Communication Manager 5.1.2 SP#3 Release Notes

Issue 1
September 28, 2009

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

Avaya fraud intervention

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

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Providing Telecommunications Security

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

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

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

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

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

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

Responsibility for Your Company's Telecommunications Security

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

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

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

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

TCP/IP Facilities

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

Standards Compliance

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

Federal Communications Commission Statement

Part 15:

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

Canadian Department of Communications (DOC) Interference Information

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

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

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

European Union Declarations of Conformity

CE

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

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Contents

Changes delivered to Communication Manager 5.1.2 SP#3

Communication Manager 5.1.2 SP#3 Release Notes

The **Communication Manager** service packs are cumulative and changes in **Communication Manager** 5.1.2 SP#0, SP#1, and SP#2 are included in **Communication Manager** 5.1.2 SP#3. The changes delivered to **Communication Manager** 5.1.2 SP #3 are grouped as follows:

- Table 1: Enhancements delivered to Communication Manager 5.1.2 SP#2 on page 5
- Table 2: Fixes delivered to Communication Manager 5.1.2 SP#0 on page 5
- Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 on page 6
- Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 on page 52
- Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 on page 68
- Table 6: Known problems in Communication Manager 5.1.2 SP#3 on page 82

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.

IA770

For information regarding IA770 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 **I** in that list. All documents starting with letter **I** are displayed.
5. Click **IA770 INTUITY™ AUDIX® Messaging Application**.

   The overview of **IA770 INTUITY™ AUDIX® Messaging Application** is displayed.
6. Under **Product Information**, click **Downloads**.
7. Choose the appropriate release from the drop-down list and click the link to the **IA 770 INTUITY AUDIX Embedded Messaging Application Patches Release x.y.z**.

Enhancements

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

Table 1: Enhancements delivered to Communication Manager 5.1.2 SP#2

Problem	Keywords	Workaround
This change allows four new capabilities associated with 1X Mobile. 1. Allowing **ASAI** to originate a call from an unregistered IP station. 2. Allow calls to terminate to a square bridged **CTI** station. 3. Allow **ASAI** to originate a call from a bridged appearance administered on a **CTI** station. 4. Allow a **CTI** extension to have an off-premise mapping associated with it.	083400	

Problems fixed in Communication Manager 5.1.2 SP#0

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

Table 2: Fixes delivered to Communication Manager 5.1.2 SP#0

Problem	Keywords	Workaround
When transfering a call to an endpoint that goes to a coverage, which is a **VDN**, the transfer is denied.	083622	Change the coverage path to avoid use of **VDN**.

Problems fixed in Communication Manager 5.1.2 SP#1

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

Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***1 of 46***

Problem	Keywords	Workaround
The duplication link dropped whenever either server (Active or Standby) in a duplicated pair, running in software duplication mode, underwent a software reload. This was expected behavior. However, when the link dropped and the software reload occurred, a major duplication alarm (#2) was sometimes generated by the Standby server. This often occurred when the Standby server was undergoing a software reload for reasons other than a server interchange. For example, this sometimes occurred on **ESS** servers following a file synchronization from the main server. System behavior was otherwise unaffected. Server(s) impacted: **S8720** & **S8730** Servers running in software duplication mode.	071816	
When Communication Manager makes or receives **SIP** calls through a **TCP** connection with the **Avaya Session Manager**, all calls fail.	073434	
Whenever the arbiter process was being patched on an Active server, where the Standby server was in service (not busied out), the customer would see a minor platform alarm (**ARB** event 14) exactly 15 minutes later. **Example**: SERVER ALARMS ============= ID Source EvtID Lvl Ack Date 2 ARB 14 MIN Y Wed Jul 23 12:07:36 MDT 2008 Server(s) impacted: S87xx only	073804	Busy out or release the Standby server within 15 minutes after applying a patch to the arbiter process.
Sometimes when a service pack was activated onto a software duplicated system, active calls may have been dropped if the standby server became active when the system was not refreshed.	074289	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***2 of 46***

Problem	Keywords	Workaround
If, while duplicating a **96XX Station**, any additional button modules are added (maximum three for **96xx Stations**) and administered, these button translations will be not saved for module one and will not show up on the **Display Station** form for **96XX Stations**.	080277	This problem can be avoided by not making changes when duplicating a Station or by making the changes on the original Station before duplicating it.
When Communication Manager received a specific **SIP** protocol request, it processed the trunk message but incorrectly kept the previous trunk resources. For example, a call was active between two entities via trunk 10. When the **SIP** message was received on trunk six, the old call should have been replaced by a new one that used trunk six. Instead, the call was kept on trunk 10, leading to additional trunk-resource consumption, with trunk members in use that were different than intended.	080306	
If **TN799 CLAN** boards were taken out of service, physically removed, or reset by a maintenance action while a ping was active, it could result in the loss of a system resource in Communication Manager. If this occurred twice, maintenance actions such as firmware download or bringing a port network back into service after a network outage failed.	080329	To recover from this problem, a **System Warm Reset** can be executed.
Calls between **Polycom Path Navigator** registered endpoints and **Communication Manager** connected endpoints may fail to get video if there are bandwidth restrictions on the call.	080386	This can be avoided by ensuring that **Path Navigator** endpoints initiate calls at a bandwidth lower or the same as the **Communication Manager** bandwidth.
Eliminate traps and system resets that may occur in systems using the dial plan transparency (**DPT**) feature.	080542	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***3 of 46***

Problem	Keywords	Workaround
Sometimes on **S8400A/B**, **S8500B/C**, or **S8510** systems, during filesynch an alarm of `_WD,A,26,MIN` was generated.	080552	
Communication Manager does not send **SIP REFER** if first button is brdg appr and rings. This problem is reproducible in all Linux Communication Manager platform when the following conditions are triggered: 1. When **brdg-appr** is assigned to first button of a **Station**. 2. The **Station** is associated with Microsoft Office Communicator. 3. The bridge appearance is ringing. 4. if a call appearance of the **Station** receives a call and the user answers the call via **Microsoft Office Communicator pop-up**.	080637	
Transfered call is never recalled. This problem is reproducible in all **Linux Communication Manager platform** with the following configuration: ● Communication Manager 5.0 load 825.4 + 15053 ● SIP Enablement Server 5.0 load 825.31 +SP1 ● Application Enablement Server 4.1 ● Microsoft Office Communicator 2005 ● ignite 2.0.3.0	080859	1. Set the **Recall** timer in system options to 15sec. 2. Blind Transfer is performed using ASAI (Addjunct switch application interface)
There were problems with the way processor occupancy was calculated on the **S8400 Servers**, which could cause Communication Manager to go into overload mode and/or to display occupancy results for the **list measurements occupancy** SAT commands which far exceeded 100%. The occupancy was, in reality, not high enough to cause Communication Manager to go into overload.	080896	
An **ACD** agent in the **Manual-In work mode** was active on an **ACD** call. A supervisor would perform an Add Skill operation for the agent. When the agent dropped the active ACD call, it would go to the **AFTER-CALL work mode**. The **AFTER-CALL button lamp** would initially light but then flutter and go out. A following system audit would re-light the lamp. Then, the **AFTER-CALL** button will stay lit, as expected.	080982	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***4 of 46***

Problem	Keywords	Workaround
There were several cases where the display information on either the monitoring or the the monitored **Station** were incorrect when using the **Team** button. If **Station A** had a **Team** button with the extension of **Station B** and **Station C** had a bridge appearance button to **Station B**, and **Station D** called **Station C** and then the user of **Station A** presseed the the **Team** button two times to pick up the call from the monitored **Station B**, then when the call was picked up the display of the monitoring **Station A** was not updated. Some incoming international calls to the monitoring **Station** using the **Team** button did not display all the prefix digits.	081073	
For every incoming call the phone starts a timer which tells the user the duration of the call. If the received call is a call transferred from another user and the received user puts the call on hold and later unholds the call, the call duration timer would be reset for every hold-unhold sequence. Thus user would not know the total duration of the call. This problem is observed only with **46XX** and **96XX** phones.	081082	
Calls whose talk path was set up using Inter-Gateway Alternate Routing (**IGAR**) did not get talk path, if the **ISDN** answer signal was delayed so that it arrived while the **IGAR** authorization digits were being transmitted. (Normally the **ISDN** answer signal should arrive first.)	081085	
Station-1 calls **Station-2**. **Station-2** has bridge appearance on some other **Station** say **Station-3**. **Station-3** answers the call using bridge appearance button. **Station-3** extends the call and goes on-hook. Before **EC500 Station** answers the call, **Station-3** presses the extend call button again to cancel the extended call and answers the call using bridge ppearance button. After **Station-1** and **Station-3** are done talking, **Station-3** drops the call. But **LED** and display on **Station-1** is not cleared.	081127	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***5 of 46***

Problem	Keywords	Workaround
If incoming network call with **CID** is transfered over **SIP** trunk, called **Station** shows transferring **Station**'s name on `line2` along with calling number on `line1` of the **Japan Terminal 24 Station**. This problem is irrespective of type of the server and media-gateway when incoming network call is transferred over a private **SIP** (Communication Manager to Communication Manager) trunk. This problem is introduced with the **SIP networking feature**.	081181	
When a **PSTN** call was involved in a three-way conference and one **Station** drops out of the conference and the remaining **Station** transfers the call to a remote **Station** via **QSIG**, the **PSTN** number was not displayed on the remote **Station** via **QSIG**.	081207	
If the Application Enablement Services (**AES**) server reset, then the ensuing **AES** endpoint registrations could cause a reset of Communications Manager.	081266	
If a call is made to a Administered Without Hardware (**AWOH**) type of **Station** and later covered, then the call may get dropped.	081274	Use normal **Stations** instead of **AWOH Stations**.
When a call is made to an **IP DECT Station** on Communication Manager which is a member of pick-up group and pick-up alerting feature is enabled then pick-up buttons at other members of the group start flashing even if the **IP DECT Station** is switched-off.	081303	
When a call was to a **96xx** phone, which was unregistered and had a coverage path to a **SIP** voice mail, the Call would go to voicemail according to the coverage path. Then log in this **96xx** extension, the line appearance would start to indicate incoming call without ringing. If try to answer, you would just hear the dial tone and nothing else.	081339	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***6 of 46***

Problem	Keywords	Workaround
Call pickup continues alerting without showing **SA8622** display alert when the coverage point is member of the pickup group. To reproduce the problem enable **SA8622** green feature and enable Send All Calls Configure **Station-A** with Send All Calls enabled with call sent to **Station-B**. After call covers to **Station-B**, display no longer alerts on Station-**B**, only its pickup button blinking.	081343	Disable Send All Calls.
After doing connection preserving upgrade when `list integrated-annc-boards` command is executed on SAT; it shows zero length even though announcements are administered in v9 slot of server.	081366	
When command **change variables** is executed to change the value for variables tod, dow or doy the values assigned to these variables in Start Assignment column was not saved correctly, therefore the values would not show up correctly when command **display variables** was executed for pages >= 15.	081367	
If an inter-network-region call triggered Inter-Gateway Alternate Routing, and the trunk selected for **IGAR** was **ISDN**, the calling number in the **ISDN** Setup message was always based on the **IGAR** Local Directory Number for Network Region one, instead of the **IGAR LDN** for the actual calling Network Region. Since some Service Providers required the latter calling number, the **IGAR** call was denied.	081370	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***7 of 46***

<table>
<tr><th>Problem</th><th>Keywords</th><th>Workaround</th></tr>
<tr><td>The customer would have no talkpath in some cases.

