higher than the number of **IP_API_A** station actually registered.	100006	
A call over a **SIP** tandem trunk may cause a system reset.	100019	
Fax call dropped if call duration exceeded session refresh interval time.	100056	
Incoming Session Initiation Protocol call fails if the principal termination is forwarded.	100066	
Users could change the 'Ethernet Link: ' field on the ip-interface form when the link was assigned to a 'communication-interface processor-channel'.	100067	
When a **SIP** INVITE with Replaces message was received, the **UCID** coming in with the replacement trunk was not being retrieved.	100076	
Under certain conditions, Communication Manager resulted in segmentation fault when **H248** gateways were involved.	100080	
Wakeup call alerts ~3 minutes prior to the scheduled time.	100087	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *6 of 18*

Problem	**Keywords**	**Workaround**
When **Communication Manager**(**CM**) sends session refresh reInvite to one of the End point, it responds with 488 and the call get dropped. But as per **RFC3261** the call should not be dropped.	100112	
Dial Plan Transparency fails when triggered by a **SIP REFER** message.	100130	
Communication Manager undergoes restart when the active Crossfire board in a duplicated setup was disabled.	100137	
Caller does not get audible feedback for calls made on **LAR** configured route pattern on Communication Manager.	100160	
Not able to do a remote ping or trace route from an **IP** endpoint on a S8300.	100174	
Session Refresh Invite is answered in 200OK by Avaya Communication Manager such that the codec list offered in Invite is answered back in answer in 200OK having a codec not administered in Avaya Communication Manager	100198	
Agent's display showed incorrect information when the incoming **SIP** trunk call shuffled under certain conditions.	100258	
If **Communication Manager** receives a REFER message with an escaped URI, the switch may reset.	100261	
Under certain system administration scenarios, the system can deadlock causing complete system reset. At least one scenario is having an administrator running a long list of remove station commands using the import remove station feature of **Avaya** Site Administration and a "list measurements clan ethernet" report on another **SAT** (System Access Terminal) at the same time.	100270	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *7 of 18*

Problem	**Keywords**	**Workaround**
- When an agent A logged on station B makes a local or **ISDN** trunk call to station C, station C shows name and number of B if the field "Identity when bridging" on system parameters form is set to station.	100274	
If a Team button was administered to a button on the second button module of a station then Send All Calls (**SAC**) Override would fail.	100275	
In a CM Evolution Server the feature "Extension to Cellular" (EC500) failed for SIP stations, if the EC500 destination was located in another CM Evolution Server or CM Feature Server.	100280	
The **ASAI** Alerting Event doesn't report the correct called number when an incoming **ISDN** call terms to a **VDN**, the Redirect on No Answer feature is enabled and the field **VDN** Override for **ASAI** Messages is set to **ISDN** Trunk.	100281	
Errors are logged to the **Avaya Aura**(TM) Communication Manager system debug logs when the maintenance process tests **H.248** media gateway **CO** trunk ports. Errors are also logged for calls that are merged that involve two or more **H.248** media gateways.	100314	
One-X portal which was configured to use **DMCC** license was unable to login in Telecommunicator Mode via **AES** and get denial event as "Registration rejected because maximum number of registered endpoints of specific product and version exceeded".	100316	
When **TTI** was enabled, and a station calls to X-ported station covered to Voicemail via **SIP** trunk was hearing generic greeting.	100317	
When a call covers to an x-ported station in a coverage path of the called **DCP** station then if the station to which the call covers contains a bridge appearance of the x-ported station as well as a team button then only the bridged appearance would indicate a call and not the team button.	100328	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *8 of 18*

