Avaya Aura™ Communication Manager 5.2.1 Release Notes

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Notice

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Avaya Inc. provides a limited warranty on this product. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language, as well as information regarding support for this product, while under warranty, is available through the Avaya Support Web site:

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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:

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

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

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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 5.2.1

Communication Manager 5.2.1 Release Notes

The **Communication Manager** service packs and releases are cumulative and changes in **Communication Manager** 5.2 SP#0, SP#1, SP#2, SP#2.01, and SP#3 are included in **Communication Manager** 5.2.1. The changes delivered to **Communication Manager** 5.2.1 are grouped as follows:

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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

New features and significant enhancements in **Communication Manager** 5.2.1 are described in the document titled "Avaya Aura™ **Communication Manager** Change Description for Release 5.2.1" which can be found at http://support.avaya.com. The following changes that are new to **Communication Manager** are also included in this release.

Table 1: Enhancements delivered to Communication Manager 5.2.1 *1 of 4*

Enhancement	Keywords	Workaround
This new feature implements a **`clear meas ip dsp`** command. It zeros out **IP DSP** Region, Port Network and Media-Gateway measurment data for ALL Regions, ALL Port Networks and ALL Media-Gateways for the current hour only. **IP DSP** Region, **PN**, and GW data in meas_m from previous hours was unchanged. This new command could be used to help debug **IP** network issues. When changes were made to the network to correct problems, this command could be used to zero out ALL **IP DSP** measurement data for the current hour. At the top of the next hour, the **DSP** measurement reports for the last hour then included only data gathered AFTER the **`clear meas ip dsp`** command was executed.	071315	
The maximum value allowed in the Preferred Minimum Session Refresh Interval field on the **SIP** trunk group form was increased from 1800 seconds (30 minutes) to 64800 seconds (18 hours). The value administered in the Preferred Minimum Session Refresh Interval field was used to delay the sending of the session refresh re-INVITE messages.	082515	
Communication Manager software includes certain third party and open source software packages, including software developed by the **Apache Software Foundation** (http://www.apache.org). **Communication Manager** 5.2.1 includes a file of open source licenses on the software CD. To view the license file, 1. Insert the **Communication Manager 5.2.1 CD** into the CD/DVD drive of a personal computer. 2. Browse the CD content to find and open the file ***`D:\Licenses\3rd-party-licenses.txt`***. This information is only accessible on the **Communication Manager** software CD and is not installed or viewable on the **Communication Manager Server**.	083034	
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Table 1: Enhancements delivered to Communication Manager 5.2.1 ***2 of 4***

Enhancement	Keywords	Workaround

The TestInadsPort Bash command was added to specify the port to use for TestInads.
The commands usage is as follows:
Usage:
`testinadsport [-p [1024-65535]] | [-?]`
`no argument: display`
`-p: sets testinads internal tcp port(1024-65535): default 21111`
`-?: usage (this)`
 	
As stated in the usage if no argument is provided it will display the current value of the internal tcp port that is used between the GMM and the testinads command.
The -p option with no value will set the internal port to a default value of 2111
The -p option with a valid value will set the port to that value.
Valid ports are 1024 - 65535
The -? option will display the usage.
	083389	

If an outgoing **SIP** trunk call loops back to **Communication Manager** through a Service Provider the call could fail.	082455	
From a **Communication Manager SAT** terminal, the customer saw that the **SIP** trunk group that just completed a call redirect (~r) did not drop the trunk immediately. If the customer executed a "`status trunk xx`" command, they saw the status was "in=service/active" and no ports were connected. It took up to 2 minutes for the **Communication Manager** Audit to run and clear/drop the trunk.	082870	
An external call transferred/conferenced locally or over a **QSIG** VALU trunk rings external or internal depending on the value of the field "External Rinnging for Calls with Trunks". The field was on the System Parameter feature form.	083151	
A warning message was not displayed when a user executed a "**reset system**" command. With this enhancement, a warning message was displayed when a user executed a "`reset system`" command, and the user must press ENTER to continue or CANCEL to abort the command. The warning message was only displayed if the new field "Display Warning Prior to System Reset" was enabled on page 1 of the system-parameters maintenance form.	083193	
2 of 4		

Table 1: Enhancements delivered to Communication Manager 5.2.1 ***3 of 4***

Enhancement	Keywords	Workaround
IP trunks often experienced temporary failures. A denial event would have eliminated a lot of the time spent in searching for the exact cause so appropriate denial events were added.	090644	
Conditional Call Extend feature did not apply to OPTIM applications that were associated with one-X Server. These fields should be inaccessible for off-pbx-telephone station-mapping with **PVFMC** application and Dual Mode of **DMX**.	090781	
A new station type (4612CL) needed for the 36xx wireless Polycom stations which would allow those stations to provide a call log feature.	090846	
If "Extension only label for Team button on 96xx H.323 terminals" on system-parameters features form was set to 'y' then only included the team extension (or name of the extension if "Team Btn Display Name" was set to 'y') in the label sent to the phone. Firmware 2.0 or newer was recommended on the phones as they provided a special icon for team buttons. This field did not impact if the label was customized by the endpoint.	091013	
SIP calls that did not have packet intervals specified by the far end would be set up with what was administered on the ip-codec-set form instead of the **SIP RFC** default values.	091192	
list measurements ip dsp-resource commands were not supported on S8300x platforms.	091220	
The "Shutdown Server" **SMI** page gave feedback to the user occasionally because sometimes after the shutdown server was requested it happened so quickly that the page could not complete before it lost connectivity to the server. Users of this page saw a message saying that a server shutdown has been requested.	091350	
The field Incoming Dialog Loopbacks appeared on the Signaling-Group form for the **SIP** signaling group type. This field allowed one of the following two values, "allow" or "eliminate". When this field was set to "allow" then calls made via trunk groups associated with the **SIP** signaling group were allowed to terminate on the originating **Communication Manager** server. The field default was "eliminate" which resulted in the calls being treated as they were prior to this enhancement.	091378	
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Table 1: Enhancements delivered to Communication Manager 5.2.1 ***4 of 4***

Enhancement	Keywords	Workaround
Customers using Simple Voice Network Status with duplicated TN2602 or TN2302 boards, where one of the boards did not support this feature, saw error type 3841 when they ran the display errors command.	091548	
On systems using Audix (**CMM**/**SAM**) transfers of very large Audix images may cause the sftp connection to timeout.	091967	
A new field, "**QSIG/SIP** Diverted Calls Follow Diverted to Party's Coverage Path", was added to page 1 of the "system-parameters coverage-forwarding" form.	092421 092277 092512	
Added a new feature access code (**FAC**) field "Message Sequence Trace (**MST**) Disable" on page 3 of the features form. When user dialed this FAC the **MST** trace was diasabled if the "DEBUG BUTTON" field on page one of the **MST** form was set to yes.	093032	
After an abrupt interruption of power (that is, sudden powerloss on the server), the OS startup scripts forced any necessary repairs to the file system if any problems did arise.	090236	
When the Split Registration Prevention Feature was on, on rare occasions immediately after a warm start, usually due to an interchange, network regions did not become auto-disabled or auto-enabled when a **LSP** became active or inactive.	090925	
If "Extension only label for Team button on 96xx H.323 terminals" on system-parameters features form was set to 'y' then only included the team extension (or name of the extension if "Team Btn Display Name" is set to 'y') in the label sent to the phone. Firmware 2.0 or newer was recommended on the phones as they provided a special icon for team buttons.	091414	
The field "Use Trunk **COR** for Outgoing Trunk Disconnect? " on page 6 of the system-parameters features form is being renamed to "Use Trunk **COR** for Outgoing Trunk Disconnect/ Alert? ". The field "Outgoing Trunk Disconnect Timer (minutes): " on page 2 of the Class of Restriction (**COR**) form is being moved to page 3 of the cor form.	091916	
Processor channels on PROCR went down and were slow to recover after a server interchange. For **CMS** adjuncts, this caused a "pump up", which can take many minutes.	092292	
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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	
		2 of 2