Typically the configuration would be like
the one explained below:

There are two switches. One switch must have at least two port networks or a port network and a media gateway. The second switch must only have media gateways. Finally, there must be a SIP direct trunk between the two switches as shown below.

Here is a typical configuration:

PN1 <--> PN2 -----SIP Trunk----> GW1-----------GW2
| (Network Region 1) (Network Region 2)
| |
| |
| |
A(DCP) B(DCP)

Station A calls Station B.

The Far end NR field on the SIP trunk from is kept blank on both Communication Managers.

There is no talk path after B answers the call. This happens every time in such a configuration.</td><td>081395</td><td>Use an IP trunk instead of the SIP trunk.</td></tr>
<tr><td>The BTD trunks involved in a conference call were not disconnected when the corresponding far end party dropped. Then, the BTD trunks involved in a conference call get dropped when the corresponding far end party drops.

This problem has been fixed only for BTD trunks on GWs, and not for BTD trunks on PNs.</td><td>081460</td><td></td></tr>
<tr><td>Rarely, the TN775 EPN Maintenance Board would get into an alarmed state that could be cleared only to return in 15 minutes or less.</td><td>081481</td><td></td></tr>
<tr><td>Using the System Logs page, part of the Communication Manager Maintenance Web Interface, would generate log entries for an undefined variable - iviewList.</td><td>081508</td><td></td></tr>
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</table>

Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***8 of 46***

Problem	Keywords	Workaround
In case of a priority call to a hunt group, if the first idle Station of hunt group has **call-forward-busy-dont-answer** (**CFBDA**) activated on it, then this Station along with the next idle Station, both ring in parallel and the call can be answered only from the next idle Station. And the Station with **CFBDA** rings continuously for two mins even after the call is answered. This occurs always when a priority call is placed to a hunt group with two Stations in it and the first Station has **call-fwd-busy-dont-answer** (CFBDA) active on it.	081561	When this problem occurs, restart the Station which is ringing continuously or wait for two minutes.
Whenever a third party **SIP** gateway sent packetisation period as non default value,Avaya Communication Manager answered with the same non default value.However the internal frame rate/packetisation period may have been configured with the default value. This resulted in jitter buffer overflow and bad audio quality. This was corrected so that Avaya Communication Manager sent the configured packetisation period to the remote.	081565	
When **Communication Manager** receives out of order **SIP** messages coming from **Avaya Session Manager**, calls are not getting completed and Communication Manager is not responding to **SIP** request.	081572	
If a call is made between two Communication Managers over a SIP trunk from a **DCP** or **H.323** Station to a Station that is busy, the callers Station may be left in an unstable state for several seconds. This fix allows the caller's Station to return to a normal, stable state and allow outgoing calls.	081576	
Earlier the call used to get routed, as the precedence was given to **Station lock COR** feature over the **Terminating Extension Group** (**TEG**) irrespecive of call restrictions. With this fix, if the call has restrictions it will consider those restrictions and will not allow the call to get routed. This problem always arises when **term-ext-group** feature is used and if the Station **Lock COR** field in **COR** setting page two is set to different **COR**. This issue exists from day one.	081589	
When a **SIP Station A** calls **SIP Station B**, the name of **Station A** if it is longer than 15 characters is truncated to 15 characters on **Station B**.	081598	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***9 of 46***

Problem	**Keywords**	**Workaround**
When User dials a restricted or invalid number on **Japan Terminal 24** phone, announcment extension number is displayed on line one. In this scenario, line one of **Japan Terminal 24** phone should be blank.	081618	
When any user makes a call using **Priority** button (administered on **change Station XXXX** form) to **Avaya Digital Terminal for Japan** user on the called side sees **To Priority** (in Japanese) on fourth line, instead of just **Priority** (in Japanese). This problem is seen only on **Avaya Digital Terminal for Japan** phone.	081622	
For incoming **H.323 IP** trunks administered to use the **LRQ** (Location ReQuest) option, if the network region of the incoming trunk (found on the sig-grp form, far-end network region field) and the network region of the call destination do not have an IP codec set defined between them, Communication Manager assumed that there was no way to connect these two entities, causing the call to fail (or be re-routed by the originating system).	081630	
Customer is using digital CallMaster set and connected to a H.248 gateway with auto answer enabled for **ACD** (Avaya Call Distribution). Call the **VDN** (Vector Directory Number) extension from either a **DCP** set attached to the **PN** (Port Network) or from an **IP** set which uses it's PN's medpro to establish the talkpath/RTP with H.248's VOIP. While the CallMaster autoanswers for **ACD** call there is a `ZIP2` tone is played back to CallMaster. Put CallMaster on hold during the `ZIP2` tone (duration of this tone is ~1300msec). Unholding the call will not have talk path	081640	Disable the autoanswer on digital CallMaster.
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***10 of 46***