Problem	Keywords	Workaround
The green lamp of **BCA** on soft phone registered in shared control with the **H.323 9620** base set was not updated to green:steady state when the bridged call had been answered by primary.	100329	
Systems with several large IP trunk groups have trouble getting all of their trunks into service.	100331	
Internetwork region test uses out of service boards to use for test. An enhancement is being made to the change ip-network-region SAT form so that the Meas field is renamed to Mtce on the form. This will allow a user to administer inter-region connectivity testing either by using ping measurement data, or by using the normal inter-network region background test (#1417), or a user could turn off testing for this region pair, e.g., in the following screen test 1417 is on for the region pair 1,5. If the "t" is changed to "m" then ping measurements will be done to test the region pair, if the value is changed to "d" then testing is disabled for that region pair.	100335	
Enhanced System Directory, does not allow users to edit second attribute 'telephone number' from search and detail screen settings from web administration pages.And allows , in the Native Name(Unicode) in the external numbers add page. It now adds a check for the minimum number of characters entered on the search string on the phone browser which is adminstered from the search screen settings web page. It displays correct displayname attribute value when configured as one of the detail attribute in Detail screen settings web page.	100343	
The 'Type' column of the 'list measurements ip dsp-resource pn summary today' form was showing corrupted data in form of random characters.	100347	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***9 of 18***

Problem	**Keywords**	**Workaround**
The user could not originate a call from a bridged appearance while using on-hook dialing (dialing the number directly from the keypad without going off-hook/pressing speaker button) when the station feature "Auto Select Any Idle Appearance" is enabled.	100359	Dial by going off-hook or using the speaker button.
The "**Cvg** Enabled for **VDN** Route-To Party?" feature on "Coverage-path" worked only for the first "route-to" step in a vector.	100368	
The list measurements announcements board today-peak report can have duplicate entries for the same extension on the same board. There should only be 1 today-peak hour on a specific announcement board per extension.	100386	
A system reset or server interchange might turn off the User Guidance Display feature.	100407	
IGAR (Inter Gateway Alternate Routing) calls failed if the **IGAR** trunk call was routed over an **ISDN** trunk group and if the call was retried because the **PSTN** selected the same B-channel for an incoming call at exactly the same time. (This is known as "glare").	100409	
MCP single process restart causes call processing to reset.	100413	
Large number of **H.323** stations registration resulted in system reload.	100415	
A **DCP** phone which was on a call showed date and time instead of the caller/callee's name when media gateway failed over to **LSP**.	100444	
Display on Call Master 603 was frozen after a call was transferred.This happened once in 30 to 40 calls when the Enhanced Conference/ Transfer Displays and Vu-Stats display were enabled.	100445	
BSR calls failed when 95 or more bytes of **UUI** data was being transported between Communication Managers. The data from the polling **VDN** was not returned so the call was disconnected.	100471	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***10 of 18***

Problem	Keywords	Workaround
Earlier Telephone Set Type of Series 84XX were not allowed to have **EC500** stations via "change off-pbx-telephone station-mapping" form. Now, support is added to administer **EC500** extensions to **84XX** sets via said form.	100489	
If an incoming **SIP** trunk call was made to an attendant and the attendant put that call on hold and made another call, then when the attendant did a split-swap to return to the previous call, there was one-way talk path. The problem was only seen if the "Network Call Redirection" option on page 4 of the SIP trunk group form was enabled.	100544	
When a **96xx** phone user attempted a blind transfer of an incoming trunk call to an "x-port" station which had a remote coverage path, the **96xx** phone user was unable to transfer the call untill the Remote Coverage cell phone answered the call.	100550	
When one call is already active subsequent callers hear busy instead of the ring tone when calling a station that had the limit calls (**LNCC**) feature activated and was answered from a bridged appearance.	100560	
CPUPERF alarms will only be generated when there are cpu performance problems on the server.	100570	
Offnet forwarding on a station does not work if incoming trunk call to this station has nil calling number.	100576	
On a server with more than 10,000 stations administered, **SNMP** walk on g3station caused MVSubAgent to hang.	100586	
When a Communication Manager is connected to a **CMS** older than **R15** (or **IQ** older than 5.0) and a service observer is on a call that is being conferenced or transferred, an extra message is sent to the reporting adjunct that can cause confusion.	100590	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *11 of 18*