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	
Potential system restart under high traffic with network outages and **IP** phone re-registrations.	091269	
		1 of 3

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

Problem	Keywords	Workaround
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	
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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	
		3 of 11

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		
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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	
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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	
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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	
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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	
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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	
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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	
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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	
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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	
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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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Problems fixed in Communication Manager 5.2.1

The following fixes were delivered to **Communication Manager** 5.2.1 in addition to the service pack fixes detailed in previous sections.

Table 7: Fixes delivered to Communication Manager 5.2.1 *1 of 24*

Problem	Keywords	Workaround
Service observer (**SO**) station received error beep tones and the LED was blinking while **SO** station was trying to activate **SO** feature for a local extension using shortcut dialing.	073576	
If AUX_WORK button was activated on 9600 **IP** phone and an incoming call to that phone was directed to coverage using "To Vmail" button, the AUX-WORK button got deactivated.	074438	
A Nice recorder failed to record a port because the **Communication Manager** thought the phone was busy when it was not.	081710	
Some outgoing calls over **ISDN** trunks failed if the trunks being used were not set to IDLE when the previous call on that trunk was dropped.	083941	
Under rare circumstances, **Communication Manager** could encounter an internal memory-management error and experience a system restart.	090062	
		1 of 24

Table 7: Fixes delivered to Communication Manager 5.2.1 ***2 of 24***

Problem	Keywords	Workaround
When a Network Call Redirection failed over a measured **SIP** trunk, a subsequent successful vector route-to step over the same measured **SIP** trunk failed to be measured.This un-measured call must originate over a **SIP** trunk and terminate to a measured **VDN**/Vector that provided some answer supervision (for example, announcement or wait hearing music > 1) to the incoming **SIP** call. Then the failed Network Call Redirection occurred with either - a "route-to number ~r.." vector step, or - a Multi-site Best-Service-Routing (**BSR**) "queue-to best" vector step with NCR enabled on the BSR Table. After the failed NCR step, vector processing continues to the next step which was a successful non-**NCR** route-to using **ARS**/**AAR** over the same NCR SIP trunk as the incoming call. This call did not measure or count by **CMS** or IQ.	090235	Do not measure the **SIP** trunk, although the **VDN** can be measured. Or Change the coverage of the second route-to w/o ~r from 'n' to 'y' 01 announcement...02 route-to number ~r...03 route-to number cov=y change the coverage to y-^ ^-route back out same incoming sip trunk.
European Telecommunication Standards Institute (**ETSI**) Explicit Call Transfer calls were marked as abandoned by Call Management System sometimes.	090305	
Immediate coverage on a station could break the path replacement feature.	090681	
Background failures of the inter-network region connectivity test did not cause Warning and Minor alarms to be logged.	090737	
When call was forwarded over a **SIP** trunk, History-info header was showing only extension and not the prefix in INVITE.	090839	
Under high traffic with Processor Ethernet, socket connections could close unexpectedly.	090996	
		2 of 24

Table 7: Fixes delivered to Communication Manager 5.2.1 *3 of 24*

Problem	Keywords	Workaround
While transfering a call over **SIP** trunk (session initiation protocol), call should not be dropped.	091532	
This issue was having multiple symptoms as follows: 1. 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 <VDN name>". 2. When a call which comes on a **VDN** is covered and goes to the second coverage point when the first coverage point does 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 3. When calls made to **VDN**, which is routed to an **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". These issues would be specific to "Avaya Digital Terminal for Japan" (**J24**) sets and would occur if "Idle Appearance Preference" is enabled on the station form.	091620	
On a **Communication Manager** with an IQ Release 5 but no CMS administered, if a hunt group was administered as externally measured, an error message was displayed stating that **CMS** must be administered.	091657	
Customers were unable to service observe from **BRI** stations.	091718	
An insane TN2602 media processor board could cause thrashing in maintenance, which in turn caused other maintenance to be unable to run properly.	091840	
If application Enablement **DMCC** endpoint was registerd in Independent mode to an extension, and base set was not registered to the extension, then third party call control make call failed.	091857	
Callers placed on hold in some network regions did not hear Music on Hold.	091888	Remove the music source from the system and then add it back again.
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***4 of 24***

Problem	Keywords	Workaround
The call-appearance of the station was stuck if that station performed a no-hold-conference with a call which was routed through vector administered with announcement and route-to some station.	091922	
Under certain fine timing conditions, combined with **IPSI** socket loss, a PKTINT could get into a bad state.	091945	To recover a "**`reset system 2`**" command could be issued from the **SAT**.
Services **NIC** was on eth4 after new installation	092098	
Dialouts from the Polycom **RMX** to the Polycom **HDX** faiedl to show content when requested, if the RMX was configured to allow the H.264 video codec for content (the H.239 content setting).	092274	Disable H.264 in the H.239 conference profile.
Alarms were incorrectly raised for power supplies in a G650 carrier that is not configured.	092302	
Internal or external calls via a **QSIG** Value Added trunk group, directed to a hunt group fail to follow the entire coverage path if the same hunt group is a coverage point within the coveage path for that hunt group.	092306	
Agents were blocked from logging-in due to data corruption.	092308	
In a multilocation configuration, Third Party Call Forwarding activation failed.	092343	
When an **IP** softphone, that was logged in telecommuter mode with a station over a Distributed Communications System (**DCS**) trunk, made a call to a station which had Call-Fwd enabled, the display on the softphone and on the call forwarded destination station showed the name of the telecommuter station.	092370	
System log files showed many unwanted process errors.	092377	
Previously there was a condition where H.323 audio calls could cause excessive logging due to video not being used on the call. Now there is no spurious logging for video events on audio calls.	092390	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***5 of 24***