Problem	Keywords	Workaround
It involves two **Communication Manager Servers** which are linked to each other via a private **SIP trunk**. Station A is on Communication Manager one and Station B is on Communication Manager 2. Both Stations A and B are **Avaya Digital Terminal for Japan**. Station A placed a call to Station B. Station B did not answer hence Station A activated the **Automatic Call-back** button, so that it will be alerted whenever there is any activity on Station B. As expected Station A received an alert when Station B is available. During this alert Station A should show Station B's extension on the first line of display and Station B's name on the second line of display but Station A's display just shows Station B's name on the second line and the first line is blank.	081679	
DCP-1/IP-1 Station calls other **DCP-2/IP-2** and parks the call. **DCP-2/IP-2** mentioned here can be on the same Communication Manager or on other Communication Manager connected via private **SIP trunk**/**ISDN** trunk(with **send name** and **send number** enabled). While unparking the call, it shows only connected name and not connected number. Problem was happening only with **DCP/IP** Stations and not for **SIP** Stations.	081681	
This problem is specific for Avaya Digital Terminal for Japan. It also involves two **Communication Manager Servers** linked via a private **SIP trunk**. **Station A** and **Station C** are connected to Communication Manager one and **Station B** is connected to Communication Manager 2. All these **Stations A** ,**B** and **C** are **Avaya Digital Terminal for Japan**. **Station A** called **Station B** and **Station B** answered. **Station A** then transfered the call to **Station C** and completed the transfer even before **Station C** answered. **Station C** did not answer the call, so after a fixed amount of time, **Station A** received an alert or transfer recall. When S**tation A** was being alerted with the recall, the display on **Station A** went blank and the **LED** of the call-originating button on which the **Station** was receiving the alert went OFF.	081684	
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Problem	Keywords	Workaround
If incoming SIP trunk call is conferenced with a local Station, TAC was displayed on **Japan Terminal 24** Station's `line1` which is on the other side of the trunk. This problem is irrespective of type of the server and media-gateway when incoming **SIP trunk** call from a **Japan Terminal 24** Station is conferenced over a private SIP trunk. This problem is introduced with the **SIP networking** feature and there is no work-around for this.	081699	
If incoming network call with **CID** is conferenced over SIP trunk, **Japan Terminal 24** (**DCP 2420J**) Station which initiates the conference shows **TAC** (Trunk Access Code) on line1 of display. This problem is irrespective of type of the server and media-gateway when incoming network call is conferenced over a private **SIP trunk** from a **Japan Terminal 24** Station. This problem is introduced with the **SIP networking** feature.	081700	
When an H.248 media gateway with **ISDN BRI** trunks administered returned from a link bounce, maintenance was turned off on all of the ports on the **BRI** board. This had the adverse effect of not bringing the trunks into service. The trunks would stay out of service forever until a busy/release trunk-group sequence was done. The trunks would come into service with the busy/release trunk-group, but maintenance still would not run correctly because all of the maintenance objects on the **BRI** board had been disabled.	081720	
This problem is specific for the **Avaya Digital Terminal for Japan** also known as **Japan Terminal 24**. A Japan Terminal 24 Station is configured with Bridged Appearance of two **AWOH** Stations. This digital terminal receives a call on the first Bridge Appearance. While that bridge appearance is still ringing, it receives another call on the second bridge appearance, and the display of the digital terminal is updated to show the information of the second call. It is expected that it continues to show the information of the oldest call.	081725	
If **DCP Station A** was on a different Media Gateway than **DCP Station B**, and **Station A** transferred **Station B** to **DCP Station C**, and **Station C** didn't answer, and the first coverage point of **Station C** is an **IP trunk**, that covered call would drop.	081727	
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Problem	Keywords	Workaround
If incoming **SIP trunk** call is transfered to a local Station, called Station shows transferring Station's name. This problem is irrespective of type of the server and media-gateway when call scenario involves private **SIP** (Comunication Manager to Communication Manager) **trunk**.	081729	
Call is placed over S**IP trunk** and locally blind transfered by originator. Transferred-to party doesn't answer the call and originator gets transfer recall after transfer recall timer expiry. At the time of transfer recall, Station that originated the call, displays only name of connected party but not connected number. This problem is encountered only if the originating party is a **SIP Station**.	081730	
If incoming network call is transfered over SIP trunk, called Station's display was not updated with incoming number after completing the transfer. This problem is irrespective of type of the server and media-gateway when incoming network call is transferred over a private **SIP** (Communication Manager to Communication Manager) **trunk**. This problem is introduced with the **SIP networking** feature.	081732	
Two Communication Manager Servers are connected via a private **SIP trunk**. First Comunication Manager has one **Toshiba SIP Phone** (**TSP**) **A** and the Second Communication Manager has Toshiba **SIP Phones B** and **C**. **TSP-B** is configured with coverage path to another local extension **TSP-C**. **TSP-B** has **Send-All-Calls** activated so any calls that come in to **TSP-B** will be covered to **TSP-C**. When **TSP-A** dials **TSP-B**, **TSP-C** is notified as expected. When **TSP-C** is ringing, the display on calling party (**TSP-A**) should show information like number and name of **TSP-C**, but instead it shows name of **TSP-B**. Line-1 on **TSP-A** is blank.	081734	
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Problem	Keywords	Workaround
A customer who made a call from a Path Navigator registered video system (For example, a Polycom **VSX**) via a Communication Manager video trunk to an audio-only endpoint registered to a second Communication Manager would never get audio. The audio endpoint would hear nothing, the Polycom **VSX** would hear continued ringback. The connection looks like this: **VSX** - **PathNav** - **CM1** - **CM2** - **IPT** (**IPT** = **IP telephone**, an audio-only endpoint).	081738	• Turn off video on the trunk between Communication Managers. • Set up bandwidth management so that the Polycom **VSX** requests more video bandwidth than the trunk supports; the resulting bandwidth negotiation phase fixes the problem. • Don't call audio endpoints via trunks from Path navigator systems (this is not a particularly useful thing to do as the point of Path Navigator is to support video).
Station A calls **Terminating Extension Group (TEG)** extension. The call is then redirected to **Station B** which is a member of **TEG**. If **Station B** supports the local call log feature, then **Station B**'s call log information shows number as **Unavailable**. Due to this, user is unable to call back the caller at a later time, since number is shown as **Unavailable**	081747	
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Problem	Keywords	Workaround
St-D is caller, **st-A**, **st-B**, **st-C** are member of same pkup-grp #1. **St-A** has a coverage path to **st-B** then to **st-C**. **st-B** is already on a call. **St-D** calls **st-A**. After 5 seconds, display on st-B and st-C shows the enhanced pickup display **st-D** to **st-A**. After coverage timeout calls covers to **st-B**. Enhanced pickup display on **st-B** is vanished <== Expected. After the coverage timeout the call covers to **st-C**. **st-B** does not show Enhanced pickup display <== Problem. When Enhanced Call Pick display is configred and if the call placed to a pickup group member whose coverage point is another member of the same pickup group, this issues appears.	081759	
Two parties **SIP** calls did not shuffle when using **G.726** codec.	081767	
When Call Center agents tried to **login** to the traditional **ACD** (not Expert Agent Selection) in Communication Manager 5.0 by using an entry from their Personal Abbreviated Dial List, the **login** operation was denied. The Abbreviated Dial List entry was defined as *41000522101, where: ***41** is the **Agent Login Feature Access Code (FAC)**, **0005** is the **ACD Hunt Group** **22101** is the **CMS/BCMS Login ID** Using the same Abbreviated Dial List entry in Communication Manager 4.0 worked well. To avoid this problem in Communication Manager 5.0, agents should login to the **ACD** by manually entering the **FAC** and the rest of the information. As an alternative, the Abbreviated Dial entry could contain only the L**ogin FAC** and **ACD Hunt Group**, but the **Login ID** would need to be entered manually.	081771	
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Problem	Keywords	Workaround
When Call Center agents tried to **login** to the traditional **ACD** (not Expert Agent Selection) in Communication Manager 4.0.3 by using an entry from their Personal Abbreviated Dial List followed by the manual entry of the **CMS/BCMS Login ID**, the system experienced a restart. The Abbreviated Dial List entry was defined as *410005, where: ***41** is the **Agent Login Feature Access Code (FAC)**, **0005** is the **ACD Hunt Group** Using the same Abbreviated Dial List entry in Communication Manager 4.0.1 worked well. To avoid this problem in Communication Manager 4.0.3, agents should login to the **ACD** by manually entering the **FAC** and the rest of the information.	081783	
When attempting to conference a new party into a call, if it went to coverage just when the conference is completed and the coverage is to a **VDN**, the conference would complete even though you cannot conference a **VDN**. This would cause problems with call center reporting (**CMS**).	081784	Complete the conference before the call goes to coverage.
A **Public Switched Telephone Network** caller routed through a Network Call Redirection route-to vector step to another **Public Switched Telephone Network Station** was not dropped automatically if Network Call Redirection invocation failed and the called Station dropped the call.	081787	
All the Stations are **SIP** phones. **Station A** calls **Station B**, **Station B** answers the call and can talk to **Station A**. **Station B** presses the hold button and puts **Station A** on hold. **Station A** can not hear the Music on Hold (**MoH**). This problem is encountered on following conditions: 1. When both the endpoints are in different network region. 2. One of the Network regions has integrated music source & other has Analog music source.	081816	Use integrated music instead of Analog music.
When a **DCP** phone or **ISDN-BRI** Phone (A) was on a call with an IP phone (B), and the IP phone transferred the call to another phone (C) using the transfer button, and IP phone D picked up the call using the call pickup **FAC**, the transfer completed, but there was no talk path between the **DCP** Phone (A) and phone D.	081823	One of the two parties on the call to hold/ unhold the call.
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Problem	Keywords	Workaround
When 'Media Encryption Over IP' on the **system-parameters customer-options** form was enabled and a call was placed from a **SIP** Phone to One-X Communicator with a **H.323** Home Station, there was no talk path between the SIP Phone and the **H.323** Home Station. **Direct IP-IP Audio Connections** was enabled on the signalling group of the SIP Phone and the **Station** form of the **H.323** Home Station.	081828	Direct IP-IP Audio Connections' had to be disabled on the Station form of the H323 Home Station.
An incoming **SIP trunk** call would sometimes fail to complete if it was routed to an **ISDN trunk** via Automatic Route Selection or Automatic Alternate Routing, and the administered minimum number of digits for **ARS**/**AAR** was less than the administered **ARS**/**AAR** maximum number of digits.	081830	
An IP Station using H.248 Media Gateway VoIP resources and using call-pickup to answer a call did not get a talkpath to the calling party.	081835	
Call vector having **oldest-call-wait <** step shows **oldest-call-wait <**" on list trace vector command.	081840	
On rare occasions, a Media Gateway was not able to register with the server although it had been registered. The **list media-gateway** command shows **p** indicating a pending registration but the Media Gateway never registers.	081845	Run the '`test media-gateway`' command.
Customers were unable to achieve a basic level of interoperability calling between Cisco and Avaya video solutions. The expection is that basic video call setup should proceed between the two vendors equipment via **standards based H.323 (H.245) video trunking** i.e. without any audio shuffling or telephony features involved. A problem had been observed when calling from **Cisco IP Communicator** (w/CUVA video) to **Avaya IP Softphone** (w/ video) via **IP trunk** to **Avaya Communication Manager**. Fast busy tone is observed. Then, this scenario connects and establishes video as expected with audio shuffling disabled.	081851	Disable video or originate all calls from the **Avaya IP Softphone** instead.
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Problem	Keywords	Workaround
Station A on Communication Manager has Time-of-day coverage configured with cover point set to **Station B** and also **Station B** has a bridge appearance of **Station A**. When a call placed to **Station A** is not answered the call would not cover to **Station B** and instead ring again on next call appearance on **Station A**. **Station A** continues to ring even after the call is dropped, and the ringing will stop when audit runs.	081860	
When a network incoming call to a pickup group member is answered using **call-pickup feature** button by another member of the same pickup group, the display is incorrect as following: ● if the pickup member has no public CPN defined, the call-appr line will show **#5 to NamePrinciple p**. ● if the pickup member has a public CPN defined, the call-appr line will show **CPNpkupmember to NamePrinciple p**. You will see this problem when you use call-pickup feature with coverage to answer the n/w call with or without **CPN** prefix defined for the pick-up member.	081865	
Under certain circumstances, systems with media gateways would see high occupancy readings, independent of the level of call processing traffic.	081870	
Customer is using digital CallMaster set and connected to a H.248 gateway with auto answer enabled for **ACD** (Avaya Call Distribution). Call the **VDN** (Vector Directory Number) extension from either a **DCP** set attached to the **PN** (Port Network) or from an IP set which uses it's PN's medpro to establish the talkpath/RTP with H.248's VOIP. While the CallMaster autoanswers for **ACD** call there is a `ZIP2` tone is played back to CallMaster. Put CallMaster on hold during the `ZIP2` tone (duration of this tone is ~1300msec). Unholding the call will not have talk path.	081875	Disable the autoanswer on digital CallMaster.
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Problem	Keywords	Workaround
IF: ● An incoming call arrives via a **TDM** trunk on a port network (**PN**), AND ● The call routes to a non-IP agent on a different port network, AND ● The agent is service-observed by a non-IP observer who is also not on the same **PN** as the originating incoming trunk, AND ● The system music source is configured to be provided via an announcement (which may exist anywhere in the system), AND ● The agent places the call on hold, THEN If the call stays on hold long enough (possibly minutes or hours), the customer on the incoming trunk may begin to hear another user overlaid on the **Music on Hold** provided by the announcement.	081876	
When an incoming call from a remote site was transferred using a speed dial number to the voice mail, the call failed. The caller heard an intercept recording from **Modular Messaging** which played back the digits received (mailbox number), missing one digit.	081881	
If an **IP Softphone** has a video call on hold and tries to make another video call to a Polycom Path Navigator registered endpoint, this second call may fail to get video. If the Path Navigator registered endpoint is called first then this problem does not occur.	081899	
When Call Center agents tried to **login** to the traditional **ACD** (not Expert Agent Selection) in Communication Manager 5.0 by using an entry from their Personal Abbreviated Dial List, the **login** operation was denied. The Abbreviated Dial List entry was defined as *41000522101, where: ***41** is the **Agent Login Feature Access Code (FAC)**, **0005** is the **ACD Hunt Group** 22101 is the **CMS/BCMS Login ID** Using the same Abbreviated Dial List entry in Communication Manager 4.0 worked well. To avoid this problem in Communication Manager 5.0, agents should **login** to the **ACD** by manually entering the **FAC** and the rest of the information. As an alternative, an Autodial button can be configured with the same information string.	081917	
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Problem	Keywords	Workaround
User-to-User Information Element in the ASAI Route-Request message may contain the incorrect protocol discriminator for the **ASAIUUI** application when vector variables were used to set the **ASAIUUI**. As a result the **ASAIUUI** was misinterpreted by CTI applications which could have resulted in misrouting a call or missing information for a call. Apparently, this behavior was intermittent. This failure only occurred when vector variables initially set the ASAIUUI. This behavior did not occur if the call already contained **ASAIUUI** or vector variables reset the existing ASAIUUI. Refer to ASAI PROTOCOL REFERENCE 03-300550 (Call Route Request message).	081919	
Repeated presses of the a **man_overid** feature button may have resulted in a system restart when configured with the following options: 1. On system-parameters customer-options, enable these two features: Tenant Partitioning? y Time of Day Routing? y 2. On **COS**, enable console permissions for a Station 3. add a button on same Station: **man-overid TOD**: 1	081923	
When a call came into Communication Manager from an **ETSI trunk** and Communication Manager tandemed the call out to the **PSTN** using another **ETSI trunk**, and the called party was busy, the calling party heard silence instead of busy tone.	081931	
Extend Call was getting droped whenenver the hangup button was pressed.The **IP-soft phone** in this case was configured with a release button.	081947	
Some types of **ISDN BRI** telephones could not originate calls when connected to an H.248 gateway. The user would hear denial tone in the middle of attempting to dial a call.	081949	
The **monitor system view1**(or `view2`)" command on SAT did not show how many pages the command output had in top right corner.	081953	
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Problem	Keywords	Workaround
When a Customer Interaction Express adjunct transferrs a call back to Communication Manager, a break in talk path, due to path replacement, will occur immediately after the transferred to party answers the call. All other path replacements not involving the Customer Interaction Express adjunct still occur in a time interval of up to 10 seconds after the transferred to party answers.	081980	
Single Step Conference invoked from Avaya Enablement Server.Single Step Conference for Toshiba **SIP** phone was not working as Communication Manager did not sent line reservation 183 Session Progress for initial Off Hook Invite.	081996	
Customers with multipoint Polycom **VSXs** and Avaya **TTS Stations** in use would see some calls from the **VSX** fail inexplicably. After one or more retries a call would eventually succeed and the rest of the call would be fine. Calls to the **VSX** would always succeed. The busier the system, and the more **TTS Stations** in use, the more likely the failure. Higher network delays also contributed to the problem.	081997	
An **H.323 IP Station** user entering digits that were then transmitted over an in-band or rtp-payload **H.323** or **SIP trunk** to an **IVR** or voice-mail-type system could lose digits if the duration of dialing exceeded 30 seconds. For example, when the user heard an **IVR** prompt asking for a password, the user might dial 1,2,3,4, then hear a 10 second annc, enter more digits, then hear another 10 second annc, enter more digits, then hear another 10 second annc, and enter even more digits. These last digits were therefore being entered ~30 seconds after the initial digits and may not have been properly transmitted, resulting in a failed **IVR** or voice-mail session.	082003	
The **SAMP** firmware update failed, if the modem was connected to the **SAMP's** USB port and incoming calls were enabled.	082015	
When a call routed to a local Station by a vector directory number (**VDN**), is forwarded to a remote **Japan Terminal 24** Station over a SIP trunk the calling **Japan Terminal 24** Station does not see the answering party's number and name.	082026	
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Problem	Keywords	Workaround
This problem is specific to the **Avaya Digital Terminal for Japan**. Two Communication Manager servers are connected via a private SIP trunk. First Communication Manager has one Station A and the second Communication Manager has two Stations B and C. All these Stations A,B & C are **Avaya Digital Terminal for Japan**. Station B has 'Send All Calls' activated, so any calls coming in on Station B will be diverted to Station C. When **Station A** placed a call to **Station B**, as expected C was notified and it answered. **Station A** then put the call on hold. After a certain amount of time, Station A was notified the hold-recall alert. But when **Station A** received this alert, the display on **Station A** went blank and the **LED** of the button used to place the call went OFF. On receiving the alert, **Station A** should show Station C's number on line one, name on line two, an alert string (in Japanese) on line three and 'cover' (in Japanese) on the fourth line.	082035	
This problem is specific for **Avaya Digital Terminal for Japan**. This involves two Communication Manager Servers connected via private SIP trunk. Communication Manager one has one Station A and Communication Manager two has two Stations B and C. All these Stations A,B and C are **Avaya Digital Terminal for Japan**. Station B has **Send-all-calls** to a Vector Directory Number which routes the calls to an Administration Without Hardware Station, so that all calls coming in on Station B will be routed to the Administration without hardware Station. Station C has a bridged appearance of the Administration without hardware Station. **Station A** called **Station B** and as expected the call was routed to the vector directory number and **Station C** started ringing at the bridged appearance. After C answered the call, display on **Station A** showed the number and name of the vector directory number. It should have shown the number of **Station C** on the first line and name of **Station C** on second line and **cover** reason code in Japanese on the fourth line of display.	082036	
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Problem	Keywords	Workaround
The **IPSI Firmware Upgrades** section of left hand menu in the Communication Manager maintenance web pages with the underlying links for download and activation of the **IPSI** firmware were not visible on Communication Manager Express servers.	082037	
This issue involves two Communication Manager servers connected via a private SIP trunk. **Station A** is on Communication Manager one and S**tation B** and C are on Communication Manager two. **Station B** has **Send-all-calls** activated to **Station C** so that all the calls that come in to **Station B** will be diverted to Station C. **Station A** called **Station B** over the **SIP trunk** and as expected the call was covered to **Station C**. The cover reason Code that was displayed on **Station C** was 'c' and not 's' or 'cover' in Japanese (for Japanese Stations).	082042	
This problem is specific for the **Avaya Digital Terminal for Japan**. Two Communication Manager Servers are connected via the private **SIP trunk**. First Communication Manger two Stations A and B and the second Communication Manager has one Station C. All these **Stations A**, **B** and **C** are **Avaya Digital Terminal for Japan**. Station B is configured with coverage path (change coverage-path x) to the remote Station C. Station B has **Send All Calls** activated, so any calls that come in on Station B will be routed to Station C via the SIP Trunk. After **Station A** placed a call to **Station B**, **Station C** was notified as expected. But, **Station A** displayed the Number and name of Station B on line one and line two. Display on Station A should be **Station C**'s number on line one and name on line two and 'cover' in Japanese on line four.	082046	
A **PRI-DECT Station** on Communication Manager is in call with another Station with two way audio. User on PRI-DECT Station presses R button (switch hook flash) to hold current call and to initiate another call thread, then if the user again presses R button or dials a number which does not answer the call and then pushes R button, instead of getting connected to the held call, the call drops.	082080	
An invalid gateway **Type** of **trm480** was allowed using the **add media-gateway** command.	082086	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***23 of 46***

