Avaya Aura™ Communication Manager 5.2 SP #3 Release Notes

Issue 1
October 12, 2009

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:

http://www.avaya.com/support

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.

European Union Declarations of Conformity

CE

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:

http://www.avaya.com/support

Trademarks

Downloading documents

For the most current versions of documentation, see the Avaya Support Web site:

http://www.avaya.com/support

Avaya support

Avaya provides a telephone number for you to use to report problems or to ask questions about your product. The support telephone number is 1-800-242-2121 in the United States. For additional support telephone numbers, see the Avaya Support Web site:

http://www.avaya.com/support

Contents

Changes delivered to Communication Manager 5.2 SP #3

Communication Manager 5.2 SP #3 Release Notes

The **Communication Manager** service packs are cumulative and changes in **Communication Manager** 5.2 SP#0, SP#1, SP#2, and SP#2.01 are included in **Communication Manager** 5.2 SP#3. The changes delivered to **Communication Manager** 5.2 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.

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 SP #2

Enhancement	Keywords	Workaround
Previously the "Force Phones and Gateways to Active **LSP**s" field on the system-parameter ip-options form was not administrable by customer login IDs.	083408	
When the duplication link went down, a major alarm was logged immediately without any resolution to retire the alarm.	091287	

Problems fixed in Communication Manager 5.2 SP #0

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

Table 2: Fixes delivered to Communication Manager 5.2 SP #0 ***1 of 2***

Problem	Keywords	Workaround
When a call over trunks was forwarded to the Bridge Appearance (**BA**) of an Administration Without Hardware (**AWOH**) station, it did not term on to the Toshiba **SIP** Phone having the **BA**.	091238	
Whenever calls over a trunk were transferred locally to the Bridge Appearance (**BA**) of an Administration Without Hardware (**AWOH**) station, or a local call to **BA** was transferred over a trunk, the line 1 display on the party having the **BA** goes blank after transfer is complete.It should have shown the connected party's number instead.	091258	
On server type S8730 the command `hardware_info` did not display any information about the hard disk drives.	091284	
In response to a particular error condition for a **SIP** call, **Communication Manager** did not clear the failed call correctly, resulting in a memory-access error that could lead to a system restart.	091288	
Under certain conditions, an internal **Communication Manager** error may result in a system restart.	091292	
Look Ahead Routing not invoked when primary trunk disconnected or on system busy and call is on coverage on that primary trunk.	091322	
After a server interchange on a Processor Ethernet for Duplicated Servers system, **Communication Manager** could experience an extra system restart.	091341	
Under very high system traffic conditions a system reset (cold-2) could happen.	091342	
With the use of Shared Mapping feature, under certain circumstances when calling from cell phone to any desk set, calling party will not be able to see caller's desk set name and number instead it sees caller's cell phone number.	091357	
On rare occasions when the "Force Phones and Gateways to Active LSPs" field is marked 'y', a change in **LSP** status can result in a warm start.	091359	
		1 of 2

Table 2: Fixes delivered to Communication Manager 5.2 SP #0 ***2 of 2***

Problem	Keywords	Workaround
SIP signaling groups may not have a listen socket established leading to **SIP** trunking failures.	091362	
After a server interchange on a Processor Ethernet for Duplicated Servers system, **Communication Manager** could experience an extra system restart.	091379	
After spontaneous server interchange the 'status socket-usage' could report incorrect socket counts.	091383	
Under certain scenarios, if a **SIP** endpoint was placed on hold, then taken off hold, its talk path would not be restored.	091443	
Non-encrpyted Media Gateway could not auto fall back to the main server's Processor Ethernet interface from Local Surviable Processor (**LSP**) or Enterprise Surviable Processor (**ESS**).	091461	
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Problems fixed in Communication Manager 5.2 SP #1

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

Table 3: Fixes delivered to Communication Manager 5.2 SP #1 ***1 of 3***

Problem	Keywords	Workaround
At the time of transfer recall, the transferring party showed line2 blank. To reproduce this problem the initial call should be made over **SIP** trunk and Toshiba **SIP** phones with Unicode name administered should be used.	090914	
Server interchange with **SIP** call traffic caused a segmentation fault.	090931	
Station A on switch 1 called over a **SIP** trunk to station B on switch 2. If station B was a **SIP** phone and transferred the call to another **SIP** phone on switch 2 the call had no *talkpath* and was dropped.	091051	
Communication Manager did not support **SIP** messages that contained **SDP** with no media lines.	091123	
		1 of 3

Table 3: Fixes delivered to Communication Manager 5.2 SP #1 ***2 of 3***

Problem	**Keywords**	**Workaround**
Potential system restart under high traffic with network outages and **IP** phone re-registrations.	091269	
Only one power supply was reported by the command '`hardware_info`' and on the maintenance web page '`Display Configuration`' for an S8510 server which is equipped with two power supplies.	091376	
When configuring SW dup on a server with a mix of 1GB/s and 100MB/s Ethernet cards (for example, S8720), the check to block assigning the dup link to anything less than 1GB/s Ethernet interface did not work.	091380	
In case of UPDATE being rejected by farEnd by sending 405 (Method Not Allowed), did not send UPDATE again for that dialog. Instead for target refresh sent ReInvite.	091434	
When **Communication Manager** created an outgoing INVITE message and there was no P-Charging-Vector available from an incoming INVITE, then **Communication Manager** created a P-Charging-Vector consisting of several identifiers to be unique. One of these identifieres was the own **IP** address. When the outgoing INVITE was routed to a public network, the **SBC** or other **SIP** entities passed on the unchanged P-Charging-Vector. By that the private IP-Address included in the P-Charging-Vector was visible in the public network even when all other private **IP** addresses were filtered out by the **SBC**. This MR fixes the problem by removing the **IP** address from the P-Charging-Vector (remaining part remains unique).	091447	
When **ESS** or **LSP** was active, the trunks in **IP** signaling groups could sometimes show an incorrect service state of in-service/idle when they should actually be out of service.	091449	
On Duplex Main Servers, the **IPSI**(s) associated with the customer **LAN** did not come back into service after upgrading to **Communication Manager** 5.2.	091498	Execute the "`cnc on`" BASH command.
Under certain circumstances involving a **SIP** call, **Communication Manager** could experience a memory-access error, possibly causing a system restart.	091646	
The system could lock up when attempting to take core files for an unexpected restart. The Avaya code was changed to prevent the lockup.	091647	
		2 of 3

Table 3: Fixes delivered to Communication Manager 5.2 SP #1 ***3 of 3***

Problem	Keywords	Workaround
When using the "Configure Server -> Set Modem Interface" on a S8400 the PPP addresses were not set correctly on the Maintenance Processor Complex (**MPC**). Also, the "Configure MPC" (S8400) / "Configure RMB" (S8500) page did not set the "Reserved (Services Future Use)" Ethernet port **IP** addressing correctly.	091648	
Communication Manager could experience a system restart with H.323 trunk administered.	091662	
Issue associated with the following keyword was also fixed in **Communication Manager** 5.2 SP #1: 091643		
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Problems fixed in Communication Manager 5.2 SP #2