Problem	**Keywords**	**Workaround**
Wrong voicemail greetings played on SIP integrated Modular Messaging when Agent blind transfers the call to a busy station.	100591	
EC500 Timer button gets blanked and/or button's lamp is turned off after periodic maintenance runs on related terminals.	100592	
When a Vector Directory Number (**VDN**) was monitored through **ASAI**, the number of calls in the queue was showing one less than actual number if the number of calls was greater than one.	100595	
Internal server audit sometimes cannot clear software corruption causing Communication Manager to restart.	100596	
If an incoming **R2MFC** trunk call was forwarded over another **R2MFC** trunk and the forwarding station had coverage point administered, the call to the forwarded to party was dropped after 1 ring and the call covered to the coverage point.	100597	
IGAR(Inter-gateway Alternate Routing) calls denied due to insufficient **FRL**(Facility Restriction Level) gave incorrect data in denial event.	100605	
Time-To-Service (**TTS**) phone could not get back in service after being powered down and up.	100612	
Calls fail to tandem through Communication Manager when **LRQ**(Location Request) is used.	100618	
If a call was made from a **EC500** associated cell phone to an unregistered **IP** station over an outbound trunk the outgoing trunk channel was used from the wrong **PN**.	100635	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *12 of 18*

Problem	**Keywords**	**Workaround**
When an Extension to Cellular (**EC500**) station, made a call on behalf of its principal station, by dialing the Feature Name Extension (**FNE**) of "Select Idle Appearance", over a **Qsig** trunk, the display on the called party showed the name of the **EC500** station instead of the calling principal station.	100636	
An Incoming **SIP**(session initiation protocol) trunk call to **Communication Manager** was answered by an **IP** agent. When the **IP** agent transferred the call to another station, there was no talk path.	100666	
Under certain internal conditions, **Avaya Aura**(TM) **Communication Manager** may reset if there is a server interchange while SIP calls are up.	100669	
Links to **IQ/CMS** were not remaining in-service across a **PE** Interchange.	100685	
If a remote **QSIG** user calls the **SIP** adjunct messaging hunt group in an effort to dial into the **SIP MM**, then the remote user does not see the hunt group name and extension on his/her phone display.	100691	
Call-Forward button is not updated even though call-forwarding is activated for the station when Far-end Domain name is Null for sip signalling group.	100706	
After Time-To-Service IP phone re-registered, **Communication Manager (CM)** would not initiate a **TCP** socket immediately after its registration even if the **CPU** occupancy was low. If the phone sent the Admission Request (**ARQ**) to **Communication Manager**, **Communication Manager** sometimes also would not initiate the socket to this **TTS** phone.	100713	
Enhanced System Directory, does not allow to add , (comma) in the Native name(Unicode) field in external numbers add web page.	100716	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *13 of 18*

Problem	**Keywords**	**Workaround**
A SIP(Session Initiation Protocol) phone on ACM(Avaya Communication Manager) should able to do conference up to 6 party by calling each of the recipient(up to 5 recipient) and join them on the conference one by one.	100731	(Identify what the customer can reasonably do to avoid the problem or, if unavoidable, recover from the problem):
When call was routed out of Vector Directory Number (**VDN**) to phantom and terminated on Toshiba SIP Phone, line 4 of the terminating phone display was not updated with **VDN** information.	100732	
There was no talk path on a call from a Toshiba SIP Phone to another Toshiba SIP Phone in a call pick-up group that was picked up by a third Toshiba SIP Phone in the call pick-up group.	100733	
If there was an announcement with extension 0 in a translation, "list announcements" wouldn't list any announcements.	100739	
With certain multi-location dial plan administration, if a user presses a button that's been preprogrammed with a full extension, the call fails, because CM tries to route it using the per-location dial plan entry. Such buttons include CM buttons like busy-ind, call-disp (from directory), and team, as well as phone buttons such as 'redial' and calls placed from the call log on 96xx phones.	100755	
A trunk call from inactive to sendonly mode should not be dropped rather the call should stay on **TDM**.	100772	
For Network call redirection(**NCR**) trunk call, after transfer is complete,shuffling should take place if the remaining end points are shufflable.	100785	
CDR did not capture the agent extension that answered the call if the incoming call traversed through Voice Portal and converse-on vector steps.	100786	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***14 of 18***