Problem	Keywords	Workaround
Under certain conditions, occupancy information reported by the system was above 100%. A bios microcode fix to the Celeron M B1 Step processor was applied to correct this behavior.	092402	
Missed call did not log beyond 13 digits for incoming call to a busy 96xx station when Enhanced Call Logging (a firmware feature) was enabled.	092417	
In some scenarios, music on hold were not heard by the held station, when called using Shared Voice Connection feature.	092426	
Incoming calls over a trunk that was configured with a blank far end node name and port are dropped if the far end did not send fast start elements.	092428	
Calls to a **VDN** with a vector having a messaging step to msa-vm hunt-group did not terminate.	092441	
A delay in getting a *talkpath* occurred if Inter Gateway Alternate Routing (**IGAR**) was involved when going to a station's coverage path.	092448	
On S8300 server the following error was being logged about every 15 minutes: `HMM:OVERLOAD:abnormal tick ...`	092460	
Internal data related to TN2302 and TN2602 media processor boards could have become corrupted resulting in erratic H323 or **SIP** call behavior or system resets.	092464	
Calling party number was not displayed on the terminating station if the incoming call was made over a CO trunk with the country code set to 18 (China).	092470	
Under certain circumstances, a domain control may receive a Busy/Unavailable Event for a user that was no longer in the call.	092495	
SIP calls being redirected back to originating server via **NCR** (Network Call Redirection) were not being tracked by **CMS**.	092500	
Under certain internal conditions, performing an audit on customized button label entries caused a segmentation fault and a reset system 4 (resetting the **Communication Manager**).	092502	
When station called **VDN** (Vector Directory Number) which routed the call to a station across a trunk using **ARS**, then display on calling station changed from "a=VDN" to "a=ARS".	092504	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***6 of 24***

Problem	**Keywords**	**Workaround**
The mgRegister logs showed the Media Gateway **IP** address in reverse order.	092521	
Calls routed by **IGAR** (Inter-Gateway Alternate Routing) were not tracked by IQ.	092532	
It was possible for the time between a **Communication Manager** server and a **CMS** system to become non-synchronized, resulting in time change related error messages in the **CMS SPI** log.	092534	
The reception of a bad "trace route" response may result in a system reset.	092550	
BRI data calls failed when the call originated from a H.248 media gateway and terminated on a traditional port network.	092551	
IP DECT phones did not hear dialtone when the user pressed the "R" button to create a conference. This behavior was observed when the **IP DECT** phone was using a H.248 media gateway **VoIP** resource and the **IP DECT** station had shuffling turned off.	092588	
With a call either being originated at, ringing at, or on hold at an available agent, a reporting audit started and sent a malformed AUDOTHER message to **CMS**/IQ. This caused some messages that followed to be discarded, resulting in some lost reporting data.	092591	
A restriction placed on the administration of multiple **OPS** application extensions on the off-pbx-telephone station-mapping form blocked the administration of Nuance Speech Attendants.	092629	
In a system with the Maximum Concurrently Registered Unauthenticated H.323 Stations registered at capacity, when a Native H.323 endpoint does a benign re-registration, it is rejected and denial event 1911 **IP RRJ**-Exceed max endpts is generated.	092653	
Configure Server, part of the **Communication Manager SMI** (System Mangement Interface) **GUI** (Graphical User Interface) - that is **Communication Manager** Maintenance Web Pages, failed on duplicated servers that was already configured but without an alias/active hostname. (for example, after an upgrade to **Communication Manager** 5.2.x)	092688	
When calling out on a personal CO line (**PCOL**), the caller could not be observed via Service Observing.	092708	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***7 of 24***

Problem	**Keywords**	**Workaround**
ISDN PRI signaling groups that were in a media gateway (**MG**) did not go into service correctly after a link bounce if there were more than one signaling group in the **MG**.	092734	
Automatic call back failed when Look Ahead Routing was enabled on the route pattern form.	092473	
The server could possiblity reset in scenarios where **QSIG** temporary signaling connections (**TSC**'s) were used.	092638	
No *talkpath* may result when an **IP** station tried to transfer an incoming call to another Avaya **PBX**.	092562	
When the **SIP** Network Call Redirection (**NCR**) feature was turned ON and a phone attempted a blind or unattended transfer of an incoming **SIP** trunk call to an outgoing **SIP** trunk, under certain conditions the calling party did not have a *talkpath* with the transferred party.	092572	
Previously, if **SBS** (separation of bearer and Signalling) with a Central Office trunk on H248 media gateway as bearer was used to route calls to a **QSIG MWI** (message waiting indication) hunt group, then the call would fail.	092656	
The list usage node-name command gave the shared virtual node name of duplicated crossfire boards on ip interface form, did not show what that node-name belongs to.	092678	
When a NON_OPTIM **SIP** station (**SIP** station on Avaya **SES** outside Avaya domain) transfers a call from Avaya **Communication Manager** to a different station on the same **PBX** via **SIP** trunk, transfer recall fails and leaves trunk members in locked up state.	092699	
The older **UCID** was not being retained on transfer when the original call was not measured and the destination call was measured. This could result in confusion by an adjunct that monitors the call.	092802	
If removing the last **IPSI** in a fiber cluster (**CSS** or **DC PNC**) and there was a non-fiber port-network up and running, the system could lock up and deny **SAT** access.	092826	
Server restarted when multiple phone-users were involved in a call like conference, transfer etc.	092838	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***8 of 24***

Problem	Keywords	Workaround
A software error during a call transfer resulted in **CMS** being unable to track a subsequent call. Scenario: An agent put a call on hold, dialed a **VDN** extension, and successfully completed a blind transfer of the held call to the **VDN**. The associated vector routed the call off the **PBX**. This was tracked properly by **CMS**. However, **CMS** stopped receiving events about the call after the transfer, and shortly thereafter received messages for a new call but which used the same identifier (**ITN**) as was used for the earlier the transferred call.	092913	
The main server could have a lockup with no indication of the problem, resulting in an outage of approximately 16 minutes. This should no longer happen.	092976	
Due to a software error, an incorrectly formatted AUDIT message corrupted the **CMS**/IQ event stream and aborted tracking of calls.	092996	
Under certain internal conditions, Avaya Aura^{TM} **Communication Manager** may reset resulting in dropped calls and/or loss of *talkpath*.	093020	
Under unknown conditions, it was possible for a call involving a **SIP** endpoint and a non-**SIP** endpoint to reach a state where the call would try over and over again to shuffle to a direct-**IP** connection, but never succeed. This had the potential to cause a system restart after various internal resources had been exhausted.	093043	
Service observer on a dropping call caused **CMS** to drop link. Here is the call scenario: An incoming call to **CM** (**Communication Manager**)-A is delivered to agent-A after processing through **VDN**-A and vector-A. The call picks up a service-observer from **VDN**-A. Agent-A transfers the call to **VDN**-A1 which does a lookahead-route-to out across trunks to **CM**-B. The call is handled at **CM**-B and then transferred or routed by vectoring back to **CM**-A to **VDN**-A2/vector-A2. Path-replacement while in queue vectoring is active so the call path replaces in vector-A2, and is in queue. While the call is waiting for an agent to become available in vector-A2 the caller hangs up, dropping the trunk. The service observer is still on the call, causing an unexpected event to be sent to **CMS**. This causes the **CMS** link to drop, resulting in a service interruption with **CMS**.	093151	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***9 of 24***