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

Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***1 of 11***

Problem	Keywords	Workaround
When a **Communication Manager** user dialed an extension on non-Avaya system using an H.323 trunk, then sometimes the call failed. **Note:** This fix, along with the fix for 091815 changes DTMF event behavior. DTMF events on IP trunks no longer default to using Q.931/H.225 INFO messages with keypad information elements to send DTMF information. With non Avaya equipment, **Communication Manager** now opens H.245, or if H.245 is already open, **Communication Manager** sends the DTMF information as an H.245 alphanumeric string or an H.245 tone event, depending on what the non Avaya equipment has advertised for capabilities. This could require administration changes in **Communication Manager**.	081214	
When the Digital Loss Group field on the trunk-group form contained an inappropriate setting (for example, a digital station loss group was specified for a digital trunk group) then features like Inter-Gateway Alternate Routing did not work as expected. A warning message will now be given to the user if the values entered in the Digital Loss Group field or in the Analog Loss group are not appropriate for the administered trunk group type.	083031	
An entry could get added in the wrong sort position on the 'tandem-calling-party-num' or 'calling-party-num-conv' form and then the entry could not be removed. The error "`Identifier not assigned`" was displayed.	083103	
		1 of 11

Table 4: Fixes delivered to Communication Manager 5.2 SP #2 *2 of 11*

Problem	Keywords	Workaround
Dial Plan Transparency (**DPT**) calls failed in the following case: a) Look-Ahead Routing (**LAR**) was enabled on the route pattern set up to handle DPT/IGAR trunk calls. b) The calling phone was a **DCP** or analog phone (that is, not H.323 or **SIP**). Also, both **IGAR** and **DPT** calls failed in the following case: a) Look-Ahead Routing (**LAR**) was enabled on the route pattern set up to handle DPT/IGAR trunk calls. b) The call was rerouted using a later route pattern preference, because the initial **DPT/IGAR** call failed with an **ISDN** Cause value that triggers **LAR**.	083190	
VDN override rules with **ASAI** were not applied to internal calls. This resulted in inconsistent behavior between internal and external calls when the features 'Allow **VDN** Override' and '**VDN** Override for **ISDN** Trunk ASAI Messages' were enabled.	083531	
Incorrect **DSP** Region measurements could be reported in the list meas ip dsp region reports.	083649	
When call was made to a Group page extension with a station Administered without Hardware as its member, then a delay of at least seven seconds was observed in getting confirmation tone.	083691	
On receiving H323 Facility Message with H.245 Socket information, Avaya **Communication Manager** opened the H.245 socket even when the facility reason was other than start H.245.	083747	
Dial Plan Transparency feature was invoked towards an unplugged IP phone causing improper trunk usage.	083845	
When EC500 set dial the idle call appearance **FNE** (Feature Name Extension) and then dialed an external number, **ASAI** did not report the called number in the Alerting and Connect events.	090016	
Both the stations in the call showed "CONFERENCE" after the transfer was completed. This problem was visible when the transferee station was not an off-PBX telephone integration & mobility (**OPTIM**) station.	090036	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***3 of 11***

Problem	**Keywords**	**Workaround**
Hunt group had only one member (**SIP** station A) and a coverage to an answer group if busy / no answer. Station A was on a call on call appearance 1. There was an incoming call to the hunt group, the problem was the call was ringing on the 2nd call appearance instead of going to the coverage answer group.	090123	
ISDN call setup retried as a result of glare conditions failed if Explicit Call Transfer or Two B-Channel Transfer supplementary service was active on the call. This error occurred only if glare happened on a call which was setup due to vector ~r route-to step.	090130	
Music was not played to the calling station when the Look-Ahead Interflow (**LAI**) checks were performed on the **Communication Manager**.	090140	
When the **PSTN** did not send the calling party number and the replacement string was configured for restricted numbers, the display at called party side (a station that is listed in a vector of a vector directory number) did not show the replacement string.	090196	
If a call to Home Enterprise Mobility User (**HEMU**) was answered on Visitor Enterprise Mobility User (**VEMU**), call pickup lamp on Home Enterprise Mobility User (**HEMU**) pickup group members kept flashing.	090209	
When call from second call appearance on **IP** station placed over **IP** trunk, which had early media and **AES** encryption enabled, the first call was fine but the second call was garbled.	090280	
Made a call to a station that was bridged on another station and answered the call on the bridged station. Transferred the call to another station. Call got dropped.	090303	
When calls were made to Vector Directory Numbers (**VDN**) which had **VDN** Origination of Announcement (**VOA**), were answered, the line four display on the station displayed "date and time" instead of "To " This problem was specific for "Avaya Digital Terminal for Japan" (**J24**) sets and would not be visible if "Idle Appearance Preference" field on the station form is set to "n".	090350	
Calls did not go to the EC500 when the Media Gateway to which the desk phone was connected was unregistered.	090472	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 *4 of 11*

Problem	Keywords	Workaround
There was neither a ring back tone nor a voice path if vu-stat feature was active on the phone and long stream of digits was dialled to originate a call.	090519	Disable vu-stat feature on the phone.
In the case of a **SIP** privacy call, if the calling party number was restricted, the calling party number was not stored in the Call Detail Record.	090544	
The customer may see intermittent failures on backups to Compact FLASH cards on S8400 systems when the card timing was outside of the manufacturer's specifications. Timing was modified to increase the window significantly beyond the specification reducing the possibility of failures.	090556	
As per RFC 3261, port in **URI** was disallowed for From/To header.	090580	
A call was made to a station A having EC500 feature enabled, this call is answered by the off-pbx extension which was mapped to station A via EC500 feature. Now when user presses any digit on this off-pbx station, caller doesn't receive **DTMF**.	090633	
This allows phones with a call appearance that is in CA_WAIT_ORIG to originate a call on that appearance. This will prevent that appearance from being stuck and considered busy by the software.	090655	
No incoming call log entry was made for the Expert Agent Selection (**EAS**) agent if that **EAS** agent's "auto answer" mode was configured to either "acd" or "all".	090662	Don’t configure “auto-answer” mode to “acd” or “all” for an agent with Expert Agent Selection.
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***5 of 11***