Problem	Keywords	Workaround
This problem is encountered under following conditions: 1. Music on hold (**MoH**) is configured to use Analog music source 2. For **MoH** when more than one medpro is configured in the port network Station A calls Station B, Station B answers the call. Station B has an extend-call button (administered on **change Station XXXX** form)which is mapped to an **EC500** number (adminstered on **change off-pbx-telephone Station-mapping XXXX** form). Station B presses the extend-call button and **EC500** starts ringing. Station B hangs up before **EC500** answers the call. **EC500** answers the call and talks to Station A. Station A puts the call on hold and **EC500** can not hear **MoH**.	082102	Use integrated music instead of Analog music.
If an IP Station was on a conference call, under certain internal conditions, the system may encounter a reset system 2.	082119	
Under rare internal conditions during a server interchange in a duplicated environment the system may experience a **WARM Level** reset.	082121	
Station-D is mapped to **Station-B** through **EC500**. **Station-A** calls **Station-B**, both **Station-B** and **Station-D** are ringing, **Station-D** (**EC500 mapped Station**) answers the call. **Station-A** transfers the call to **Station-C**, **Station-C** does not answer and **Station-A** completes the transfer. When **Station-A** receives the recall after transfer timeout, it shows its own number and **Station-B's** name. **Station-A** should show **Station-B**'s number and name.	082123	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***24 of 46***

Problem	Keywords	Workaround
The '+' could not be administered in the vector **route-to number** step with ~r used for Network Call Redirection (NCR) in any version of Communication Manager. Therefore, the '+' character could never be sent back over the SIP NCR trunks. Note that the '+' character may only be meaningful for certain SIP service providers. As a result of allowing the administration of a '+' character as described above, the "route-to number" vector command can be any of the following: **route-to number ~r<digit-string>**, where <digit-string> can be up to 14 digits. **route-to number ~r<variable>**, where <variable> can contain up to 16 digits. **route-to number ~r+<digit-string>**, where <digit-string> can be up to 12 digits. **route-to number ~r+<variable>**, where <variable> can contain up to 14 digits Note the different limitations based on whether ~r or ~r+ is used in the vector command. Furthermore, the "Interflow VDN" field in the **Best Routing Application** form can also be administered with a '+', but ONLY if the "Net Redir?" field is set to "y", meaning that NCR functionality is enabled. In this case, the "Interflow VDN" field is of the form **+<digit-string>**, where `<digit-string>` is up to 14 digits. The NCR functionality supported by ASAI via Adjunct Route does not support the passing of the '+' character.	082140	
Notify messages (message-summary) bombarded on Communication Manager from **SIP Enablement Server** caused overlaod condition.This fix resolves in send 503 Service Unavailable message from Communication Manager to **SIP Enablement Server** in overload condition such that **SIP Enablement Server** stops sending any further messages.	082144	
When audit 552 (MO_FTING) runs, under some internal condition (For example, corrupt data) the system may undergo a restart level 1 and further get escalated to levels two and four. This will lead to an interchange of servers in a duplex system. During this process calls may get dropped.	082146	Disable audit 552.
In Communication Manager a **CLAN** (**TN799**) would not come into service if the only **CLAN** ethernet link/data module was removed, re-added, and re-enabled.	082151	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 *25 of 46*