Problem	Keywords	Workaround
Given certain misconfigured networks resulting in time outs on connections to survivable processors, **Communication Manager** may hang causing loss of service for up to 16 minutes.	093235	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-207. To see this document, go to http://support.avaya.com and search for that number.	090542	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-114. To see this document, go to http://support.avaya.com and search for that number.	091113	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-130. To see this document, go to http://support.avaya.com and search for that number.	091160	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-162. To see this document, go to http://support.avaya.com and search for that number.	091544	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-161. To see this document, go to http://support.avaya.com and search for that number.	091550	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-258, ASA-2009-191, ASA-2009-167. To see this document, go to http://support.avaya.com and search for that number.	091596	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-228. To see this document, go to http://support.avaya.com and search for that number.	091887	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-240. To see this document, go to http://support.avaya.com and search for that number.	091972	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009--239. To see this document, go to http://support.avaya.com and search for that number.	092016	
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Table 7: Fixes delivered to Communication Manager 5.2.1 *10 of 24*

Problem	Keywords	Workaround
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-244. To see this document, go to http://support.avaya.com and search for that number.	092017	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-296. To see this document, go to http://support.avaya.com and search for that number.	092355	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-373. To see this document, go to http://support.avaya.com and search for that number.	092416	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-384. To see this document, go to http://support.avaya.com and search for that number.	092527	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-350. To see this document, go to http://support.avaya.com and search for that number.	092529	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-277 and ASA-2009-349. To see this document, go to http://support.avaya.com and search for that number.	092545	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-368. To see this document, go to http://support.avaya.com and search for that number.	092568	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-380 and ASA-2009-374. To see this document, go to http://support.avaya.com and search for that number.	092665	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-385. To see this document, go to http://support.avaya.com and search for that number.	092732	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***11 of 24***

Problem	Keywords	Workaround
On systems with a large number of **IP** stations, background maintenance (periodic and sheduled) took a long time to finish. The current cycle times were available via the '`status periodic-scheduled`' command. These long cycle times delayed the detection of faulty hardware as well other background activities such as periodic lamp and ringer audits. Required a large number of registered **IP** stations (more than 5000) to experience the problem.	031800	
When a Whisper from OPTIM station A routed over H.323 or **SIP** trunk to station B busy on a call is answered back by station B, two way voice path between station A and station B was not established.	064566	
With Media Gateway present, when a Whisper from OPTIM station A routed over H.323 or SIP trunk to station B busy on a call was answered back by station B, two way voice path between station A and station B was not established.	064574	
On rare occasions, an incorrect trap for a device was reported.	072673	
A.1 called A.2, and the call covered to A.3. A.3 did not answer, and the call covered remotely to B.1. When the call to B.1 was setup, **LAR** (Look Ahead Routing) REHU (REHUnt) was triggered. When B.1 was ringing, A.3 did not stop ringing. If B.1 did not answer, the call covered to the 3rd coverage point A.4, but A.4 was unable to answer the call. The problem was seen only if the call from switch A to B triggered the LAR REHU feature and the trunk used to reach B.1 was a **QSIG** trunk with the **QSIG** Value-Added field set to y on page 4 of the trunk group form.	073067	
When a call was made to a phantom station (Administered without Hardware) having Send All Calls activated to some other extension, cover tone was not heard at caller side when call covers to cover point.	074417	
The **CLAN** Access Control List incorrectly filled up with **IP** addresses, blocking the registration of new **IP** phones.	080334	
In certain configurations, **IP** Server Interfaces were not able to receive a time synchronization.	080581	
If a call was made from an **ISDN BRI** telephone across a **QSIG** trunk to a station with an active posted message, the posted message did not display correctly on the **BRI** phone.	080602	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***12 of 24***

Problem	Keywords	Workaround
For **SIP** endpoints, Music On Hold (**MOH**) was not necessarily played back from the nearest location, that is,. from the listener's network region (**NR**).	081072	
Console could not use short multi-location extension to invoke Automatic Wakeup feature for another user when the call type for this short dialed number was set to udp.	081440	Use long extension to invoke Auto Wakeup. OR Change the 'udp' call type to the 'ext' call type in the Dial Plan Analysis form for the short extension.
When a customer administered 2 "Enhanced Call Forwarding" buttons for a station left the extension field of one button "blank" and assigned its own extension to the other button a wrong error message was displayed. Instead of displaying "`Have cfwd-enh buttons with blank ext and with station's ext`" the following error message is displayed: "`Type requires a TN2464, TN767D, MM710, TN464E or later DS1 board`".	081461	
If a vector contained a "goto vector" step with a 4 digit vector field, the Avaya Site Administration (**ASA**) Advanced -> Report feature displayed the vector step without the fourth digit in the vector field. For example, the vector step "goto vector 1234 @step 99" (where "1234" was the vector field) was displayed by **ASA** Advanced -> Report as "goto vector 123 @step 99" (the vector field was truncated to four digits). **Note:** This is only a display issue; the vector step is actually stored and processed correctly.	081664	
User saw 24 "*"s instead of "**UCID** Info" when trying to view this button if the language was "user-defined".	081967	
If a station or agent service observer was on a call that was held by the observee (either with hold, conference or transfer), reporting was not informed that the service observer was waiting to observe.	082462	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***13 of 24***

Problem	**Keywords**	**Workaround**
When a service observer hung up during a call, it was possible that reporting would not be told that the service observer was no longer waiting for a call to observe.	082596	
Trunk Selection field on off-pbx-telephone station-mapping form was not handled consistent with other fields on the form.	082782	
After OPTIM ONEX applications were added by oneX Server and both VISITOR and HOME Enterprise Mobility Users (**EMU**) registered for the same station extension, de-registration of both VISITOR and HOME EMUs caused the switch to restart.	082838	
Corrupted packets on the control network could cause boards, especially TN799 **CLAN** boards and TN2602 boards to be taken out of service and not come back into service for up to an hour. To bring the boards back into service immediately on port network 'x' execute the "`reset port-network x level 2`" command on the **SAT**.	082862	
If a best-service routing (**BSR**) table did not have a "name" assigned, it could not be removed.	083127	
Whenever a digital handset extended a call and dropped off before the extended to party could answer the call, the other party on the call would hear Music On Hold (**MOH**) instead of ring back.	083278	
When Dynamic Queue Position was active and an agent became available with "Service Objective" set to yes, the skill's service objective overrode the Dynamic Queue Position.	083287	
DTMF tones were not recognized on an incoming **SIP** trunk.	083304	
When a **SIP** station already in an active call attempted to send a whisper page to another station, the original call was dropped.	083329	
If the caller was on an **IP** trunk and the agent and service observer were local, when the service observer went into listen/talk mode, the flashing lamp could go solid even though the service observer was still in listen/talk mode.	083385	
The title of "Home User" field on "list mappings-acquired" form was not consistent with documentation.	083437	
In scenarios involving XMOBILE stations, the caller unexpectedly heard music for a small duration before the call was covered.	083535	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***14 of 24***