Problem	Keywords	Workaround
Faxes were failing when **Communication Manager** received a fax re-INVITE with a=inactive just prior to requesting a switch to T38 fax. Now, **Communication Manager** responds with 200 OK, a=inactive, and null IP address/port and does not begin a transition to fax until receiving a re-INVITE with a=sendrcv.	090687	Do not send a fax re-INVITE with a=inactive prior to requesting a switch to T38 fax.
If a queue button was pressed on attendant immediately after dialing any single digit and continue dialing remaining digits, the displays shows digits interspersed.	090740	
Agent coming out of aux to take call did not get indication of **VDN** where call was in queue.	090742	
When a user logged into the **SAT** (System Access Terminal) through the TN799 (**CLAN**) board and the password had expired, the user was not prompted to change the password.	090745	
When "list trace vector" displays a route-to command with a collected digit of "0" or "#", the character "a" (10) or "c" (12) is actually displayed instead.	090750	
When **IP** phone on **Communication Manager** called **DECT** phone via **QSIG** trunk and **DECT** phone happened to be switched-off, call got dropped instead of providing any feedback to internal caller.	090780	
When there was a hold recall on the call and caller dropped the call, the station remained off-hook.	090788	
Busy out on H.323 trunks in certain intermediate call states was not allowed. The busyout command, in this case, failed.	090793	
Remote coverage calls were dropped if the covering station was an x-ported analog station. The problem was only seen if "Don't Answer Criteria For Logged Off **IP/PSA/TTI** Stations" was set to 'y' on page 3 of the system-parameters features form and "Maintain **SBA** At Principal?" set to 'y' on system-parameters coverage-forwarding form.	090801	
If two or more Busy Tone Disconnect (**BTD**) trunks were involved in a meet-me conference, those **BTD** trunks which joined the call after the first **BTD** trunk, were not disconnected when the caller dropped.	090813	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***6 of 11***

Problem	Keywords	Workaround
When administering the dialplan at the **SAT** (System Access Terminal) with certain dialplans, the customer would see the following error message when making any changes to the public-unknown-numbering form form: "Ext code inconsistent with dialplan". which would block them from making the changes they wanted to make.	090814	
Long hold recall alert was not working if Single Step Conference or Service Observer involves in call.	090818	
When a call which comes on a Vector Directory Number (**VDN**) was covered and went to the second coverage point when the first coverage point did not answer, with a coverage answer group as the second coverage point, the station answering the covered call displayed "c" on line four of its display instead of **VDN** name. This problem was specific to "Avaya Digital Terminal for Japan" (**J24**) sets.	090857	
This problem affects all servers. Previously, the **ISG** would crash in the pacer service software. Now, the **ISG** will verify the pointers are valid before executing the pacer service software. An error will be logged if, the **ISG** finds any invalid pointers.	090872	
Call progress tones may not be heard when using H.323 overlap sending/receiving trunks.	090876	
CallMaster V or 64xx stations did not clear the display when it was on a call with headset and transferred the call.	090883	
Error message was displayed after "display internal-data sta-port XXXXXX" on a **IP** station port, or H.323 **LAN** port.	090899	
If the **VEMU** (Visitor Enterprise Mobility User) called another station on the visitor switch and that station transfered or conferenced the call, the call was dropped after a few minutes.	090932	
When Call to prime is tranfered to **VDN**/HUNT, the display on transfered party was showing calling party's information.	091025	
When calls made to Vector Directory Number (**VDN**), which is routed to an Administration Without Hardware (**AWOH**) station, is covered to a coverage answer group, the line four display on the stations in the coverage answer group showed "c" instead of "date and time". This would occur when "Idle Appearance Preference" is enabled on the stations in the coverage answer group. This problem would be specific for "Avaya Digital Terminal for Japan" (**J24**) sets.	091029	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***7 of 11***

Problem	**Keywords**	**Workaround**
Abbreviated dialing having too many digits in the dialed string used to cause **PCD** (Packet Control Driver) congestion.	091033	
System resets can occur on **Communication Managers** using the **ASAI** feature.	091035	
Intermittently, certain button pushes (like serv-obs) could be incorrectly denied.	091065	Remove the service observing port and add it back.
Under certain internal conditions **Communication Manager** slowed down during Automatic Call Distribution(**ACD**) calls.	091066	
The Call Pickup feature had a special algorithm to determine which call was to be picked up next. The pickup display was updated to reflect any changes to the next call to be picked up. The display was not updating properly in case of Enhanced Call Pickup alerting.	091118	
The system restarted when an **ISDN BRI** endpoint was connected to **Communication Manager** and the endpoint sent an **ISDN** message containing an information element with an invalid length field.	091131	
Under certain circumstances, a softphone user in 'telecommuter' mode could hear distorted dialtone and voice when the administered "callback number" on the softphone was routed to a shuffable **SIP** trunk.	091149	
The "group-sel" button on the attendant console did not work. After pushing the "group-sel" button, you could not dial the group digits successfully.	091177	
Data for the g3trunksta **MIB** group displayed garbage values when a walk was performed on the g3mib.	091186	
Call Forwarded FNU INVITE failed when Fast Connect on Origination field on off-pbx-telephone configuration-set form was set to 'y'.	091204	
Display on bridge appearance of a 2420 digital station was not cleared when it had two bridge-appearances for two different principal stations wherein the first call was answered by the principal station and the second call to another principal station was dropped by the originator before answering.	091215	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***8 of 11***

Problem	Keywords	Workaround
For a **TSP** station, when the auto-call-back alert was received over a **SIP** trunk and the administered name for the calling party was Unicode, then the line-2 display was blank. This occurred always when the trunk between the 2 **Communication Managers** was a **SIP** trunk and the **TSP** stations had Native Unicode names administered.	091331	Administer Name-1 values.
When Toshiba SIP Phone (**TSP**)-1 called **TSP**-2 over a Session Initiation Protocol (**SIP**) trunk and if **TSP**-2 had call forwarded to **TSP**-3 over a **SIP** trunk, as **TSP**-3 rang, display on **TSP**-1 showed name and number of **TSP**-2 instead of **TSP**-3.	091332	
When a trunk call was blind transferred over the trunk again to an Administration Without Hardware (**AWOH**) station, the station having the Bridge Appearance (**BA**) of that **AWOH** station displayed the information of the transferring party even after the transfer was complete. It should have displayed the information of the other connected party. This problem is specific for Avaya Digital Terminal for Japan (**J24**) stations and would only occur if the "Bridged Idle Line Preference" field on the station form is set to "y".	091335	
When Attendant-1 transferred the call to Attendant-2 and canceled it using 'cancel' button, Attendant-2 kept beeping though the call was canceled.	091349	
During a conference call, phone display garbled if it dropped last added party which was across **SIP** trunk to other **Communication Manager**. This problem was specific to "Avaya Digital Terminal for Japan" (2420J) stations.	091352	
A general check was there for sip_max_forwards, which should be only applicable for **SIP/OPTIM** originated calls.	091411	
The calling party name was not displayed on the principal station in the case of an incoming **USNI** (United States Network Interface) trunk call if **SAC** (Send All Calls) was activated.	091416	
Changing ping parameters on page one of the "system-parameters ip- options" form on an S8300 causes invalid TTR-LEV alarms to appear.	091427	
On all the list measurements ipserver-interface [hourly|summary] reports, sometimes the Up-link and Down-link Throughput values were out of range and displayed incorrect data.	091517	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***9 of 11***