Problem	Keywords	Workaround
The user of a **Station A** dialed the number of a Terminating Extension Group which consisted more than one member (B and C), whereas every member of the group was a monitored Station, which was supervised by monitoring **Station D**. Both Stations of the Terminating Extension Group were ringing and the user of monitored **Station C** answered the call. **Note**: **Station C** was not the first entry in the group list. If yes, there was no problem existing. The call was cut through between **Station A** and **Station C** and all other Stations stopped ringing. On the monitoring **Station D** the **Team** button of the answered **Station C** indicated busy state. **Station A** dropped the call by going on-hook. Both Stations (A and C) were in idle state but the monitored Station C was further indicated as busy on the monitoring **Station D**. Note: When **Station C** was dropping the call there was no problem existing with the **Team** button indication. A Terminating Extension Group exists with Group member 1 = **Station B** Group member 2 = **Station C** Station D has got two **Team** buttons: one points to monitored **Station B** and one points to monitored **Station C**.	082164	Going of hook and on hook of the monitored Station corrects the wrong **Team** button indication.
User of **Station A** dialed the number of a Paging Group which consists more than one member (B and C), whereas every member of the Paging Group was a monitored Station, which was supervised by monitoring **Station D**. The call was cut automatically through between **Station A** and all Stations in the Paging Group, whereas none of the Stations were indicated as busy on the monitoring Station. A Paging Group exists with Group member 1 = **Station B** Group member 2 = **Station C** Station D has two **Team** buttons: one points to monitored **Station B** one points to monitored **Station C**.	082169	
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Problem	Keywords	Workaround
User of **Station A** dialed the number of a **Station B** which had in his active coverage path (Coverage Criteria: Don't Answer) a Coverage Answer Group instead of an extension number. The Coverage Answer Group consisted more than one member (C and D) which were supervised by monitoring **Station E**. The termed **Station B** started ringing but its user didn't answered the call. After timeout the call was routed to the Coverage Answer Group and all members in the group started ringing. The monitored **Stations C** in the Coverage Answer Group answered the call, the call was cut through between Station A and Station C and the other Stations stopped ringing. On the monitoring **Station E** the **Team** button of the answered Station C didn't indicated busy state. A Coverage Answer Group existed with **Station C** and **D**. **Station B** had an active coverage path (Coverage Criteria: Don't Answer) with a Coverage Answer Group as first coverage point. **Station E** had two **Team** buttons which pointed to monitored **Stations C** and **D**.	082174	.
If an incoming call arrives on a SIP trunk and the call then goes out as an outgoing call, again over a SIP trunk, first due to the call forwarding offnet feature, and then the call goes out a second time due to the remote coverage feature, the call would fail. If the scenario occurred frequently, this would escalate to system restarts.	082192	
When a user either called SIP voice mail directly, or reached SIP voice mail via a coverage path, the message sent to the SIP Modular Messaging server did not contain the voice mail handle as it was administered on the Communication Manager hunt group. This led to **SIP Enablement Services** not being able to route the call properly to voice mail and these calls failing.	082213	
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Problem	Keywords	Workaround
The User-to-User Information Element in the **ASAI** Route-Request message could contain the incorrect protocol discriminator for the **ASAIUUI** application when vector variables were used to set the **ASAIUUI**. As a result the **ASAIUUI** was misinterpreted by CTI applications which could have resulted in misrouting a call or missing information for a call. Apparently, this behavior was intermittent. This failure only occurred when vector variables initially set the **ASAIUUI**. This behavior did not occur if the call already contained **ASAIUUI** or if vector variables reset the existing **ASAIUUI**. Vector variables will set the protocol discriminator as IA5 characters when no **ASAIUUI** exists with the call. Otherwise, vector variables will use the existing protocol discriminator, if ASAIUUI exists for the call and vectors are used to modify the contents of the **ASAIUUI**. Refer to ASAI PROTOCOL REFERENCE 03-300550 (Call Route Request message)	082257	
When an IP Softphone was administered on Communication Manager with a coverage path to SIP Modular Messaging (MM), but this softphone was not registered (no endpoint had logged into this extension), a call to this softphone extension would cover to MM, but MM was unable to answer due to invalid content in the SIP message sent from Communication Manager.	082468	
It is possible that media resources will not be released when they should when making or receiving a SIP call. This fix will ensure that media resources releasing will not be delayed.	082528	
Station A receives an incoming call from a service provider via a **SIP trunk**. **Station A** initiates an attended transfer to **Station B** on another switch. The call is routed out back to the service provider via a **SIP trunk.** After the call is ringing or answered by **Station B**, **Station A** completes the transfer. The call **may** be incorrectly disconnected after 30 seconds, dependant on the service provider's **SIP** implementation. (All trunks are shuffled.)	082611	
For the calls originated by Telecommuter Softphone using feature buttons such as **Last Number Dial**, and **Abbreviated Dial**, Autodial dialed the destination number without waiting service link to come up.	061716	
Crisis alert calls to attendent consoles went through tenant partitions.	071106	
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Problem	Keywords	Workaround
If the caller dialed a final # digit for an outgoing **ISDN** overlap sending trunk call before the far end sent ALERT, then that # digit was sometimes outpulsed when the call was answered, even though the trunk group field **Suppress # Outpulsing** was enabled.	073701	
A corrupted non-**ACD** hunt group member record was being saved in translations. On a subsequent system re-boot the corrupted record was recognized causing translation corruption. Once translation corruption occurred, the system was not allowing the saving of translations.	074222	Contact the Services team to clean the corrupted data. However, this would not prevent the corruption from re-occurring.
In certain call scenarios, calls covered to a **Modular Messaging** adjunct had encountered a non-integrated greeting, instead of getting a specific party's voice mail box.	074295	
Personal Station Access operation failed for a **DCP** phone (2420) when login attempt from a soft phone was done with a short extension.	080064	Personal Station Access succeeded only when DCP Phone attempted to login to softphone if registration was attempted with a complete long extension.
With **SIP** station A on switch 1 and **DCP** or **IP** station B on switch 2, where an **ISDN** trunk connected switch 1 and switch 2, and where national and international prefixes (country or state codes) were configured to use in the calling party number, when station B called station A, these prefixes were not appended in the call log. Thus, station A could not recognize the user of station B completely and could not call back using the call log entry.	080468	
The end-to-end **DTMF** (Dual Tone Multi-Frequency) signaling over an **ISDN** trunk with calls involving Crisis Alert was sometimes incorrect causing the call to fail.	080479	
Reduced **USB** alarms on **S8500** B/C and **S8400** servers.	080942	
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Problem	Keywords	Workaround
Incoming trunk calls from an **EC500** mapped mobile user to a busy station on **Communication Manager** did not provide busy tone to the mobile user.	081045	
If a call from user A to user B was transferred by user B via **QSIG** to user C in ringing state, and after the transferred call was released by user A (C was still in ringing state), the missed call log of user C showed name and number of user B instead of user A.	081048	
An external application, for example, **AVAYA Softconsole OSPC** expected **CTI** events if a call was picked using the **Team Button Pickup** functionality. The application in this case the **AVAYA Softconsole OSPC** could not change the displayed status of a phone due to the missing events and presented the phone permanently in a busy state.	081067	
In case of call forwarding Busy or Don't Answer (DA), and Enhanced Call forwarding Busy or DA, **Communication Manager** did not send Adjunct Switch Application Interface (**ASAI**) events of forwarded call. Hence users did not get an alert message on their application for the redirected call.	081068	
When a **DCP** station unplugged and plugged into another (or same) port, it used to display button-labels and the station extension administered on the previous port. Buttons used to function as per administration for the port, though. Only the labels did not show up. **Note:** When **DCP** station was plugged back within 6 seconds, then the button labels would not get updated.	081094	
When a call on a **SIP** station that had the **Confirm Answer** option set was taken off hold, it failed to reconnect the voice path.	081197	
The call from different multinational locations did not shuffle eating up media resources. To reproduce this problem, Enabled multinational feature and configured 2 locations with different location-parameters and tone generation plan. Configured a port network in region 1 location 1 and a gateway in region 2 location 2. Then **DCP** phone in location 1 called **IP** phone in location 2 which had auto-answer enabled for all calls. The call was answered, but was not shuffled.	081286	
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Problem	Keywords	Workaround
There were display issues for minor digits in firmware version on **status station <Extension>** and **list registered ip-stations** form.	081301	
Abnormal temperature readings for the **S8710**, **S8720**, and **S8730** servers were indicated only by a major alarm trap. A new minor alarm trap will now be sent providing earlier warning of such a condition.	081390	
Following problems were reported for Station 64XX series: 1. When **Headset** button was ON, calling party information was cleared after 30 seconds even if call was ringing. 2. The 64XX station sets with the **Headset** button ON, were not updating their display after getting a missed call 3. Two calls were ringing on a 64XX station with **Headset** button administered. First call was answered with **Headset** button and transfered to another station. After transfer completed, station 64XX showed time and date even though second call was ringing.	081420	
Communication Manager could try to bring up a socket to the Time-To-Service (**TTS**) phone forever if the socket establishment to the phone failed due to any reason.	081537	
When a user made a call across a **SIP** trunk to an extension that did not exist on the far end, and the far end had an announcement administered in the **DID/Tie/ISDN/SIP Intercept Treatment:** field on the **system-parameters features** page, the call originator should have heard this administered announcement. Instead, intercept tone was heard.	081587	
Incoming calls to media gateways and Outgoing calls from media gateways were getting blocked due to resource exhaustion.	081611	
B-Channel *out of service coming back into service* state was not reported to **CMS**.	081633	
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Problem	Keywords	Workaround
Personal Station Access operation failed for a sage (**16xx H323 IP**) phone when login attempt from a soft phone was done with a short extension. Personal Station Access operation also failed for an IP hard phone when login attempt from another **IP** hard phone was done with a short extension. Personal Station Access operation succeeded for a **DCP** phone when login attempt from an **IP** soft phone was done with a short extension.	081659	Personal Station Access succeeded when **IP** Phone (Softphone/ hardphone) attempted move to another **IP** Phone (Softphone/ hardphone) and if registration was attempted with a complete long extension.
On a system with Modular Messaging with a **SIP** Station administered as receptionist, and the caller and the receptionist were on the same Communication Manager home (the call was not an incoming call over a **SIP** direct trunk), when a user called into Modular Messaging and was transferred to the receptionist by selecting the option for help or staying on the line after all the options were given, sometimes the caller was dropped. This issue is resolved in Modular Messaging 5.0.	081695	Change the receptionist Station type to a **DCP** Station.
Calls received by call-center agents in auto-answer mode from an Interactive Voice Response (**IVR**) unit had no *talkpath*, and those calls would drop when transfered to a G860 gateway.	081717	Place agents in manual-answer mode.
Attendant was not receiving second wakeup reminder call for *vip-wakeup*. This problem was occuring only when **do-not-disturb** button was activated on guest station and **Cancel Do-Not-Disturb for Wakeup Calls?** in **system-parameters hospitality** field was set to **y** and **extended Do-not-disturb** was set to terminate at the same time as that of vip-wakeup	081749 082507	Deactivate **do-not-disturb** button throughout OR Set **Cancel Do-Not-Disturb for Wakeup Calls?** to **n**
Call appearance on **IP** phones got stuck after system came out of Local Survivable Processor (**LSP**) mode. When user went offhook after this, there was no dial tone.	081751	
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Problem	Keywords	Workaround
When a customer experienced a network outage that persisted beyond 30 seconds to a minute and employed the IPAgent soft agent using the **Automatic Answer** feature, then the first call after the IPAgent recovered, had to be answered manually.	081756	
When **misoperation alerting** was turned ON, calls to voice mail or coverage point did not drop intermittently, hanging the port. Steps and administration to reproduce the problem: 1. Enabled **misoperation alerting**, **Don't Answer Criteria For Logged Off IP/PSA/TTI Stations** and **Intercept Treatment On Failed Trunk Transfers** on **system-parameters features** form. 2. Analog station called another station, which did not answer the call. Then analog station put the call on hold and called x-ported analog station, which had coverage path administered. 3. Coverage point was ringing, however analog station hung up. 4. The coverage call was not dropped.	081785	Turn off **misoperation alerting**.
The problem occured when a system had **EPNs** failed over to an **ESS**. When the MAIN was administered with auto return set to scheduled and the control network remained faulted through the scheduled time, the system would continue to be fragmented (that is, **EPNs** would remain under control of the **ESS**). However, the **EPNs** would be taken over by the MAIN at a subsequent time when the network was repaired. This was unintended, since **ESS** scheduled return was designed to be a one time event.	081795	
Agents intermittently did not hear zip tone or did not get *talkpath* after retrieving a call from hold.	081896	
When a **SIP** user did an attended transfer of an incoming call to a hunt group, which had a **SIP** endpoint as its member, then the call might drop at the hunt group member on completion of this transfer.	081901	Make the **SIP** endpoint hunt group member station type 4620SIPCC or 16CC.
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Problem	Keywords	Workaround
A race condition existed between a new music-on-hold (MoH) connection being established across network regions via **IGAR** and music being disconnected from other calls. When an **IGAR** connection for MoH was in progress and music was disconnected from other calls, if there were no other MoH listeners active in the system, the in-progress **IGAR** MoH connection was not established correctly. Afterward, MoH requests between the network regions of the impacted **IGAR** connection failed to hear music.	081932	
Look Ahead Routing (**LAR**) did not take place for Internet Protocol (**IP**) and Integrated Service Digital Network (**ISDN**) trunk calls during the Alternate Route selection (**ARS**) digit conversion process.	081945	Remove the Automatic Alternate Routing (**AAR**) access code on the **feature-access-codes** form.
An extra character that was erronously displayed next to the **CDR EC500** field on the `list off-pbx-telephone configuration-set` output was removed from the list output.	082000	
If a **SIP** trunk between a **Communication Manager** and a **SIP** Enablement Server was used for outgoing calls, then the calls could have dropped approximately 3 minutes after they were established.	082061	
Calls using an **IP** softphone and a permanent service link to a **SIP** hard phone did not have *talkpath*. The first call made using the **IP** softphone worked, but subsequent calls made while the **SIP** hard phone was left off-hook did not have *talkpath*.	082070	
Communication Manager in certain cases disallowed the transfer of an active call, when that call was established via **QSIG** diversion with rerouting.	082090	
If certain character combinations such as **%d**, **%c**, or **%s** were administered on the **display-messages** forms, then **Communication Manager SAT** (System Access Terminal) session would terminate and eventually **Communication Manager** would reboot. .	082126	Avoid the use of any of these special keywords in user-defined message translations.
During a **SIP** call, **Communication Manager** reset because of an internal error.	082152	
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Problem	**Keywords**	**Workaround**
When **Headset** button was ON for an incoming call, the calling party information display was cleared after 30 seconds even as the call was ringing.	082153	
If an incoming **QSIG** call was transferred into a **VDN**/vector that had as its first step **wait step hearing music**, **Communications Manager** sent an incorrect indication to the calling user via **QSIG** that, the call was alerting instead of answered.	082196	
Extend call was getting dropped on pressing hangup from **IP**-Sofphone.	082206	
The **Communications Manager** could go through warm or cold 2 reset due to one internal software infinite loop.	082221	
Outgoing **ISDN** Trunk calls monitored as *Adjunct Switch Application Interface Domains* were not logged as connected even after the calls made were answered.	082292	
After a **Communication Manager** event that loads translations, the **ip-network-region** form, region **250 AGL** (Alternate Gatekeeper List) field entries that were blank had zeroes for rows that had values in the codec set field. The zeroes in the AGL fields could cause softphones not to register under certain conditions.	082311	
During recovery of thousands of Time-to-service (**TTS**) phones, requests to establish sockets to the phones overwhelmed the **TN799** (**CLAN**) board. In extreme conditions the **CLAN** reset, forcing recovery of all the existing sockets on the board. The reset of the **CLAN** board delayed the recovery of the phones.	082315	
G723 codec with variable bitrate supported.	082316	
A denial event was reported sometimes while listening to a music source. This occured only when: 1) Tenant partitioning was enabled. 2) **Facility Access Test** feature was used to listen to the music source. 3) Music sources were not stored sequentially.	082320	
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Problem	Keywords	Workaround
Multiple link bounces of a gateway caused problems with recovery of D-channel links for **ISDN PRI** interfaces. The second link bounce got interpreted as a short link bounce and caused **Communication Manager** and the H.248 media gateway to go out of synchronization.	082328	
Ringback was not turned OFF when Inter-Gateway Alternate Routing (**IGAR**) calls from a media gateway covered to **Modular Messaging**.	082333	
If an **H.323** set called another **H.323** set that was busy and the call covered to a **DCP** set, pressing **DTMF** digits from the calling **H.323** set was not heard by the **DCP** set. If the DCP set were a voice mail endpoint, the voice mail coverage would fail.	082341	
The top line of **IP** phone display changed from short digit form to long digit form after changing password from **Communication Manager** System Access Terminal (**SAT**).	082357	
Under specific conditions, every station in a call (for example, the primary parties plus bridged users) might not alert when the (**IGAR**) feature was invoked. Scenario: Station A had bridged stations B and C. A separate call was made to Station D. The user for Station D pressed the **Transfer** button and called Station A. IF: 1) Station D's call to Station A invoked **IGAR** in order to connect to Station A and its bridged stations, AND 2) Station D completed the transfer before the **IGAR** connection had finished establishing, AND 3) The resulting call (the call without Station D in it) did *not* require the **IGAR** connection, THEN only Station A would alert, none of the bridged users would.	082371	
When a call was made to an **IP DECT** station in Location 1 from another station in Location 2 using **IGAR**, a delay of four to five seconds was observed in *talkpath*. Call from **IP DECT** in Location 1 to another station in Location 2 worked fine.	082376	
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Problem	Keywords	Workaround
Call to listed-directory-numbers (**LDN**) failed when administerd above 15th position Steps to reproduce: Adminstered listed-directory-numbers at 16 position or above administer attendent-group, Tenant and listed-directory-numbers to route to attendent, enabled night service on attendent, and made a call to listed-directory-numbers group. Then call failed.	082389	
If a call had been placed on hold from a **SIP** phone and you tried to answer the same call from a different **SIP** phone with a bridged appearance of the extension originally called, then there was no *talkpath* and the caller remained on hold.	082440	
When a call was picked up by a station in the same pickup group as the called party, the display on calling station showed the name and number of the called party. It should have shown the name and number of the answering party. This issue was specific to **Toshiba SIP Phones**.	082467	
Users were unable to activate/deactivate **EC500** through Telecommuting Access Extension, which is administered in a vector route-to step. This problem used to occur when the users used to call a Vector Directory Number (**VDN**) and the vector corresponding to this **VDN** had a telecommuting access extension administered in its vector route-to step to enable/disable **EC500** along with unequal min and max values administered on the Alternate Route Selection (**ARS**) analysis form.	082473	
When a service observer used to observe an active call on the agent wherein the agent was involved in a single step conference then the service observer used to display **calling to Conference 2 so**, but it should show **calling to called so**.	082475	
The **change modem settings** button on the configure modem web page of configure server failed with an error when one of the newer **Avaya** supported modems was attached to the server.	082485	
When the `trace-route ipaddress` command was executed repeatedly, the **SAT** session hung and the **Communication Manager** went through a warm restart.	082494	
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Problem	Keywords	Workaround
Station A on **Communication Manager** had a coverage path, which had a cover point as coverage answer group. That coverage answer group had all its member as **IP DECT** station. When an external call was made to station A, it went to cover. When no one answered it and if any of the **IP DECT** stations in coverage answer group were switched off then rest of the **IP DECT** members of the coverage answer group would not be alerted.	082497	
The following message was displayed when the **Network Time Sync** maintenance web page was selected, **SYSTEM ERROR:No date to display**. With this fix, the **Network Time Sync** web page and the `ntpq -c pe` conmmand now show the expected NTP time sync information.	082510	
If **mg-recovery-rule** on **MEDIA GATEWAY AUTOMATIC RECOVERY RULE** form was changed to **blank**, translations were saved and reset system 4 was executed, the **LSP** used to show previously administered **mg-recovery rule** instead of **blank**.	082514	
On inter-gateway incoming calls to agents, the agent might have heard *crosstalk*, if the agent did a hold and unhold of the call.	082522	
If a person on a paged station (utilizing the **Group Page** feature) pressed a button with digits associated with it (for example, autodial), other stations could potentially hear the digits. Also, if a person on a paged **H.323** station pressed digits on the keypad (for example, 1-9, *, #) during a group page, other stations could potentially hear the digits.	082540	
IP phones with **EC500** mapping did not alarm if they failed *keepalives*.	082543	
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Problem	**Keywords**	**Workaround**
Under the following circumstances, registration with **Communication Manager** failed for a softphone attempting to take over an **IP**-mapped **H.323** station capable of Time-To-Service behavior. 1) The softphone's **IP** address was not in the **ip-network-region** map (unmapped), AND 2) The **Call Server Address** on the softphone was in network region **X** (NR-X), AND 3) NR-X used the **Administrable Alternate Gatekeeper List for IP endpoints** feature (administered on the **ip-network-region** form), with >= 1 Gatekeepers permitted from its own NR-X and also >= 1 Gatekeepers from a connected NR (for example, NR-Y), AND 4) NR-Y also used the **Administrable Alternate Gatekeeper List for IP endpoints** feature, with *0* (no) Gatekeepers from its own NR-Y and >= 1 Gatekeepers from connected NR-X, AND 5) The softphone tried to take over, or share, an ip-mapped TTS-capable IP station in NR-Y.	082557	
When a call was made from one **Avaya** switch to another over an **IP** trunk, intermittently static or dead air was observed on the call when the two switches were shuffling the call at same time.	082567	
User was not able to do a **change agent-loginID** from one System Access Terminal (**SAT**) while simultaneously doing a **change vector** from another **SAT**. The error message **Transient data conflict detected, please try again** was displayed.	082569	
Call failed when an endpoint's invite only included **maxptime** for packet size negotiation in it's Session Description Protocol (**SDP**).	082598	
The **IPSI** sent a bad power supply angel ID, causing an alarm for a power supply in a **G650** cabinet that did not exist.	082599	
With **PRI-DECT** b-isdn termination on one gateway and any other kind of termination on another gateway, on doing hold/unhold on the **PRI-DECT**, caused loss of *talkpath*.	082615	
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Problem	Keywords	Workaround
On **SAT** form **change station** page 3 of a monitored station-A one kind of enhanced call forwarding was filled with an extension number which was the extension number of monitoring station-B and **Active** flag was set to **yes**. The **enhanced call forwarding** button on monitored station-A indicated that at least one kind of enhanced call forwarding was active, but the **Team** button on the monitoring station-B did not change its appearance to indicate that monitored station-A had at least one active enhanced call forwarding towards the monitoring station-B. Steps to reproduce the problem: Monitoring station-B had a **Team** button assigned that pointed to the monitored station-A. Monitored station-A had neither call forwarding active to the monitoring station-B nor send all calls with the monitoring station-B as first coverage point in its coverage path.	082648	
cpn-blk feature did not work over **SIP** trunks, exposing the caller identity to the called party.	082674	
On switches with a large number of duplicated **IPSI** port-networks, certain server interchanges that caused all **IPSI** port-networks to also interchange led to a cold port-network restart for some of the port networks.	082710	
The help message for the **Time to login** field on the **Enable Session** form stated **Enter a number between 0 to 255**. This message was not correct, as the number 255 was not valid.	082717	
When polling **Communication Manager** server with Multi-Site Administration (**MSA**) and storing station details in **MSA's** text database output, the text database did not contain station bridged appearance details.	082788	
A ping test failure was executed via periodic or scheduled maintenance had a negative effect of leaving a bad **TN2602** media processor board active in a bearer duplication configuration.	082800	
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Problem	Keywords	Workaround
An **IP** station A on **Communication Manager** called an **IP** station B, and the call was answered on the bridge appearance of an **IP** station B which was on an **IP** station C. Another **IP** station D called an **IP** station B, and the call was answered on the bridge appearance of an **IP** station B which was on an **IP** station C. When the **IP** station C answered the second call, the first call was automatically put on hold. If the **IP** station B then tried to unhold the first call, the *talkpath* on the second call between the **IP** station D and **IP** station C would be lost. If the **Media Encryption** field on the **ip-codec-set** form was set to **none**, then this problem did not occur. Also, if the **Direct IP-IP Audio Connections** field was set to **n**, on the **IP station** form, the **ip-network-region** form or the **system-parameters features** form, then this problem was not seen.	082811	
Communication Manger responded to hold re-INVITE from third party Voice Mail server with 403 Forbidden (Service Link).	082836	
The transfer to **Voice Mail** feature access code did not work for all scenarios if the voice mail system was trunk (**PRI**, **H323**, **SIP**) integrated and **Communication Manager** was translated to disallow trunk to trunk transfers.	082843	
In an **NFAS** arrangement with backup D-channels, with the D-channels on two different H.248 media gateways (MG), the D-channels could get into a state where they never came into service after both gateways took two link bounces that continued longer than the link loss delay timer (**LLDT**). This happened when the media gateways link bounced and then re-registered, first in one order, then link bounced and re-registered in the opposite order. For example, MG 1 and MG 2 with **ISDN PRI** D-channels in an NFAS arrangement, both MGs link bounced longer than the **LLDT**, MG 1 registered back to a server, then MG 2 registered back to the same server 20 seconds later. Later, both MGs link bounced again longer than the **LLDT**, MG 2 registered back to the server, then MG 1 registered back to the server 20 seconds later. **ISDN** D-channels were both out of service.	082846	
96xx **IP** station could not blind transfer the call to station which had call-forward activated.	082859	
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Problem	Keywords	Workaround
The `refresh ip-route` command would delete the route to the default gateway causing **IP** phones to unregister and making the **TN799 CLAN Circuit Pack** useless.	082890	Busyout and then release the **TN799 CLAN**.
The **path replacement** feature could stop working in **Communication Manager** due to poor error recovery handling.	082909 082914	
Under certain unusual circumstances, a software restart could occur.	082920	
When the **Avaya One-X Communicator** was used in shared control mode, the incoming call display was not showing up. This problem occurred always when **Avaya One-X Communicator** was used in shared control mode.	082921	
In the case of blind transfer between **SIP** endpoints across different port networks for **VDN** with call shuffling enabled, audio clipping was heard.	082934	
Station-A called Station-B and conferenced Station-C on the other **Communication Manager** (Connected with **SIP**/**ISDN** trunk). Now all three stations showed display as **Conference-2**. When Station-B held/unheld the call Station-A showed Station-C's name and number instead of **Conference-2** on the display. This problem occured only for Japan Terminal 24 stations.	082982	
SIP station A called **SIP** station B and the call was shuffled. A subsequent **DTMF** REINVITE from station B was not interpreted by **Communication Manager** and caused the call to drop.	082993	
Communication Manager 5.0 onwards, when an Analog phone user on an active call put the call on hold and initiated a flash-hook transfer but did not complete it, the held call alerted again on the Analog phone even if **Misoperation Alerting** was turned off.	082994	
A system with H.248 media gateways and ephemeral caching enabled frequently went through resets unnecessarily.	083005	Turn OFF ephemeral caching or disable the maintenance internal data audit.
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Problem	Keywords	Workaround
An internal software error would cause system interchange.	083036	
An internal software error caused reset system 2 or 4.	083082	
SAFE (Self Administration for EC500) did not work for an off-pbx station whose cellular extension validation failed to find a route if all-location **ARS** table had no entries.	083101	Populate the **ARS** all-locations table with location number.
When an AAS agent's skill were updated using a **FAC** (feature access code), **monitor bcms skill** did not show the update until after a busy/release of the station. Steps to reproduce the problem: 1. Administered an AAS agent with AAS skills on the system. 2. Removed one of the skills through a **FAC**. 3. On the **SAT**, entered `monitor bcms skill <num>`. 4. The display showed the old status (the skill was logged in forthe agent, though). 5. Adding a skill via a **FAC** resulted in the same behavior.	083142	**Busy station** and **release station** to update the status. OR make the skill changes using **CMS**.
Sometimes, **AVAYA IP** Endpoints failed to register after registration of Non-**AVAYA** Endpoints.	083144	
QSIG Path Replacement failed if the **QSIG** call was set up using **EC500**. A1, A2, and A3 were on CM-A. B was on CM-B. A2 had **EC500** to B with both incoming and outgoing orig mapping. A1 called A2. Call rang at A2 and B. B answered, and did a transfer of the call to A3. (This looked like a call from A2 to A3 because of orig mapping.) A3 answered. The path was not replaced.	083173	The **QSIG**-trunk loop is now released.
A **SIP** trunk using TLS encryption could lead to a system restart.	083177	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***43 of 46***