Problem	**Keywords**	**Workaround**
The "status ip-board [loc]" command, for some, but not all, **IP** boards, showed all zeros, including the Reset date and time.This started occurring after midnight of the 1st day after a system reset/reboot or after midnight of the day that a board was inserted.	100803	
For Time-To-Service (TTS) Duplicate Processor Ethernet (DUP PE) capable phones, if a phone sends CM a setup message after a socket was re-restablished, CM would not respond with the Call Proceeding and Call Connect message. As a result, when Annex H was enabled, the phone could be stuck and could not make phone calls.	100817	
The **CDR** fie created in /var/home/ftp/**CDR** directory had the incorrect server id and module id.	100827	
Under certain internal and/or external conditions, **Avaya Aura**(TM) **Communication Manager** may not be able to connect users to announcements or music-on-hold. This primarily effects customers who have H.248 media gateways connected to **Communication Manager**.	100835	
Craft requested 'reset system 1' resulted in second reset.	100839	
In a configuration where a mix of fiber connected Port Networks and **IP** connected Port Networks co-exist, we could end up in a situation where a **TDM** endpoint,on fiber connected Port Network, acting as a service observer cannot hear a **TDM** endpoint based agent connected to a **IP** connected Port Network.	100842	
ASAI-originated calls that cover remotely to Voice Mail don't hear the welcome greeting.	100849	
User loses **SAT** sessions until they eventually can not login.	100861	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***15 of 18***

Problem	Keywords	Workaround
Avaya Aura(TM) **Communication Manager** may send duplicate **DTMF** digits to a phone or trunk if the phone or trunk is connected to a H.248 media gateway.	100868	
When Far end rejects capability negotiation offer from **Avaya Communication Manager,Avaya Communication Manager** shall retry the offer in Invite with "**sip**:"and no capability negotiation attributes.	100876	
g3intvlPollingIntvl does not reads value from mvmgt.conf file at initialization.	100878	
An **IP TTI** phone always is given Group ID 0. A wrong Group ID can cause massive trouble including wrong firmwares and wrong call managers.	100880	
The "list measurements ip dsp gw summary today" command showed 99% and 98% for all **GWs** above 60 in the "Out of Srv" field for every hour. This happened independently of the amount of traffic on those **GWs** and was clearly incorrect.	100892	
If call to a bridge appearance triggered **IGAR** (Inter-Gateway Alternate Routing) then the call failed sometimes.	100893	
Under some internal conditions Avaya Communication Manager ends a call when "Alternate Route Timer" expires even if Look Ahead Routing(**LAR**) was not administered.	100899	
No alarm was shown when **AC** power was lost on an S8800 server with redundant power supplies.	100903	
Bridge record and button audits fail due to translation corruption. When a warm start occurs, the audits run and if they fail they will cause a CM reboot.	100907	
Caller used to hear a ringback thrice upon calling a duplicate X-ported analog station that has merged-unmerged using **PSA** (Personal Station Access) feature access code.	100920	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 ***16 of 18***

Problem	**Keywords**	**Workaround**
The customer can administer a location in the "Location for Routing Incoming Calls:" field on signalling group form that is not defined.	100924	
Firmware download would not work if the source board (e.g., **CLAN**) and the target board were in two different ip connected port networks.	100930	
The Backup/Restore **SMI** pages did not accepted a backslash in user name field which was needed for domain user names of some **FTP** servers on Windows.	100952	
Previously, the H.323 station calling out on a **SIP** trunk, which had the option "H.323 station outgoing Direct Media?" set to "y", would get no ringback. Now, it will.	100954	
SNMP walk on g3platcmds **MIB** caused the **MV** sub agent response time to progressively slow down until it came to a halt when reading large amounts of data. Now data for only the last 1000 commands executed on the server will be reported.	100989	
After migrating from 5.2.1 to **Communication Manager** 6.0, if the administered ROIF timeout (Switch Hook Query Response Timeout) value is non-blank, **ACD** calls to "softphone" 7434ND agents on an **IVR** fail although they should have redirected.	100992	Change sys feat page 14, Switch Hook Query Response Timeout to blank to turn off **ROIF**.
Provide SIP output for list trace station and tac commands that route over a SIP trunk.	100998	
Calling Party Number (CPN) was not sent to the far end and thus the far end rejected the call.	101006	
Enhanced System Directory, does not display Name or **CN** attribute when displayname is configured as one of the search attribute in search screen settings administration web page.	101007	
The login into the **SAT** console failed for an **LDAP** user.	101029	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *17 of 18*