Problem	Keywords	Workaround
In case where **SAC** (Send-All-Calls) for remote was administerd first and then **SAC** for prime, when **SAC** for prime was activated, **SAC LED** for bridge was on.	091527	Administer a self-**SAC** button before remote-**SAC** button.
When third party Send All Calls (**SAC**) by a Toshiba **SIP** Phone (**TSP**) was denied, the primary appearance of the **TSP** was reflecting the feature denial and was getting stuck.	091528	
When Station A called Station B over a Distributed Communication System (**DCS**) trunk, and the call covered to a **SIP** Modular Messaging system over a **SIP** trunk on no-answer at Station B, Station A received a non-integrated greeting.	091539	
The system could lock up when attempting to take core files for an unexpected restart. The Avaya code was changed to prevent the lockup.	091543	
In case of call redirection, the new INVITE should have proper request **URI**, with Coverage Of Calls Redirected Off-Net enabled or disabled.	091549	
When using the "Configure Server -> Set Modem Interface" on a S8400 the PPP addresses aren't set correctly on the Maintenance Processor Complex (**MPC**) Also, the "Configure **MPC**" (S8400) / "Configure **RMB**" (S8500) page doesn't set the "Reserved (Services Future Use)" Ethernet port **IP** addressing correctly.	091582	
When an external call was made to a Busy **IP DECT** station, caller was getting reorder tone instead of getting busy tone.	091586	
Under certain circumstances involving a **SIP** call, **Communication Manager** could experience a memory-access error, possibly causing a system restart.	091592	
Communication Manager could experience a system restart with H323 trunk administered.	091602	
For duplicated **Communication Manager** servers employing "software duplication" with an encrypted duplication link, the active server may reset when the standby server is stopped, started, reset, busied out or released.	091604	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***10 of 11***

Problem	Keywords	Workaround
Video calls to the Polycom RSS2000 may fail in unpredictable ways; they may get no video, or simply drop. The behavior is dependent on the bandwidth requested for the call, the endpoint calling, etc. This fix is required for use of the RSS2000. Other third-party video devices may be affected if they have not been part of a development collaboration with Avaya. The Polycom **RMX**, **MGC**, PathNavigator, **CMA**, **HDX**, **VSX** are all unaffected; as are Codian and Cisco devices.	091626	
Server interchanges caused by a network outage with phones connected via PROCR may lead to incorrect socket counts and failure to re-register.	091649	
Calls fail to conference after covering and routing from a **VDN** to a valid extension.	091668	
Media Gateway (**MG**) could get removed after the server interchange if this **MG** registered to the Processor Ethernet (**PE**) interface.	091674	
When Toshiba **SIP** Phone (**TSP**)-1 called **TSP**-2 over a Session Initiation Protocol (**SIP**) trunk and if **TSP**-2 had Send-all-calls activated to **TSP**-3 over a **SIP** trunk, as **TSP**-3 rang, display on **TSP**-1 showed name and number of **TSP**-2 instead of **TSP**-3.	091681	
IP agent calls were getting dropped in certain scenarios involving high call traffic.	091687	
After un-parking the call, Avaya Digital Terminal for Japan (**J24**) station would show feature access code (**FAC**) along with the station's number for which it was un-parking the call. This was happening intermittently.	091695	
For **Communication Manager** systems utiizing H.248 gateways and an Application Enablement Server, outgoing calls generated via **AES** failed.	091697	
An unexpected reset of MAIN server from running **SAT** command '`disable ess all`' or '`disable ess cluster 1`' would occur if the main server was controlling only Media Gateways but no **IPSI** port networks.	091716	
After a server interchange some phones registered to the Processor Ethernet could not get dial tone.	091753	
On S8300D servers running **Communication Manager** and **SIP** Enablement Services (**SES**) co-resident, Provision could not create 450 **SIP** users in **SIP** Enablement Services.	091773	
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Table 4: Fixes delivered to Communication Manager 5.2 SP #2 ***11 of 11***

Problem	**Keywords**	**Workaround**
Neither a History-Info header nor a **SIP** Diversion header was created for **VDN** redirections for the "route-to number" step (with or without coverage) and the "route-to number" step where the number has ~r for network call redirection.	091870	
Issues associated with the following keywords were also corrected in **Communication Manager** 5.2 SP #2: 091273, 091289, 091516, 091551, 091671, 091701		
		11 of 11

Problems fixed in Communication Manager 5.2 SP #2.01

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

Table 5: Fixes delivered to Communication Manager 5.2 SP #2.01

Problem	**Keywords**	**Workaround**
An Avaya **Communication Manager** user could not login on the IA770 to retrieve voice messages. **Note:** This fix, along with the fix for 081214 changes DTMF event behavior. DTMF events on IP trunks no longer default to using Q.931/H.225 INFO messages with keypad information elements to send DTMF information. With non Avaya equipment, **Communication Manager** now opens H.245, or if H.245 is already open, **Communication Manager** sends the DTMF information as an H.245 alphanumeric string or an H.245 tone event, depending on what the non Avaya equipment has advertised for capabilities. This could require administration changes in **Communication Manager**.	091815 092571	

Problems fixed in Communication Manager 5.2 SP #3

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

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***1 of 12***

Problem	Keywords	Workaround
Sometimes the `list measurements blockage pn today-peak/yesterday-peak/last-hour` command showed incorrect data for 'Time Division Multiplexed (**TDM**) Usage' while running on a idle switch.	073919	
A customer had an Octel Voicemail adjunct in the coverage path and a station A on switch A that had its calls forwarded to station B on switch B heard both ring back tone as well as Voice mail greeting if the two switches were connected over a **QSIG** trunk and had **QSIG VALU** set to yes.	081964	Turn off the **QSIG VALU** field.
If Redirect on OPTIM Failure (ROOF) occured for a call to a non-**ACD** hunt group, the **Communication Manager** server was going into overload.	083527	
PSA users were being blocked and denial event 1098 (**TTI** merge/unmerge failed) was being seen when the customer failed to complete the `duplicate station` or `duplicate agent-loginID` commands at the **SAT** (System Access Terminal). The **SAT** was displaying the list output of the stations or agents it added after the submit key was pressed. If the "Next Page" key or "Cancel" key was not pressed to complete the command, **PSA** users would be blocked.	090616	Complete the list of `duplicate station` or `duplicate agent-loginID` commands by either pressing the "Next Page" key or canceling the command using the "Cancel" key.
If a call-center agent using an H.323 endpoint or softphone was offered a call, but their endpoint was not configured to support a compatible audio codec (per the ip-codec-set admin forms), the call could not be connected, and it would not be offered to any other agent (possibly resulting in a `calls in queue, agents available` condition).	090668	
Leave word calling did not work for mixed length dial-plan connected via **DCS** trunk.	090683	
		1 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***2 of 12***