Problem	Keywords	Workaround
ASAI adjunct route coupled with Look Ahead Interflow (**LAI**) caused tracking of the call by **CMS** to abort. The customer scenario that caused the problem was as follows: 1. An incoming call to VDN-1/vector-1 interflowed to **VDN**-2/ vector-2 which did an adjunct route to give control to an ASAI adjunct. 2. The ASAI adjunct directed the call to an agent. 3. The agent started a conference, putting the call on hold and called **VDN**-3/vector-3. 4. This interflowed to **VDN**-4/vector-4 and again did an adjunct route giving control to the **ASAI** adjunct. 5. The ASAI adjunct sent back a route request, sending the call out on a trunk to the PSTN. 6. While **Communication Manager** was waiting for feedback from the **PSTN** on the outgoing call, the agent completed the conference call, joining the incoming call and the outgoing call together. 7. **Communication Manager** then received the ALERT or PROGRESS message back from the **PSTN**, causing **Communication Manager** to send an unexpected message to **CMS**. This resulted in the calls being ignored by **CMS**.	083202	
Agents using **CTI** applications failed to transfer calls because of lost **DTMF** digits.	083214	
An attendant transfer call from a **Communication Manager** with Network Call Redirection (**NCR**) enabled could not be routed to another **Communication Manager** system over the Verizon **IPTF** service (**SIP** trunk).	083222	
In the rare case that memory was corrupted upon an upgrade, which lead to an eventual segmentation fault that would disrupt service.	083227	
The `list vdn` command shows an extra line when the **Evnt Noti Adj** column contained a non-blank value. This change removed the extra line. This was purely an aesthetic issue and did not affect any translations or call processing behavior. The only impact was seeing this extra line on non-OSSI type **SAT**. **OSSI** type **SAT** terminals like **Avaya Site Administration** were not affected.	083280	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***44 of 46***