Problem	Keywords	Workaround
If AUX_WORK button was activated on 96xx **IP** phone and an incoming call to that phone was directed to coverage using 'To Vmail' button, the AUX-WORK button got deactivated.	083726	
When a call follows a coverage path containing three coverage points the third or final coverage point was not reached. This occurred when the coverage path was admininistered with a first coverage point that routed the call via a **QSIG** trunk group, the second coverage point routed the call via a non-**ISDN** trunk group and the final coverage point was a local assigned extension.	083760	
A six-party conference was not established if the last party joining the call was a **SIP** station trying to use a bridged appearance to join the call.	090000	
When a call was made to a principal **IP DECT** station which had another **IP DECT** station as it's bridge and the call was rejected from bridged **IP DECT** station, principal **IP DECT** station was still ringing but caller was getting busy tone. After answering the call at principal **IP DECT** station, there was talkpath but caller was still getting busy tone.	090160	
A station with a status busy indicator had that indicator lamp stuck ON if it was tracking a Visiting Enterprise Mobility User (**VEMU**) endpoint.	090211	
The **IP** softphone was not showing a lamp update when used in shared control mode with a 9650 or similar set type.	090364	
A switch name of 20 characters was not always displayed correctly on some admin and measurement forms. The last character would get truncated.	090377	
TTS and Link bounce were not working properly after changing the administration of "Near End Establishes **TCP** Signaling Socket?" on the ip-network-region form from "y" to "n" or vice versa and rebooting the phone.	090399	
SAT command "**`display button-labels`**" was not working for 16xx station set types.	090430	
Conditional Call Extend fields were not reset to default values when application type changed from **PBFMC** application to SPFMC application.	090461	
list measurements call-summary was not counting telecommuter service links.	090486	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***15 of 24***

Problem	Keywords	Workaround
The following "`list usage button-type`" **SAT** (System Access Terminal) commands allowed input of data that was not used in the query for the buttons; "crss-alert", "hunt-ns", "night-serv" and "trunk-ns". Also, the following commands had cosmetic problems with their help message; "`list synchronization`", "`list ip-interface val`", "`list ip-interface clan`", "`list ip-interface medpro`", "`list node-names v4`" and "`list node-names v4`".	090506	
On entry of a location field value on the off-pbx-telephone station-mapping form for applications that did not support the location feature, displayed error message did not specify **SPFMC** as one of the supported applications for the location feature.	090537	
When off-pbx-telephone station mapping was added for EC500 application containing the number mapping of Country Code of blank and Phone Number of CC1+NUM1, where CC1 and NUM1 are the Country Code and Phone Number mapping of an existing ONE-X application, the Country Code of ONE-X application mapping was incorrectly set to blank.	090538	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-207. To see this document, go to http://support.avaya.com and search for that number.	090542	
Rarely, an **ESS** server in a duplex pair configured to use software duplication may reboot a second time after a "`save trans ess`" command was executed on the main, or when this command was run as a part of scheduled maintenance.	090558	
If Non **TTS IP** endpoints are registered via the PROCR (processor interface) of a duplicated **Communication Manager**, a server interchange may leave these **IP** phone registrations around if the **IP** phones do not re-register with **Communication Manager**.	090562	
The list measurements ip dsp-resource hourly commands allowed invalid options like: "`list measurements ip dsp-resource pn hourly`" OR "`list measurements ip dsp-resource gw hourly`".	090571	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***16 of 24***

Problem	**Keywords**	**Workaround**
When a call covered to an announcement and the calling party was hearing it, and if the principal party tried to bridge-on to the call, the announcement was not dropped. This occurred always when 'Maintain **SBA** at Principal' is set to 'y' on 'system-parameters coverage' form and a call covered to an announcement.	090635	Set 'Maintain **SBA** At Principal?' to 'n'.
Connections involving traditional port network **VoIP** resources may not be re-routed if the port network **VoIP** resource fails. Such connections could involve **SIP** or H.323 stations, **SIP** or H.323 trunks, and **IP** connections between **IP**-connected port networks and H.248 media gateways.	090657	
Unnecessary errors were printed to logs for a direct-**IP** to **IP** station call.	090685	
After OPTIM ONEX applications were added by oneX Server and both VISITOR and HOME Enterprise Mobility (**EMU**) Users registered for the same station extension, de-registration of both VISITOR and HOME **EMU**s caused data loss for some of the OPTIM ONEX applications.	090741	
On the 'Link Port Status' form, the value of the 'Service Port Location' field was not fully visible as it was getting truncated.	090795	
Whenever a Toshiba **SIP** Phone (**TSP**) joined a conference, using Single Step Conference (**SSC**) and dropped off, the server would reset. This problem was specific to **TSP**'s having 2.05.T7 firmware.	090809	
When a 96xx (Spice) station, having a pending conference, received an unrelated incoming call from another station, the incoming call was wrongly treated as part of a conference call.	090893	
Some help messages on system-parameters features form and video-bridge form were not fully visible.	090910	
Call log entries for the calling party and called party were incorrect after adding **AAR**/**ARS** for internal calls.	090947	
Under certain circumstances an intercepted **DID** call did not route to the attendant group, even though the field **DID**/Tie/**ISDN**/**SIP** Intercept Treatment on the system-parameters feature form is set to 'attd'.	090958	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***17 of 24***

Problem	Keywords	Workaround
When path replacement took place, for calls transferred over Qsig Trunks to a "Vector Directory Number (**VDN**)" which is redirecting to an Agent, the display on the agent showed the "Calling Party Number" instead of "Calling party name to **VDN** name".	091017	
If the "Force Phones and Gateways to Active **LSP**s" feature was on and the customer had Enterprise Survivable Servers (**ESS**), then minor alarms were generated on the **ESS** for unregistered Local Survivable Processors (**LSP**).	091078	
Communication Manager had certain vulnerabilities described in Avaya Security Advisory ASA-2009-114. To see this document, go to http://support.avaya.com and search for that number.	091113	
On rare occasions, after an interchange, a one minute discrepancy may occur between the **MG**'s link status as shown on the 'list media-gateway' form and on the 'status media-gateway' form.	091120	
After various test scenarios were run (resets, server interchanges, etc...), "**`status socket-usage`**" command showed fewer sockets than expected in "Registered **IP** Endpoints with **TCP** Signaling Socket Established:" field.	091152	
On some occasions, using the commands **`'enable nr-registration'`**, **`'disable nr-registration'`**, **`'enable mg-return'`**, or **`'disable mg-return'`** commands could interfere with internal network region operations due to previous user commands, a change in **LSP** status, or the Time of Day **MG** registration process. The end result could be one or more network regions in an incorrect state.	091175	
Under certain conditions involving control network outages and server interchanges, standby TN2312 **IP** server interface boards could stay out of service.	091221	
An error message was displyed while changing **IP** address of a node-name belonging to H.323 Signaling Group.	091237	
On rare occasions, a call setup on a **LSP** could be dropped if the **MG** returned to the main server and then shortly thereafter went to the **LSP** again.	091242	
Specific internal **Communication Manager** conditions could lead to a gradual loss of available system memory, potentially impacting **Communication Manager** operation.	091246	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***18 of 24***