Problem	**Keywords**	**Workaround**
When an Avaya H.323 **IP** phone called over an H.323 **IP** trunk to a Cisco phone controlled by a Cisco Call/UC Manager, and the Cisco phone then put the call on hold, music-on-hold was successfully applied by the Cisco end and heard by the Avaya phone. When the Cisco phone unheld the call, talkpath was not restored.	090823	
For 4624 **IP** sets, the firmware release field on the "status station " and "list registered ip-stations" forms was not correct.	090965	
A team button was configured for a station. A call is picked using this team button. Transfer of this call to the voice mail server failed.	091009	
Abbreviated dialing containing ~w fails where dialing string contained remote access extension and authorization codes in it.	091032	
SIP service link did not shuffle back to direct **IP** after agent unhold a call.	091040	
Previously, there where race conditions where the drop of a domain controlled party occurred between the alerting of a second domain controlled party and the connection of the second domain controlled party that caused the **Communication Manager** to fail to respond to certain 3PCC commands from the adjunct application. Now, if a domain controlled party drops at any time it will be handled properly and **Communication Manager** will continue to respond to 3PCC commands.	091044	
		2 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***3 of 12***

Problem	Keywords	Workaround
Calling/called parties heard an unexpected beep after the call was answered in the following cases: - Called a vector that routed to a variable-length **AAR**/**ARS** number. - If vector step had cov=y, worked fine with no beep. - If vector step had cov=n, heard a beep. **Note:** A few customers may have entered a final # in route-to-number steps to keep the step from hanging if the route-to-number was variable length (for example, the **AAR** / **ARS** minimum length was less than the maximum length). With this MR change, this final # is no longer needed, and will result in a "beep" played after the call is answered. Any customer with a final # in a route-to-number step should remove it after getting this MR change.	091247	
If call was answered on Bridge Appearance (**BA**) and call park button was pressed twice on that station (**BA**), call appearance button LED on that **BA** remained ON indefinitely.	091333	
DID/Tie/**ISDN**/**SIP** Intercept to Announcement was failing with Separation of Bearer and Signaling (**SBS**) trunks when the caller mis-dialed the number.	091338	
When EC500 user dialed the idle call appearance **FNE** (Feature Name Extension) and then dialed an external number, **ASAI** reported an incomplete called number in the Alerting and Connected events if, - the Digit Handling field on the trunk group form was set to "overlap/overlap". - the field "DTMF over IP" on the H.323 signaling group form was set to "in-band". - the user dialed the digits very slowly.	091395	
When the NICE recording application was used and network disruptions caused media gateways to lose connectivity with the **Communication Manager** server, stations being recorded were left in an Out-of-Service state after the gateways re-registered with **Communication Manager**.	091439	
3 of 12		

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 *4 of 12*

Problem	Keywords	Workaround
The customer may experience dropped calls or no *talkpath* after issuing a **`reset media-gateway`** level one or level three command and the H.248 media gateway was re-registered (for a level one reset) or the H.248 media gateway modules were back in service (for a level three reset).	091442	
PAM security traps were missing logname, uid, euid, tty ruser, rhost, and user information after the FPAgent processed them. The complete trap was displayed in the messages file located in /var/log/messages.	091480	
If an H.323 IP station or a **DCP** station dialed into an Expanded Meet-Me Conferencing Vector Directory Number which routed over H.323 or **SIP** trunks to an external conference bridge, then the party entering the conference did not get cut through to the rest of the conference.	091511	
When an **ESS** became active due to network fragmentation, causing calls between the main and the **ESS** to use the dial-plan transparency feature, some calls to or from the **ESS** location could have experienced a lack of *talkpath* if the **ESS** happened to be controlling a port network configured with DS1 trunks.	091570	
IQ/**CMS** could abort tracking of calls deflected between **Communication Manager** servers by Network Call Redirection (**NCR**) on **SIP** trunks.	091573	
Avaya Voice Portal (**AVP**) endpoints were not getting registered when there were no signaling resources in the connected network regions.	091593	
Conditions: a) An **ASAI** adjunct initiated a call to an OOS station on the same server but in a different Network Region. b) The call invoked Dial Plan Transparency to reach the OOS station. c) The **ASAI** adjunct dropped the call early, before the call was set up. The above case led to a system restart.	091655	
An example file and directory indicated in a logging man page for configuring views were not clearly identified as an example, causing users to think they actually existed on the server.	091666	
		4 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***5 of 12***

Problem	**Keywords**	**Workaround**
The voicemail adjunct reported the wrong calling party number in some "transfer to voicemail" scenarios involving X-ported stations.	091670	
In the case of an H.323 **IP** station with three button modules, one bridge call appearance was configured on the third button module. Then that bridge call appearance was moved to the first button module. When a call was made to that bridge call appearance extension, a blank display was seen and also the call log showed as unavailable.	091692	
When EC500 State was disabled using station form for a station with off-pbx-telephone station-mapping for EC500 application, the first attempt to enable the EC500 using the EC500 button failed. The feature was deactivated again instead of being activated. Pushing the EC500 button again activated the EC500 feature.	091743	Push the button again to activate the feature.
Bridged calls answered by Extended to Cellular (EC500) could not be service observed	091745	
Incoming **SIP** calls from some non-Avaya systems may fail.	091747	
After 96xx **IP** station A took over the extension for another 96xx **IP** station B, and then station B takes back the extension from station A and then the process was repeated a few times, both stations could end up in a locked up state.	091754	
Second page of the duplicate station form failed to open when duplicating XMOBILE station with Mobility Trunk Group of 'ars' and 'aar', Also, duplicate XMOBILE stations failed intermittently with Mobility Trunk Group set to a trunk group number.	091766	
Under circumstances pertaining to a specific method of **SIP** signaling, calls over a **SIP** trunk connected to a Service Provider dropped.	091777	
After a server interchange, **Communication Manager** could experience an extra system restart.	091783	
Under certain internal conditions, the system may reset during normal call operation.	091784 091785	
Under certain internal conditions, **Communication Manager** may reset, impacting call processing.	091786	
		5 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***6 of 12***

Problem	Keywords	Workaround
For an external call coming over an **ISDN** trunk to a local station on Communication Manager, an **ASAI** call recording application would shut down and do a server interchange when the called station was incorrectly identified as "National" instead of "private local number" in the Alerting event.	091796	
Customers monitoring stations with **ASAI** may see a # sign at the end of the Called party number when user classified calls were placed using **TAC** dialing.	091797	
TN2602 Medpro alarms were incorrectly generated on an **LSP** after the **LSP** was upgraded.	091800	
Native Administered 96xx phone A was in a conference call with phone B and phone C. Then phone A pushed the transfer button to dial phone D, D rang. Then phone B dropped from the conference call, and phone A cancelled the transfer call. At this time, phone A could not go back to talk with phone C.	091807	
A user was allowed to press a pickup-group button to pickup a call of a fellow pickup-group member, when that group member was listening to a **VDN** of Origin Announcement (**VOA**) for a Vector Directory Number (**VDN**). That resulted in a situation where the user did hear the **VOA**, but, when connected to the caller, did not have *talkpath* (and the party who originally answered was dropped).	091811	
All the **IP** stations on the server were abruptly rebooting at the same time.	091818	
When the 'ANALOG BUSY AUTO CALLBACK Without Flash?' field was enabled on system-parameters features form, 'Busy Auto Callback without Flash?' field did not appear on station form for callrID stations.	091819	
When an attendant vector had two queue-to attendant/attd-grp steps and the first one failed, a spurious "forward" event was reported to IQ/**CMS**. This could result in the appearance of calls-in-queue with agents available.	091862	
A **SIP** phone with a call from a phone on a gateway that was on hold, could not be unheld and the call was soon dropped.	091863	
		6 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***7 of 12***