Problem	Keywords	Workaround
When a call was transfered to an endpoint that went to a coverage, which was a **VDN**, the transfer was denied.	083297	Change the coverage path to avoid using a **VDN**.
ISDN Network Call redirection (**NCR**) feature was activated and call was placed to **SIP** station. When **SIP** station transfered the call to the vector that performed **NCR**, then call was not getting transferred.	083306	
Server resets could occur as a result of **Communication Manager SIP** trunk traffic.	083322	
Excessive TN799 (**CLAN**) socket creation and destruction caused a system resource exhaution that resulted in a system warm restart. Required a socket creation and destruction rate in excess of 50 sockets per second.	083367	
On a **Communication Manager** with **LSPs**, sometimes the **LSP's** KeepAlive Registration Request (**KARRQ**) caused a restart on **Communication Manager**.	083369	
On a **Communication Manager** with **IP** trunks, sometimes the **IP** trunk call caused a restart on **Communication Manager** because of an internal software error.	083370	
If call was placed to a sip-adjunct hunt group and the far-end domain on the signaling group was empty, then the request URI of the outgoing INVITE was also having empty domain and as a result call was failing.	083558	
Various services that are dependent upon the **libarb.so** shared library were supposed to be stopped and started during the installation of a new Service Pack if any changes to **libarb.so** was being made. Otherwise the change of the shared library out from under them could have unintended results. Servers(s) impacted: All Linux servers	083763	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***45 of 46***

Problem	Keywords	Workaround
On the **Communication Manager** servers using a non XL memory configuration, calls to hunt groups using time-of-day coverage with a hunt group number above a server specific threshold caused a software segmentation fault, which could eventually lead to a system reset. Systems with these attributes and hunt groups numbered above the following server specific thresholds could experience this problem: **S8300**: 68 **S8400**: 16 **S8500** / **S8510** / **S8700** / **S8710**: 128 **S8720** (Standard memory configuration only): 128	083871 083874	Re-administer hunt groups so that any hunt groups with time-of-day coverage paths have a hunt group number that is at or below the thresholds defined in the problem description section.
If a H.323 **IP** station was connected to a H.323 **IP** trunk, and the **IP** trunk was configured for **DTMF** transmission with either in-band mode or rtp-payload mode, and the **IP** station and trunk were in a direct-IP connection, then digits entered at the **IP** phone would not be sent across the **IP** trunk. No digits would be detected by a connected Interactive Voice Response (**IVR**) device, voice mail system, and so on.	083341	
If an **IP** signalling group were administered with **DTMF over IP: in-band-g711**, then the far end of the trunk would not hear any digits pressed on a local **IP** station's keypad. Digits stored behind the user's administrable buttons (For example, autodial) also would not traverse the trunk. **Note:** For non-IP stations such as analog and **DCP** endpoints, such digits *would* traverse the trunk.	083839	
Whenever a station initiates Crisis Alert call, **Communication Manager** should notify attendant consoles only in same tenant partition, but it was notifying all attendant consoles regardless of the tenant partitions.	083915	
Calls that tandemed through a **Communication Manager** switch, both arriving at the switch and leaving the switch over **SIP** trunks, failed more than fifty percent of the time. The call would ring once at the station on the terminating switch and then the call would drop.	083926	
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Table 3: Fixes delivered to Communication Manager 5.1.2 SP#1 ***46 of 46***

Problem	Keywords	Workaround
If an **IP**/**SIP** signaling group was admininistered with the following attributes: ● Direct IP-IP Audio Connections? y ● DTMF over IP: rtp-payload then under certain circumstances, when a user would press digits on the phone, those digits would not be heard on the far end of the corresponding trunk. In particular, if an **IP** station, which can shuffle, called over the trunk, no digits pressed would be heard on the far end of the trunk.	090020 083386	
Calls that involved H.248 media gateways and executed transfers via particular **SIP** messages, failed due to a delay in connecting media resources.	090082	
After upgrading from **Communication Manager** 4.0 or 5.0 to **Communication Manager** 5.1.x, the web access mask profiles were not displayed correctly on the **Web Access Mask** profile page. There were several check marks missing.	090296	
Under certain circumstances, if two switches were connected using **DCS** trunking, and a call covered from the first switch to an **X-port** (station administered without hardware) on the second switch, the second switch could experience a restart. The first switch would remain unaffected.	090327 090309	
When shuffling was enabled and Music on Hold (**MOH**) was disabled, a conference between three **IP** endpoints resulted in one of the conference participants having no *talkpath*.	090474 083744	
When calls were made involving integrated music sources over a trunk, the trunk would sometimes lock up after the call was completed, even though it appeared to be idle. If enough trunks ended up in this locked up state users were blocked from making additional trunk calls, received a denial event, and were connected to busy tone.	090778	
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Problems fixed in Communication Manager 5.1.2 SP#2

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

Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 *1 of 16*

Problem	Keywords	Workaround
If the "Mode Code Interface" was enabled on the "system-parameters features" form, then calls routing over the **QSIG MWI** (Message Waiting Indication) hunt groups to voice mails intermittently received the generic greeting. The problem was not seen if the "Mode Code Interface" was disabled on the "system-parameters features" form.	074284	
30% of R2MFC trunk calls tandemed to an **ISDN** trunk fail to complete.	080633	
S8730 server interchanged, contrary to requirement, when one of two power supplies failed.	080696	
If an incoming R2MFC trunk call to an **IP** station was forwarded over an **ISDN** trunk, the call failed sometimes.	080745	
An Agent on an H.323 station on a port network that received an automatic call distribution (**ACD**) call had no *talkpath* when the call came from a media gateway and the **VDN** of origin Announcement (**VOA**) of length 0 came from a port network. If the agent then attempted to put the call on hold, the call dropped.	080841	
An incoming **ISDN**/**IP** trunk call to **Communication Manager** A terminated to **VDN**-A1/vector-A1 that routed the call to an **IVR**. The **IVR** answered the call and then started a transfer of the call to **VDN**-A2/vector-A2 on **Communication Manager** B. **VDN**-A2/vector-A2 did a **BSR** poll across **ISDN**-**PRI**/H.323 trunks to **Communation Manager** C, **VDN**-C1/vector-C1. Station-A1 completed the transfer while the **BSR** (Best Service Routing) poll call was still in process to **Communication Manager** C. **VDN**-A2/vector-A2 then queued the call locally to skill-A1 and delivered the call to an agent on station A2. Station A2 saw the **VDN** information on it's display which was quickly overwritten with "UNKNOWN NAME".	080863	
Incoming R2MFC trunks calls to stations on the G350 gateway did not complete.	080888	
No ringback was played on incoming R2MFC trunk calls if the principal was busy, and the call was sent to coverage.	080967	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***2 of 16***