Problem	**Keywords**	**Workaround**
In certain scenarios, the DCON message sent to reporting was identified as a DCON20 instead of a DCON22. The contents were the same, though.	091251	
When a remote conference was active over an **ISDN-PRI** trunk using a Media Gateway, the conference tone stopped playing if the conference originator (**IP** phone) pressed a digit on the dialpad.	091290	
Changes of Mapping Mode from "both" to "termination" or from "termination" to "both" were blocked for off-pbx-telephone station-mapping entries that included Country Code.	091298	
The "**IP** Address/Mask:" field on the status media-processor form truncates part of the Mask when the **IP** address is 16 characters.	091304	
When a server failed over to an Enterprise Survivable Server (**ESS**), voice/network statistics measurement reports were not generated for media processors.	091321	
When a call was made to an **IP DECT** station A which had Call forward Busy/**DA** enabled to another **IP DECT** station B. As soon as **IP DECT** station A started ringing, pressed the reject button on **IP DECT** station A, call was forwarded to **IP DECT** station B. At this time **IP DECT** station B rang but caller got reorder tone instead of ring back tone. When call was answered by **IP DECT** station B, there was a *talkpath* but caller again heard reorder tone.	091355	
Incoming **IP** trunk calls to a station on a Media Gateway failed if the incoming call traffic was very high.	091372	
The user was not allowed to enable the Enable **VoIP**/Network Thresholds? field on the ip-interface form for active/standby crossfire boards if the standby board was administered on the meas-selection media-processor form.	091377	
When the first station registered on a Media Gateway joined a Meet-me conference other parties did not hear the associated conference tone.	091450	
If a coverage path was assigned with first coverage point as remote coverage and second probably local with coverage criteria as 'all', then rings given on coverage path for remote coverage were not used.	091454	
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Table 7: Fixes delivered to Communication Manager 5.2.1 *19 of 24*

Problem	Keywords	Workaround
When the 'Force phones and gateways to active **LSP**s' field is 'y' and **MG**s have registered to the main server because one or more **MG**s have satisfied their Time-Day-Window recovery rule, an unregistered **MG** will cause all of the network regions to become auto-disbled at the end of the hour and no **MG**s will be allowed to register until the next **TDW** has been satisfied.	091455	
On intermittent occasions, the 'list measurements ip dsp gw' form showed 'n/a' for a media gateway.	091471	
Call Detail Record output using the enhanced expanded or customized format could have failed or been corrupted if the record was close to the maximum size.	091475	
When adding a non-**SIP** station, the "**SAC**/**CF** override" field on the station form has an "a(sk)" option. However, if this option is used and then this station is added on the off-pbx-telephone station-mapping form where the application is **OPS** and the extension number is same as the phone number, on submitting the form a new warning message is displayed, "WARNING: '**SAC**/CF Override' 'ask' option not supported on **SIP** OPS endpoints."	091483	
When running test board repeat 5 on a TN763 board test 114 will PASS the first time and FAIL everytime after.	091502	
User noticed a missing and duplicated field ID's (**FID**) for the 'list partition-route-table' command.	091515	
The proc error "`CM5_proc_err: pro=7171,err=203,seq=7832,da1=176(0xb0),da2=0(0x0)]`" was created on every **SIP** call if the signaling group used did not have the far end domain administered. It was not an error to leave the far end domain blank so this proc error was unnecessary and was removed.	091519	
For 96xx (Spice) **IP** stations where the system-parameters coverage-forwarding form has "Maintain **SBA** at Principal" set to "y", when a call was picked up by a coverage point, the call appearance on principal would continue flashing.	091525	
Using the ossit term type, the list ip-interface all command showed non-duplicated ip-interface records as being duplicated.	091530	
SIP Diversion Hdr will now contain the "sips" **URI** when sent over a secure **SIP** trunk.	091555	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***20 of 24***

Problem	Keywords	Workaround
After adding a static **IP** route on the PROCR ethernet interface, the "list ip-interface all" **SAT** command shows an incorrect gateway address for the PROCR ip-interface.	091576	
A Modular Messaging station having its **MWI** (Message Waiting Indicator) ON was logged out when an unsuccessful attempt was made to removed this station from **Communication Manager**.	091591	
Incoming calls from some Non-Avaya systems via a **SIP** trunk may result in no *talkpath*.	091595	
There was no two-way audio path after processing re-Invite with **SDP** having a=recvonly.	091597	
SIP trunk calls between **Communication Manager** and some non-Avaya systems may lose *talkpath* if the far end went on hold, stayed on hold for a while, and then unheld the call.	091599	
Call originator should hear reorder tone when attempting an outgoing trunk call and no operational outgoing trunk facilities exist.	091606	
The number of power supplies was not shown correctly for a S8710/20/30 server.	091607	
When doing a "remove ip-interface" of a board used on the Meas-Selection Medpro form, no warning message was displayed to alert the user that the ip-interface being removed was administered on the Meas-Selection form and would automatically be removed from that form.	091613	
Denial event 1958 "**IP** GRJ-Invalid extension" sometimes included extra characters in the extension number, making the data hard to interpret.	091630	
List-trace output for fax was missing the T.38 mode line which displayed the negotiated parameters. Now this line was displayed after fax signaling had completed.	091676	
9610 **SIP** was not a valid or supported station type and was no longer allowed when adding or changing stations.	091722	
The time/date and lamp update messages were sent down to a TN771 board, which caused an alarm on board and required board reinsertion.	091767	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***21 of 24***

Problem	Keywords	Workaround
Softkey labels from avaya_user-defined.txt file present on a Avaya **Communication Manager** server were not displayed correctly on phones like 84xx, 64xx and 4624 that support 5 character softkey displays.	091770	
A vertical scroll bar for the working area of the Syslog Server **SMI** page did not appear when the window size was not large enough to display all the entries for the page.	091776	
If the "`get forced-takeover ipserver-interface`" **SAT** command was executed on an **ESS** Server, followed shortly by a "`get forced-takeover ipserver-interface`" command on the main server, it could leave boards and ports in a Port network out of service. To recover from the problem, execute a "`reset port-network x level 2`" **SAT** command.	091781	
The Alternate Gatekeeper List page of the '`status station`' command had a note at the bottom of the page redirecting users to the BACKUP SERVER fields of the **IP**-NETWORK-REGION form that bacame confusing with the introduction of **ESS**s as backup servers because it only mentioned **LSP**s.	091798	
This change modifies our media processor's advertised T.38 receive capabilities for maximum jitter buffer and packet size. Values for these parameters were previously over-advertised. This could cause buffer overflow and fax failure if **Communication Manager** received a fax from an endpoint that exceeds our buffer or packet size limits.	091803	
If an Avaya **IP** softphone is registered to an extension on the **Communication Manager**, and a **DMCC** (Device Media and Call Control) **CTI** based softphone registers to the extension in shared control mode, before the Softphone made any calls, then Softphone cannot make call using the **IP** Softphone's normal **GUI** interface (entering the digits and pushing the dial button).	091830	Use the phone-gui option in the **IP** Softphone application, and go off hook and dial the number once.
If an agent transferred a hard held call to a destination that did not answer before the transfer completed (for example, a **VDN** with a long wait step), at the point of transfer completion, IQ/**CMS** reported that the call was abandoned while in queue or while ringing.	091864	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***22 of 24***