Problem	Keywords	Workaround
Customers with a feature access code administered on their switch for PIN Checking for Private Calls Using **ARS** Access Code or PIN Checking for Private Calls Using **AAR** Access Code could not register **IP** H.323 virtual endpoints (that is, **IP** Softphone, **IP** Agent).	091880	
When the station on a call dialed the announcement extension after pressing the no hold conference button, three party conference did not proceed & the phone display showed `Connecting to 403`, where 403 was the announcement extension.	091885	
Executing a **`list trace hunt-group`** command for an Audix hunt group failed to output any records when calls terminated to the group.	091896	
There should not be any core even when the called Number is NULL.	091900	
Port change in 200 OK against display reinvite or session refresh reinvite caused *talkpath* break or call drop.	091923	
The call transfer destination station was showing transferring station's information as connected party information after call was transferred. This problem was observed on 64xx phone only with headset button turned ON.	091926	
In case of Calling ID blocking, cellular service provider originated call from mobile phone was not working.	091936	
When vu-display was enabled and an external call landed on an **IP** station, the starting part of the display on the call appearance button was truncated. This happened for **IP** stations like 4610, 4620, 4621, 4622, 4625, 9620, 9630, 9640, 9650, 1608, 1616 which display the incoming/outgoing name and extension on the call appearance, only when the VuStats feature was enabled.	091953	
When the switch had a 2 port network setup and we did call pick up or call unpark the *talkpath* should be there. If shuffling was enabled, then the call went to direct **IP**.	091976	
		7 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***8 of 12***

Problem	Keywords	Workaround
Preconditions for the error were: There was a hunt group with two stations \`A\` and \`B\` as members. Both members of the hunt group had a team button configured: - Station \`A\` had a team button which pointed to station \`B\` - Station \`B\` had a team button which pointed to station \`A\`. The setting of the feature "Temporary Bridged Appearance on Call Pickup" on page 18 on the "system-parameters features" form must be set to "n". When the hunt group extension was called from a third station then either \`A\`or \`B\` was ringing. In the same time the team button started blinking on the station which was NOT ringing. After pushing team-button twice (!) on this non-ringing station the call was picked-up, but the team-button kept blinking. Even after the call was ended.	091977	Avoid the error: The team button is working properly, when the feature "Temporary Bridged Appearance on Call Pickup" on page 18 on the "system-parameters features"-form is enabled.
Port number shall not be sent as 0 in From header for a **SIP** tandem call scenario	091985	
Avaya Site Administration "Export" vector failed to provide "skill" value for pages 2-6 with vector steps: -"check skill", -"consider skill", -"goto .. if expected-wait for skill"-"goto .. if rolling-asa for skill" " The exported text file contained no value for the fields containing "skill".	091990	
If a H.323 station outgoing direct media call was made to a non-Definity **Communication Manager**, then call got dropped.	092002	
When a call from **PSTN** was transfered to an unregistered station, which was connected to Modular Messaging via **SIP** trunk, a general greeting was played.	092003	
Modifying a **SMI** user profiles name to contain one of the characters '&' and '<' led to make this profile unreadable. This effect could also occur during an upgrade, if one of the existing user profile names contained one of these characters.	092009	
Calls directed to an **IP** Agent with auto answer enabled were being dropped.	092023	
An attendant transferring a call back to the original called **IP** phone resulted in no *talkpath* between caller and transferred to station.	092031	
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Table 6: Fixes delivered to Communication Manager 5.2 SP #3 *9 of 12*

Problem	Keywords	Workaround
SDES SRTP call did not work across **SIP** trunk.	092047	
Analog media modules were receiving inconsistent downlink messages. Customer saw constantly increasing errors `MG_ANA 3840`.	092056	
For an agent in auto-answer mode, after the **VDN** (Vector Directory Number) of Origin Announcement (**VOA**) played and the call connected, the call timer did not start. This occurred always when there was an incoming call to a **VDN** that had a **VOA** and the call routed to an agent that was logged onto a 46xx or 96xx series station.	092063	
Using the list trace TAC command to trace a trunk call that was added to a conference call and then later dropped fails to record the next trunk call for the trunk group. The user must exit and re-execute the command to trace another call.	092067	
Whenever a call was forwarded to a Vector Directory Number (**VDN**), which was routed to an Administration Without Hardware (**AWOH**) Station (P1), the display on the line four of the station having the Bridge Appearance of the **AWOH** station showed "To P1" instead of "To **VDN**". This problem was specific to "Avaya Digital Terminal for Japan" (2420J) stations.	092093 092095	
Whenever a call from **PSTN** was covered to a Vector Directory Number (**VDN**), which was routed to an Administration Without Hardware (**AWOH**) Station (P1), the display on the line four of the station having the Bridge Appearance of the **AWOH** station showed "To P1" instead of "To **VDN**". This problem was specific to "Avaya Digital Terminal for Japan" (2420J) stations.	092096	
In certain circumstances, calls originating on a TOSHIBA **SIP** phone did not get proper display updates on the caller's set when the call was forwarded.	092114	
Vector "route-to" an **ARS/AAR FAC** suspended vector processing.	092125	
A H.323 trunk call between **Communication Manager** and Media Module could cause a seg fault if Media Module replied with zero codec information in faststart reply.	092139	
Sending out the INVITE with the "P-Charging-Vector" header (with icid-value only) even though "IMS Enabled" was turned off, was causing call failure.	092142	
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Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***10 of 12***