Problem	Keywords	Workaround
Ringback failures seen when calling from a **DCP** phone on one switch via a shuffled direct **SIP** trunk to a **DCP** phone on the far end when the **DCP** phone is in a different network region than the direct **SIP** trunk.	101030	
When using **Avaya** Site Administration **GEDI** mode, if there are any Policy Routing Table numbers that are 4 digits long (e.g., 2001), list policy-routing-table displays only 3 digits (e.g., 200) - the last digit is truncated.	101034	
Telecommuter calls fail intermittently.	101036	
Enhanced System Directory checks for Require a min number of characters in search string.	101086	
When running SIP call traffic, a lot of SIP "422 Session interval too small" messages were being generated under call load.	101130	
Under certain internal conditions, **Avaya Aura**(TM) **Communication Manager** may reset, interrupting call service.	101140	
Previously, the CLAN H.323 listen socket on port 1720 got "stuck" and would not allow incoming connections (new phone registrations). This would happen if there was a sudden burst of traffic to a newly created service CLAN.	101170	
There was no audio path for the IP direct call between two Nortel 1100 series SIP phones registered to Session Manager(SM), supported by an Avaya Aura (TM) Communication Manager - Feature Server(CM-FS) using Off-Premise Station (OPS) configuration and the 'ip-codec' set was G.722-64k, G.711mu.	101217	
After this fix, a call will be able to go to the hunt group without failure.	101224	
Previously, CMS / IQ systems connected via a Processor Ethernet would not survive a server interchange. A pump-up would occur.	101325	
Under certain conditions Conference call over SIP trunks had one-way talkpath.	101332	
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Table 7: Fixes delivered to Communication Manager 5.2.1 SP # 3 *18 of 18*

Problem	Keywords	Workaround
Enhanced System Directory rpm may not work due to the path issue of the rpm when built the same patch with Service Pack SP2.	101351	
If an incoming SIP Via: header contained multiple "received="parameters (a malformed Via: header), Aura Communication Manager could restart if the problem was persistent.	101374	
Avaya Aura (TM) Communication Manager (CM) could experience one system restart when the remote office h.323 IP station sent a registration request (RRQ) to CM.	101375	
Under certain internal conditions, Avaya Aura(TM) Communication Manager, executing SIP calls, will run out of memory, forcing a server reset.	101472	
List trace station and tac commands may record the wrong SIP trunk that is not associated with the traced call.	101488	
The Time-To-Service(TTS) capable IP phone could be stuck in registration loop when a large number of buttons are administered for the phone.	101583	
When displaying VuStats with a display format that contains a split reference that does not exist for that agent, the AUX reason code field does not update immediately upon changing. It does update when the interval is refreshed. A workaround is to make sure the split references in the vustatus display format exist for that agent.	101601	
Many Intra switch CDRs were generated without extensions for calls involving TSCs(Temporary Signalling Connection) if the feature (SA8202) - Intra-switch CDR by COS was enabled.	101605	
Under certain internal conditions, Avaya Aura(TM) Communication Manager may reset, causing call failures and service disruption.	101768	
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Known problems

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

Table 8: Known problems in Communication Manager 5.2.1 SP#2

Problem	Keywords	Workaround
For systems running **Communication Manager** 5.2.1 SP #1 (17849) or greater, calls over **SIP** trunks between branch gateways will fail if the signaling group parameter RFC 3389 Comfort Noise is set to "y" and the gateways are running firmware versions less than 30.11.3.	100283	Set RFC 3389 Comfort Noise to "n" or upgrade all gateways to firmware version 30.11.3 or greater.
A call covering to **SIP** integrated Modular Messaging (**MM**) through a **SIP** trunk does not cover to the correct voice mailbox and erroneously receives the generic greeting. There are possibly additional issues with conferencing, coverage, forwarding, or transfers that might be encountered with **SIP** calls. See PSN002766u for additional information.	091180 100859	In **Communication Manager** (CM) 5.2.1 and later releases, calls traversing **SIP** trunks, in addition to calls traversing **ISDN** trunks, must have the corresponding extensions administered in the appropriate private and public numbering tables on the CM **SAT** forms (public-unknown-numbering & private-numbering forms). Calls to extensions terminating on a **SIP** integrated **MM** must now be populated in the appropriate public-unknown-numbering or private-numbering **SAT** forms and not have a prefix configured for the **SIP** trunk group by which the MM is integrated.