Problem	**Keywords**	**Workaround**
Unnecessary proc errors were generated and saved into software error log if an audio call was carried over a video-enabled IP trunk.	091879	
Layer 3 test was not running correctly on H.323 and **SIP** signaling groups on a survivable server when the survivable server was active resulting in the signaling groups going out of service when they should not.	091883	
When station A tried to call station B which was unregistered, station A would get ring back if the "Don't Answer Criteria For Logged Off IP/PSA/TTI Stations" was set to "y" on "system-parameters features" form. This was expected behavior. The problem was after station B registered, station B did not get any indication of the incoming call, but it could answer the call by pushing the first line appearance button.	091911	
When running some of the **SMI** (System Managment Interface) **GUI** pages with AJAX, the page could exceed the max amount of memory allocated resulting in a blank page, that is, `no output, without informing the user`.	091997	
When using the "change extension-station" **SAT** (System Access Terminal) command, customers were unable to relate the extension being changed to the new extension because the command history log did not include the new extension.	092018	
Whenever call made to a Vector Directory Number (**VDN**) was forwarded over an **ISDN-PRI** trunk, and was answered by the remote station, the display on the calling party showed the Trunk Access Code (**TAC**) and the trunk group name. It should have shown the name and number of the party to which the VDN was routing. The problem was specific to "Avaya Digital Terminal for Japan" (**J24**) stations.	092094	
Users were allowed to administer **CLAN** ip-interfaces, PPP data-modules, and communication-interface processor-channels with a link of 254. However, the PROCR ip-interface used this link by default.	092124	
Customers were unable to add entries to the Uniform Dial Plan form if the entry had the Len column greater than 13 and the Net column was set to "ext" (for "extension").	092189	
SIP trunks were not properly released at the remote end when the service provider sent a BYE in the early dialog phase.	092229	
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Table 7: Fixes delivered to Communication Manager 5.2.1 *23 of 24*

Problem	Keywords	Workaround
When call was placed over **QSIG**-value trunk and **SIP** phone had bridge-appr for the called party, then **SIP** phone showed Name-1 even if it was capable of showing unicode name.	092238	
For a switch which had a two port network when call pick up or call unpark was performed, *talkpath* must be there and if direct ip was enabled then the call must be in direct **IP**.	092260	
The DCP phone is in disconneted state as the port is changed from a regular port to a LAN port. This happens because when AES DMCC endpoint takes over the extension (by registering as main) and later unregisteres, **Communication Manager** does restore the DCP (regular)port. This applies only if the endpoint was an AES DMCC endpoint.	092271	
When 1XC (One-X Communicator) was in shared control mode with a 96xx station and was busy, any new call made to that station was not logged in the Missed Call Log on 1XC. Whereas in the same scenario, a Missed Call was logged on the 96xx hard phone as expected.	092525	
If a Polycom **RMX** was reset while it was configured as a video bridge and actually in use, **Communication Manager** segfault and may provided inaccurate bridge status information.	092554	Terminate running conferences from the **RMX** admin screen before resetting.
After succesfully installing a phone FW package, using the "Download Files" **Communication Manager SMI** (System Management Interface) **GUI** (Graphical User Interface) - that is the **Communication Manager** Maintenance Web pages - the user was presented with a "tripwire" button if Tripwire was enabled. Clicking this button yielded "`SMI GUI not found`" or "`Page not found`".	092589	
Video calls failed and were possibly dropped, depending on the endpoints. Affected endpoints were Tandberg 1700 behind a Tandberg **VCS** using H.264; Polycom V500 using DBC-2 as an H.239 codec; and Polycom **HDX** using H.263 with custom picture format(s) using custom clock frequencies.	092703	Turn off the affected codec.
When the medium priority Expected Wait Time (**EWT**) and/or the low priority **EWT** were greater than 255, neither was specified correctly to **CMS**/IQ via an EWTAUDIT20 message.	092738	
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Table 7: Fixes delivered to Communication Manager 5.2.1 ***24 of 24***

Problem	Keywords	Workaround
If a TN799 **CLAN** link was down, the **`list measurements clan`** **SAT** command could be missing data for a different CLAN.	092769	
When an incoming **SIP** call terminated to a **Communication Manager** endpoint that had call forwarding activated the **SIP** 181 message was received without the "Call Is Being Forwarded" phrase.	092782	
There was no *talkpath* if two calls were made to an **IP** station, and the **IP** station answered the second call.	092919	
The system gets 100% call failure rate (and 100% **CPU** occupancy) as all **Communication Manager** to **Communication Manager SIP** calls failed repeatly sending **SIP** 422 Session timer refresh value too small messages under call load.	091804	
G729 annexb=yes and ip-codec-set admined with G729 only, G729 selected.	091963	
If an **AES DMCC** (H.323) endpoint registered in "main" mode un-registers while on a call and there is another **AES** endpoint registered in "independet" mode is currently reigstered to the same extension, the talk capability is not transferred to the "independent" endpoint.	093077	
The MM118, a non-supported board type, could be administered in a G430 media gateway.	091327	
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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 ***1 of 6***

Problem	Keywords	Workaround
Communication Manager 5.2.1 and **SIP** Enablement Services 5.2.1 only support the US Robotics USR5637-**OEM** modem (comcode700464506) when running on the S8800 Server. All other servers supported by **Communication Manager** and **SIP** Enablement Services 5.2.1 support the legacy MultiTech modems in addition to the US Robotics modem, as defined in **PSN** 1938.	NA	
"Server down" alarming for S8800 Simplex Servers running **Communication Manager** 5.2.1 or **SIP** Enablement Services 5.2.1 requires **SAL** 1.8, which will not be available to the field before December 2009. S8800 Simplex Servers using a modem for alarming and access do not have the server down alarming capability, which is provided by the **SAMP** maintenance card on the S85xx Series Servers. When available, **SAL** 1.8 will be made available to customers automatically via **PLDS** download.	NA	
H.248 Branch Gateways administered with a recovery rule may fail to automatically re-register with the main server(s) when the main comes back into service.	093316	
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Table 8: Known problems in Communication Manager 5.2.1 ***2 of 6***

Problem	Keywords	Workaround
Due to memory constraints, **SIP** trunk integration to Voice Portal is not supported with the following Communication Manager configurations: 1. S8300B/C/D **LSP** servers with either standard or XL memory configuration 2. S8710 or S8720 MAIN/**ESS** servers with standard memory configuration 3. S8500B/C **LSP**/**ESS** servers with XL memory configuration 4. S8720 MAIN/**ESS** servers configured with software duplication AND XL memory.	NA	
Due to memory constraints, S8720 MAIN/**ESS** servers configured with software duplication and XL memory, and the S8500B/C **LSP**/**ESS** servers configured with XL memory will only support up to 6K **SIP** endpoints and 7K **SIP** trunks with non-Voice Portal and non-VIDEO call traffic.	NA	
An S8500B, S8500C or S8510 server running **Communication Manager** 5.2.1 as a survivable server (**ESS** or **LSP**) with only Processor Ethernet connectivity cannot support the full capacity of a main server(s) using the XL memory configuration. The S8500B, S8500C or S8510 when in survivable mode will only be able to support a total of 2800 **IP** connections. One **IP** connection is required for each **IP** station (H.323), **IP** Trunk, (H.323 and **SIP**) and Media Gateway supported by the server.	093422	
		2 of 6