Problem	Keywords	Workaround
When a call consisted of two **IP** phones that were directly connected and one station hit hold, the resulting announcement as Music-On-Hold would be sourced based on the initial network region of the first party that initiated the call and not the remaining party that was listening to Music-On-Hold. This could cause additional **IP** resources to be allocated when listening to the **MOH**.	092169	
On **SIP** 96xx phone auto callback button's LED used to remain ON in case of outgoing call was made on a trunk and then tried to invoke auto callback feature.	092176	
When call was placed from a **SIP** originating station to a **SIP** terminating station, the call first went to **TDM** and then **Communication Manager** shuffled the call to connect direct **IP**. While Shuffling **Communication Manager** sent null INVITE to both the end points wherein if one of the end point responded with Request pending (491), then the call should not be dropped and should go to direct **IP**.	092180	
An attendant transferring a call back to the **IP** phone originally called resulted in no alerting on a bridged **IP** phone when Inter-Gateway Alternate Routing (**IGAR**) was triggered between the network region of originally called **IP** phone and the network region of the attendant.	092181	
The entire call was dropped when Integrated Music On Hold was being played and Single Step Conference party hung up the call.	092192	
When the user changed the node-name of a disabled **CLAN** ip-interface, the associated Link information was not updated. This eventually caused two **CLAN**s to have the same **IP** address, which caused phones not to register.	092195	
Under certain circumstances a call between two Motorala phones failed.	092214	
Duplicated TN2602 (Crossfire) Media Processor boards were not getting sent a ’state of health’ update from **Communication Manager**. This could lead to the incorrect board of the pair being active.	092220	
INTERCEPT tone was not played after the timeout interval when an authorization code was required on a trunk and the user did not enter the authorization code.	092227	
		10 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 *11 of 12*

Problem	Keywords	Workaround
User couldn't change 2 different Vector Directory Number (**VDN**)/hunt group's simultaneously when both **VDN**'s/hunt group's share same Computer Telephony Integration (**CTI**) link.	092235	
Station A which covered to a SIP integrated Modular Messaging voice mail adjunct. If station A received a call over a trunk and the call covered to voice mail, then the call was not covering to the voice mailbox of Station A.	092249	
Enable synchronization and disable synchronization commands always returned the following error `Identifier command word(s) omitted; please press HELP`.	092251	
SAT login IDs that have specific vectors administered in their extended-user-profile were unable to see all assigned vectors when using the **`list vector`** command.	092259	
The country-to/from information was incorrect in **CDR** reports for calls made with the Multi-National Location feature enabled.	092273	
SIP phones connected to a Connection Manager with an extended numbering environment (public numbering or private numbering enabled) may encounter some unexpected behavior. These issues may be caused by a new parameter "avext" which is included in some **SIP** messages but only valid in an Avaya Aura™ environment.	092319	
On rare occasions, the Time Slot Record Audit (**TSRA**) may provide some faulty data as shown on the 'status audits cumulative' form.	092322	
If an outgoing R2MFC trunk call was transferred internally, the transfer failed.	092363	
After an R2MFC trunk call was established, if the originator pressed any digit, the **DTMF** tone was not heard at the far end.	092364	
Incoming **SIP** INVITE messages that contained a Replaces header sometimes resulted in failed calls.	092382	
Under certain **SIP** traffic conditions where network errors occurred, **Communication Manager** experienced a reset.	092422	
When an incoming call was transfered by an agent using a third party application over a **SIP** trunk, which had **NCR** (Network coverage redirection) enabled, the Agent application should not fail.	092477	
		11 of 12

Table 6: Fixes delivered to Communication Manager 5.2 SP #3 ***12 of 12***

Problem	Keywords	Workaround
Messaging Vector step did not deactivate vector processing. As a result, any call queued to huntgroups/skills prior to the messaging step queued until the caller was disconnected.	092590	
When MultiSite Administration user attempted to execute a "duplicate station" operation, an internal system error occurred and the operation failed.	092757	
Defensive fix for **SIP** Timer Expiry in case of network outage.	092770	
After receiving **DTMF** tones embedded into an incoming **RTP** streams, **Communication Manager** did not forward these **DTMF** tones over a H.323 trunk.	092775	
Wrong station heard **DTMF** tones when call was initiated using autodial button with ~p and **DTMF** digits.	092835	
If music or an end-to-end signal (for example, button press) was added to a direct **IP** call, then under certain circumstances, neither the music nor the signal would be heard.	092880	
When a call is placed over **SIP** (session initiation protocol) there should be *talkpath* irrespective of the phone type from which or to which the call is placed and also the call should go to direct **IP**.	092883	
SIP transfer call involving multiple port network or gateway should have *talkpath*. It should no way depend on the type of phone to which the transfer was made, from which the transfer was done.	093036	
Issues associated with the following keywords were also corrected in **Communication Manager** 5.2 SP #3: 073528, 081948, 091469, 091577, 091756, 091828, 092079, 092193, 092232.		
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Known problems

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

Table 7: Known problems in Communication Manager 5.2 SP #3 *1 of 7*

Problem	Keywords	Workaround
If the user, from the local phone application, has customized some of their button labels on their EU24 expansion module, and the administrator changes the station type to another station type that supports both customized button labels and an EU24/expansion module, then after the change any customized button labels on the EU24 are deleted and replaced with the default button label for the given button type. **Note:** The custom labels on the primary set/main module are not deleted.	083112	Do not change station type on stations using EU24/ expansion modules.
S8300 platforms do not currently support the `list meas ip dsp region` and `list meas ip dsp gw` commands.	091220	
		1 of 7

Table 7: Known problems in Communication Manager 5.2 SP #3 ***2 of 7***

Problem	Keywords	Workaround
When using web administration tools with **Communication Manager**, if the user switches between servers running different releases and/or loads, in some cases the web pages cannot be displayed or are incorrect. This is especially true when going backwards in releases (for example from **Communication Manager** 4.1 on Server 1 to **Communication Manager** 3.0 on Server 2).	NA	Perform the following steps to clear the web browser cache on the computer: 1. Open the Internet Explorer browser, and select **Tools** > **Internet Options**. 2. In the Temporary Internet files section, click **Delete Cookies**. 3. Click **OK**. 4. In the Temporary Internet files section, click **Delete Files**. 5. Click **OK**. 6. In the History section, click **Clear History**. 7. Click **Yes**.
		2 of 7

Table 7: Known problems in Communication Manager 5.2 SP #3 *3 of 7*

Problem	Keywords	Workaround
S8400 can be used as an Enterprise Survivable Server (**ESS**) with specific limitations. These limitations apply to both the main server and the S8400 **ESS** as shown below. These limits are not enforced by **Communication Manager** software. The sum of the following three components must not be greater than 1500: 1. The number of H.323 Signaling Groups administered on the main server. 2. The number of H.323 Trunk members administered on the main server. 3. The number of H.323 **IP** endpoints that will register with the S8400 **ESS**. The total number of H.323 **IP** endpoints on the system can be larger than the number that will register with the S8400 **ESS** as long as the above limits are met. If the above limits are exceeded and the S8400 **ESS** goes active, it will not be able to register all the **IP** endpoints. The H.323 Signaling Groups and the H.323 Trunk members will first use the administered number of "**IP** User Records" from the pool of 1500. Any remaining **IP** User Records will be used for registering **IP** endpoints. For example, if the main server has a total of 1000 H.323 Signaling Groups and Trunk members, then 500 **IP** stations can register with the S8400 **ESS**. The main server will not experience any limitations due to the S8400 **ESS**.	NA	Use S85xx simplex servers or S87xx server pairs as **ESS** servers on larger systems.
		3 of 7