Technical Support

Support for Communication Manager is available through Avaya Technical Support.

If you encounter trouble with Communication Manager:

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

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

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

When you request technical support, provide the following information:

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

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

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

Appendix A: Acronyms

AAR	Automatic Alternate Routing
AAS	Auto Available Split/Skill
ACD	Automatic Call Distribution
ACM	Avaya Communication Manager
AES	Application Enablement Services
ARS	Automatic Route Selection
ASAI	Adjunct Switch Applications Interface
ASG	Access Security Gateway
AVP	Avaya Voice Portal
AWOH	Administered WithOut Hardware
BA	Bridge Appearance
BCMS	Basic Call Management System
BRI	Basic Rate Interface
BSR	Best Service Routing
BTD	Busy Tone Disconnect
CDR	Call Detail Record
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 Connection/Connectivity
COR	Class Of Restriction
CPU	Central Processing Unit
CPN	Calling Party Number
CSP	Cellular Service Provider
CTI	Computer Telephony Integration
DCP	Digital Communications Protocol
DCS	Distributed Communication System
DECT	Digitally Enhanced Cordless Telecommunications

DMCC	Device Media call Control
DPT	Dial Plan Transparency
DSP	Digital Signal Processors
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
EMU	Enterprise Mobility User
ESS	Enterprise Survivable Server
ETSI	European Telecommunications Standards Institute
EWT	Expected Wait Time
FAC	Feature Access Code
FNE	Feature Name Extension
GUI	Graphical User Interface
HDX	A Polycom high definition video room system
HEMU	Home Enterprise Mobility User
IGAR	Inter-Gateway Alternate Routing
IMS	Integrated Management Suite
INADS	Initialization and Administration System
IP	Internet Protocol
IPSI	Internet Protocol Server Interface
ISDN	Integrated Services Digital Network
ISG	Integrated Services Gateway
ITN	Internal Trunk Number
IVR	Interactive Voice Response
J24	Avaya Digital Terminal for Japan
LAN	Local Area Network
LAI	Look Ahead Interflow
LAR	Look Ahead Routing
LED	Light Emitting Diode
LSP	Local Survivable Processor
OPTIM	Off-Premise Telephony Integration with MultiVantage
MG	Media Gateway
MGC	Media Gateway Controller
MM	Modular Messaging
MIB	Management Information Base

MOH	Music on Hold
MPC	Maintenance Processor Complex
MSA	MultiSite Administration
MST	Message Sequence Testing
MTA	Message Tracer Analysis
NCR	Network Call Redirection
OPS	Off-Premise Station
PAI	P-Asserted-Identity
PAM	Pluggable Authentication Modules
PBX	Private Branch Exchange
PCOL	Personal Central Office Line
PE	Processor Ethernet
PNC	Port Network Connectivity
PRI	Primary Rate Interface
PSA	Personal Station Access
PSTN	Public Switched Telephone Network
PCD	Packet Control Driver
QSIG	International Standard for inter-PBX feature transparency at the Q reference point
RDTT	Reliable Data Transport Tool
ROIF	Redirect On IP Failure
RMB	Remote Maintenance Board
RMX	A Polycom media conferencing platform, used by CM as a video and audio bridge
RPM	RedHat Package Manager
RTP	Real-Time Protocol
SAC	Send All Calls
SAT	System Access Terminal
SBA	Simulated Bridge Appearance
SBC	Separation of Bearer and Signaling
SBS	Separation of Bearer and Signaling
SES	SIP Enablement Services
SIP	Session Initiation Protocol
SDP	Session Description Protocol
SNMP	Simple Network Management Protocol
SMI	System Management Interface

SVNS	Simple Voice Network Statistics
TAC	Trunk Access Code
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
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
USNI	United States Network Interface
USB	Universal Serial Bus
VAL	Voice Announcement Over LAN circuit pack containing one hour of storage
VALU	Value-Added
VDN	Vector Directory Number
VOA	VDN of origin Announcement
VEMU	Visitor Enterprise Mobility User
VLAN	Virtual Local Area Network
VSX	A Polycom standard definition video room system
WAN	Wide Area Network