Table 8: Known problems in Communication Manager 5.2.1 *3 of 6*

Problem	Keywords	Workaround
Call failures and degraded system performance can occur if **SIP** call hold times are longer than the session refresh value administered on the trunk group forms. If the average queue/hold time is greater than 600 seconds (10 minutes), the default value on the trunk group form then the session refresh value must be changed to a value greater than 600 seconds. The maximum session refresh value that can be entered is 64,800 seconds (18 hours). This is very important in Contact Center installations with G860 based **SIP** trunks coming into **Communication Manager** (non-Voice Portal implementation), where the queue times may be very long.	093338	
Message Trace Analysis (**MTA**) does not work on servers running **Communication Manager** 5.2.1. The System Log web page returns the following message when the interpreted Message Tracer (**MTA**)" text box is 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	
An upgrade to **Communication Manager** 5.2.1 will not be call preserving if PPP links are active during the upgrade (stable calls will drop). A COLD 2 reset can also occur if the PPP links account for more than 50% of the active calls on the system (unlikely). This issue is not impacted by a modem connected to the server using PPP.	093396	Disable all active PPP links prior to the upgrade, other than the modem. Bring up a **SAT** session on the active server and for every PPP link issue the following commands: **`change data-module <data-module-extension>`** change the Establish Connection? field from y to 'n'.
		3 of 6

Table 8: Known problems in Communication Manager 5.2.1 ***4 of 6***

Problem	Keywords	Workaround
The “Reset CM” button will fail after you install a new license on a **Communication Manager** 5.2.1 survivable server (**ESS**/**LSP**) using the "Install the license file specified below" option on the License File page of the System Management Interface (**SMI**). The “Reset CM” button is only presented if the license is successfully installed and when the previous license state was “no license” (missing). This would be the case just after an upgrade across a major release boundary (for example, 4.x to 5.2.1) or with a new installation of 5.2.1.	093400	Use the “Install the license file I previously downloaded option” instead and the “Reset CM” button works.
Pre-upgrade installation patches are required for S8710 servers upgrading to **Communication Manager** 5.2.1 from prior releases and for all servers upgrading from **Communication Manager** 2.2, 2.2.1 and 2.2.2. See **PCN** 1687P and 1688P at http://support.avaya.com for additional information.	NA	
After upgrading S87xx duplicated servers to **Communication Manager** 5.2 or greater, the **IP** Alias field on the Set Identities/Configure Interfaces form is blanked out. This field should be re-populated and submitted following the upgrade and at the same time that Processor Ethernet is enabled for any **ESS** servers associated with the main server pair. This form is available under the Installation -> Configure Server option of the System Management Interface and the address of the **IP** Alias can be obtained from ***`/etc/hosts or /etc/opt/ecs/servers.conf`*** on either main server.	093362	
		4 of 6

Table 8: Known problems in Communication Manager 5.2.1 ***5 of 6***

Problem	Keywords	Workaround
Calls may be dropped and port networks may reset on systems with software duplicated main servers and duplicated **CSS**/**ATM** center stages with the B-side active at the time of the upgrade to **Communication Manager** 5.2.1.	093387	Do a **PNC** interchange before the upgrade.
Migrations from S87xx Servers running prior releases of **Communication Manager** to S8800 Duplex Servers require a manual step for memory configuration. Immediately following dataset restoration to the S8800 Server, run the Installation -> Configure Sever option of the System Management Interface and set the memory configuration to whatever it was on the original S87xx server pair (Standard or XL) unless there is a pre-determined reason to convert the memory configuration. Typically S87xx servers running Communication Manager 3.x and earlier should be considered "Standard" memory configuration.	NA	
		5 of 6

Table 8: Known problems in Communication Manager 5.2.1 ***6 of 6***

Problem	Keywords	Workaround
When the "Force gateways and phones to active **LSP**s" field is 'y' in the "system-parameters ip-options" form, there are occasions when media gateways with a time of day window recovery rule are not forced back to the **LSP** if all gateways in the **LSP** group have not re-registered with the main server(s) at the end of the specified time of day recovery window. In addition, there are occasions when running the '**`disable nr-registration`**" command elicits a false 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	Execute the **SAT** command "**`enable mg-return`**".
		6 of 6

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
ASA	Avaya Site Administration
ASAI	Adjunct Switch Applications Interface
ATM	Asynchronous Transfer Mode
AVP	Avaya Voice Portal
AWOH	Administered WithOut Hardware
BA	Bridge Appearance
BSR	Best Service Routing
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
CMM	Communication Manager Messaging
CMS	Call Management System
CNC	Control Network C
COR	Class of Restriction
CPU	Central Processing Unit
CSS	Center Stage Switch
CTI	Computer Telephony Integration
DC	Direct Current
DCP	Digital Communications Protocol
DCS	Distributed Communication System
DECT	Digitally Enhanced Cordless Telecommunications
DMCC	Device Media and Call Control
DPT	Dial Plan Transparency

DSP	Digital Signal Processor
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
EMU	Enterprise Mobility Users
ESS	Enterprise Survivable Server
ETSI	European Telecommunication Standards Institute
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
MST	Message Sequence Trace
MTA	Message Trace Analysis
MWI	Message Waiting Indicator
NCR	Network Call Redirection
NIC	Network Interface Card
NR	Network Region
OEM	Original Equipment Manufacturer

PAM	Pluggable Authentication Modules
PBX	Private Branch eXchange
PE	Processor Ethernet
PSA	Personal Station Access
PSTN	Public Switched Telephone Network
PCD	Packet Control Driver
PCOL	Personal Central Office Line
PNC	Port Network Connectivity
QSIG	International Standard for inter-PBX feature transparency at the Q reference point
RDTT	Reliable Data Transport Tool
RFC	Request for Comments
RMB	Remote Maintenance Board
RMX	A Polycom media conferencing platform, used by CM as a video and audio bridge
RTP	Real-Time Protocol
SAC	Send All Calls
SAT	System Access Terminal
SAL	Secure Access Link
SAMP	Server Access and Maintenance Processor
SBA	Simulated Bridge Appearance
SBC	Separation of Bearer and Signaling
SBS	Separation of Bearer and Signaling
SES	SIP Enablement Services
SIP	Session Initiation Protocol
SDP	Session Description Protocol
SO	Service observer
SMI	System Management Interface
SVNS	Simple Voice Network Statistics
TAC	Trunk Access Code
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TSC	Temporary Signaling Connection
TSP	Toshiba SIP Phone
TSRA	Time Slot Record Audit
TTI	Terminal Translation Initialization

TTS	Time To Service
UCID	Universal Call ID
URI	Uniform Resource Identifier
USNI	United States Network Interface
USB	Universal Serial Bus
VALU	Value-Added
VDN	Vector Directory Number
VOA	VDN of origin Announcement
VoIP	Voice over Internet Protocol
VEMU	Visitor Enterprise Mobility User
VLAN	Virtual Local Area Network
VSX	A Polycom standard definition video room system