Table 7: Known problems in Communication Manager 5.2 SP #3 *4 of 7*

Problem	Keywords	Workaround
Systems with duplicated S87xx main servers and TN2312 (**IPSI**) circuit packs on a customer or corporate **LAN** instead of a private network may lose communication with the **IPSI**s after an upgrade to **Communication Manager** 5.2. Systems with **IPSI**s on a corporate **LAN** and S87xx duplicated main servers must have Control Network C (**CNC**) configured prior to the upgrade to **Communication Manager** 5.2 to avoid **IPSI** communication problems. Enabling **CNC** after the upgrade will also restore communication to the **IPSI**s. To check status and enable **CNC** use the following bash commands. These commands can be run by susers group logins such as "craft," "init," "dadmin," or customer defined susers group logins. **`cnc status`** Control Network C has NOT been configured. **`cnc on`** Control Network C has been configured.	NA	
With **Communication Manager** 5.2 the administration of the route calls to agents by skill level preference on the check step can not be done using Avaya **CMS** vector administration (R15 or earlier). **Communication Manager** administration will be required to add check steps with the pref-level parameter and any vectors that contain these steps can not be accessed by **CMS** vector administration.	NA	
When the feature access code for PIN Checking for Private Calls Using **ARS** Access Code is used, the external trunk is not seized and the external extension cannot be reached.	091484	
		4 of 7

Table 7: Known problems in Communication Manager 5.2 SP #3 *5 of 7*

Problem	Keywords	Workaround
Terminal servers used for **CDR** that are connected to duplicated S87xx servers via Processor Ethernet may lose connectivity after a server interchange. The **TCP** software in the terminal servers may not close the socket to the previously active server and open a new socket to the newly active server.	NA	Use the Avaya Reliable Data Transport Tool (**RDTT**) available at the Avaya Support web site.
Customers cannot enable the Simple Voice Network Statistics (**SVNS**) feature for a duplicated TN2602 (Crossfire) pair when the standby Crossfire is administered in the "meas-slection" form. In this case the following error is displayed: `Meas-selection form is not in sync for Voice/Network Statistics feature.`	091377	1. Enable the **SVNS** feature in the "system-parameters ip-options" form: Enable Voice/Network Stats = y 2. Determine which Crossfire board is the active board: status media-processor board <dup_Crossfire_board> 3. Administer the "active" board in the "meas-selection media-processor" form: change meas-selection media-processor 4. Once the active board is entered in the meas-selection form, the feature can be enabled for the dup Crossfire pair on Page 4 of the ip-interface form.
An upgrade from S8710 configurations with **Communication Manager** 2.2, 2.2.1, 2.2.2, 3.0 or 3.0.1 to **Communication Manager** 5.2 will fail.	091522	Prior to the upgrade obtain, install, and activate the pre-upgrade installation patch appropriate for your **Communication Manager** release. The pre-upgrade installation patches can be found at http://support.avaya.com.
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Table 7: Known problems in Communication Manager 5.2 SP #3 ***6 of 7***

Problem	Keywords	Workaround
The Branch Gateway Customer install script associated with the branch Gateway Firmware load 29.22.3 will not provision the G250 Ethernet port and G350 MM314/MM316 ports and the chassis Ethernet **LAN** port into the customer **VLAN** provisioned by the script. Follow the workaround to finish the provisioning on the platform. This problem is resolved in Gateway Firmware Load 29.23.0 and greater.	090271	G350 1. Login to the **CLI** using the "root" login and new password provisioned using the script. 2. Enter the command: **`set port vlan <vlan number>10/3`** 3. If an MM314 is present, enter the command: **`set port vlan <vlan number> 6/1-24`** 4. If an MM316 is present, enter the command: **`set port vlan <vlan number> 6/1-40`** 5. To save the new configuration options programmed, enter the command: **`copy running-config startup-config`** 6. The above has now provisioned all the G350 Ethernet ports into the new vlan and the configuration is saved. G250 1. Login to the **CLI** using the "root" login and new password provisioned using the script. 2. Enter the command: **`set port vlan <vlan number>10/3-10`** 3. To save the new configuration options programmed, enter the command: **`copy running-config startup-config`** 4. The above has now provisioned all the G250 Ethernet ports into the new vlan and the configuration is saved.
		6 of 7

Table 7: Known problems in Communication Manager 5.2 SP #3 *7 of 7*

Problem	Keywords	Workaround
When a **USB** Modem Dial-In connection to a G430 is up and **CPU** utilization is high, the dial-in session might disconnect and further attempts to setup a new dial-in connection all fail. The dial-in session failure usually happens a few hours after the connection was setup. The time until the dial-in session failure depends on the **CPU** load. At 100% **CPU** utilization, the failure occurs within 1 or 2 hours. At 30% **CPU** utilization, this issue is much less frequent.	090291	If the dial-in session disconnects and cannot be setup again, unplug the **USB** modem device from of the G430 gateway and reinsert it.
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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
ACD	Automatic Call Distribution
AES	Application Enablement Services
ARS	Automatic Route Selection
ASAI	Adjunct Switch Applications Interface
AVP	Avaya Voice Portal
AWOH	Administered WithOut Hardware
BA	Bridge Appearance
BRI	Basic Rate Interface
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
CMS	Call Management System
CNC	Control Network C
CPU	Central Processing Unit
CTI	Computer Telephony Integration
DCP	Digital Communications Protocol
DCS	Distributed Communication System
DECT	Digitally Enhanced Cordless Telecommunications
DPT	Dial Plan Transparency
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
ESS	Enterprise Survivable Server
FAC	Feature Access Code
FNE	Feature Name Extension
HDX	A Polycom high definition video room system
HEMU	Home Enterprise Mobility User
IGAR	Inter-Gateway Alternate Routing

IP	Internet Protocol
IPSI	Internet Protocol Server Interface
ISDN	Integrated Services Digital Network
ISG	Integrated Services Gateway
J24	Avaya Digital Terminal for Japan
LAN	Local Area Network
LAI	Look Ahead Interflow
LAR	Look Ahead Routing
LED	Light Emitting Diode
LSP	Local Survivable Processor
OPTIM	Off-Premise Telephony Integration with MultiVantage
MG	Media Gateway
MGC	Media Gateway Controller
MIB	Management Information Base
MOH	Music on Hold
MPC	Maintenance Processor Complex
NCR	Network Call Redirection
PAM	Pluggable Authentication Modules
PE	Processor Ethernet
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
RMB	Remote Maintenance Board
RMX	A Polycom media conferencing platform, used by CM as a video and audio bridge
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
SMI	System Management Interface
SVNS	Simple Voice Network Statistics
TAC	Trunk Access Code
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TSP	Toshiba SIP Phone
TSRA	Time Slot Record Audit
TTI	Terminal Translation Initialization
URI	Uniform Resource Identifier
USNI	United States Network Interface
USB	Universal Serial Bus
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