Problem	**Keywords**	**Workaround**
Calls forwarded to a sip-adjunct voice mail went into an endless loop.	081262	
The System Log web page displayed system log information based on the check boxes selected for log types and log views (for example, bash and cron) along with an optional entry for a pattern match. With certain software releases and/or service packs, no output was displayed when multiple views were selected or when a match pattern was entered.	081656	
Customers monitoring stations with **ASAI** may see incomplete called party numbers when calls are manually placed from the monitored station over an **ISDN**-**PRI** trunk that was administered with overlap digit sending.	081804	
Principle station (initiator) will not hear caller (via **PSTN ISDN** trunk) after ready indication tone was applied.	081886	
When bandwidth limitations were met between two network regions, subsequent calls between the regions failed instead of routing using alternate route pattern preferences.	081928	
ASAI Orig call dropped intermittently when a call was covered on a **DCS** coverage trunk.	081938	
Enhanced Call Forwarding might not always be executed correct if a user has an entry in the "Off-PBX" station table (for example, Extension To Cellular). In this case if a call arrived at the forwarding station it was not always forwarded but sometimes stayed ringing at the forwarding station. Servers impacted: All Linux Media Gateways impacted: Not Specific	082184	
If station B has a team button, which pointed to station A, then when station C tried to call station A that was unregistered, station C would get ringing tone if the "Don't Answer Criteria For Logged Off **IP**/**PSA**/**TTI** Stations?" is set to "y" on "system-parameters features" form. This is expected behavior. The problem was after station A registered, the call would be automatically answered even if station A did not try to answer the call.	082189	
When an **IP** softphone made a trunk call over an H.323/**PRI** trunk and did not receive an ALERTing message, its display may not be updated with the dialed digits.	082432	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***3 of 16***

Problem	Keywords	Workaround
When there were more than 15 **CTI** links the '`status aesvcs cti-link`' command displays garbage on the second page when another status command is run at the same time.	082827	
For users that access the directory feature after going offhook or after conference or transfer button pushes would timeout from dialing before they could complete the directory lookup.	082855	
An incoming Russian toll trunk call was answered at a principal having bridged appearances. Under certain conditions, the bridged appearance call remained active after the trunk call was dropped.	082949	
In recovery scenarios from control network outages of 45 seconds or more with high traffic volume, a system WARM restart of Communication Manager could occur.	083007	
When the Digital Loss Group field on the trunk-group form contains an inappropriate setting (for example, a digital station loss group is specified for a digital trunk group) then features like Inter-Gateway Alternate Routing may 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	
Under particular circumstances when the **SBS** (Separation of Bearer and signaling) feature is enabled, an incoming call to a **VDN** over a trunk that is in night service mode is routed to the **VDN** and the agent receiving the call can't answer it.	083047	
When a given region has multiple interconnections (for example, region 1 to 2, region 1 to 3, and region 1 to 4), and that region has failures with more than one of its interconnections, the "`test failed-ip-network-region`" command does not work correctly. The test will only do the first failed region pair and not successive region pairs correctly. The command shows garbage for the rest of the region pairs that it tested.	083126	
CDR reported long duration calls for many incoming and outgoing trunk calls in the system. Also during a busy hour or busy day all analog loop-start trunks were busy, even though many ports showed no connected ports. This could be remedied by manually busying out and releasing the trunk ports to get them back in service.	083372	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***4 of 16***

Problem	Keywords	Workaround
When voice mail was accessed via an autodial button which has address digits for the voice mail adjuncts followed by a pause and a pound the user was prompted with a welcome message instead of directly asking for password.	083381	
An **ASAI** generated outbound call with answer supervision via a call classifier and a **SIP** out bound trunk may have resulted in the called party hearing ringback when they answer if they are using a **SIP** phone.	083524	
For restricted call over trunk From: header for INVITE message should contain anonymous@anonymous.invalid.	083564	
The **Communication Manager** watchdog process will no longer reset if it detects more then 4090 number of processes. The limit is now 32750.	083605	
When a call was made across a **SIP**/**ISDN** trunk to a **VDN** that routed to a hunt group and an agent answered, the caller display showed the hunt group name and number, even though the **ISDN**/**SIP** Caller Display field was set to blank on the hunt group form.	083632	
When a conference call was transferred to a station of type 96xx and the station answered the call, the phone displayed the Ringing icon on the call appearance instead of displaying the Conference icon.	083638	
If an encrypted, direct-ip (shuffable), H.323 trunk was administered with the following characteristics on the **QSIG** TRUNK GROUP OPTIONS page: Path Replacement? y Path Replacement with Retention? n Path Replacement Method: always then under certain circumstances a call across that trunk could appear garbled or disappear altogether after a few seconds.	083648	
When call was made to a Group page extension with a station Administered with out Hardware as its member, then a delay of at least 7 seconds was observed in getting confirmation tone.	083691	
Extension display format of an extension was not updated as per the last known network region of the same extension on an IP phone when that extension was logged off or was out of service The work-around is to dial "Refresh Terminal Parameters Access Code" on the IP phone as administered on "FEATURE ACCESS CODE" Form.	083699	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 *5 of 16*

Problem	**Keywords**	**Workaround**
This change fixes a problem where Watchdog was recording multiple alarms on mdmtty starting and stopping.	083701	
"`List measurements blockage PN Last-Hour`" command showed high usage for some port networks that were not experiencing a lot of traffic. Also agents were getting one-way talk path and sometimes zip tone was not heard.	083720	
After a migration the init user was no longer allowed to access Systems Administration Terminal.	083762	
The following scenario occasionally caused a system reset. With the long hold recall feature enabled station A called station B. Station B answered. Station A conferenced station C. Station C put the conference on hold. Station A put the conference on hold. The long hold recall feature began alerting station A. Station C unheld the conference. At this point the system incurred a segmentation fault.	083836	
Under certain circumstances, there was no *talkpath* or dial tone on some IP station calls if the previous call involved hold or conference.	083870	
PAM password complexity requirements were not being enforced when changing a password via the System Management Interface (**SMI**) GUI.	083882	
External incoming calls terming onto the logical agent due to call-fwd or call-coverage over a **QSIG** trunk were not following the coverage path administered on the agent's form.	083889	
Call to a logged-off IP station which had **SAC** (Send All Calls) activated and "Maintain SBA At Principal" field set to y was dropping after sometime when answered at the coverage point.	083910	Set "Maintain SBA At Principal" field to 'n'.
`List usage extension` command display incorrect data when any station is registered using the softphone with telecommuter mode. This problem goes away by unregistering all stations who are registered using soft phone with telecommuter mode.	083943	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***6 of 16***

Problem	Keywords	Workaround
'`display button-labels n`' command displays labels in the supported unicode language when phone is registered. To display the labels in the supported language Communication Manager get language data from the end-point. When the phone is unregistered there will no end-point. If there is no end-point then **Communication Manager** will not get language data from the end-point. Then Communication Manager displays in the default English language. Now we are adding the following note on the display button-labels to know the user that 'display button-labels n' will default to English when end-point is not registered. 'Note: Unicode labels will default to English if the endpoint is not in service or does not support the language specified'.	090007	
When an audio group contained announcement boards that were all in network regions without an ip-codec-set defined to the network region of a requesting port, yet connectivity did exist via media processors in other NRs defined within the same Port Network as the requestor or announcement board, the system did not select any member from the audio group and the call failed.	090015	
No alarms were sent for invalid login attempts.	090035	
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	
While the transferee was ringing the other party in the call showed "ANSWERED BY" and transferee's number. This problem was visible when the transferee station was not an off-PBX telephone integration & mobility (OPTIM) station.	090037	
A call may get dropped when shuffling a call using a **SIP** (session initiation protocol) phone.	090040	
IGAR calls failed nearly every time. **IGAR** admin was normal; what was unusual was the public **ISDN**, which was sending PROGRESS instead of ALERTING most of the time for the **IGAR** trunk calls.	090042	
When an agent is automatically logged out of a skill group due to the Forced Agent Logout by Clock Time feature, no logged out event is sent to the monitoring device.	090105	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 *7 of 16*

Problem	Keywords	Workaround
A system reset might occur when the Enhanced Call Forwarding feature is administered or the translations are saved (automatically or on administration request). Server(s) impacted: All Linux based servers Media Gateway(s) impacted: Not Specific	090108	
A special dial tone was not played if a domain control was active on the station.	090122	
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	
A burst of static noise was heard while placing a call over an Alternate Voice/Data (**AVD**) trunk. The voice bearer capability was set to zero for both the trunks, on the first page of trunk group form.	090128	
9620 set type could not cancel the call if there was a Single Step Conference party active on the held call.	090131	
Turret stations adminstered as OPTIM endpoints could not use their call-pickup feature to retrieve a held call, due to use of SIP messaging that was not previously required or supported for OPTIM endpoints.	090151	
With a SONUS sip gateway, calls using **SRTP** and shuffling may get dropped.	090156	
If the announcement queue became corrupted, CPU overload could occur, resulting in a system restart.	090157	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***8 of 16***

Problem	Keywords	Workaround
In an **NFAS** arrangement with backup D-channels, with the D-channels on 2 different H.248 media gateways (**MG**), the D-channels can get into a state where they will never come into service after both gateways have taken 2 link bounces that continue longer than the link loss delay timer (**LLDT**). This would happen if the media gateways link bounced and then re-registered, first in one order, then link bounced and re-registered in the opposite order. For example, **MG** 1 and **MG** 2 with **ISDN PRI** D-channels in an **NFAS** arrangement, both MGs link bounce longer than the **LLDT**, **MG** 1 registers back to a server, then **MG** 2 registers back to the same server 20 seconds later. Later, both MGs link bounce again longer than the **LLDT**, **MG** 2 registers back to the server, then **MG** 1 registers back to the server 20 seconds later. **ISDN** D-channels are both out of service.	090166	
There are apparently a number of **ISDN**s (or perhaps data networks with Q.931 interfaces at the edge) where the talk path is cut through before the CONNECT signal arrives at the calling side. In this **DPT** case, the initial **IGAR DTMF** digits arrived from the called side before CONNECT, and the talk path from calling to called side) was sometimes not open for a second or two. This caused a few leading **DPT DTMF** digits to be lost about half the time. Those digits identified the calling station, so the display on the called station showed an incorrect station name or UNKNOWN NAME.	090178	
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	
The Avaya IQ reporting adjunct cannot track a transferred call in which the first call leg is unmeasured, the second leg is a measured **ACD** call, and the transfer is completed while the **ACD** call is ringing an agent.	090214	
If an AUDIX node name was used on the station form, the **`list usage node-name`** command would not list that station in its output. This made it virtually impossible to change or remove that node name using the **`change node-name`** command.	090216	
A trunk call ringing at a **CMS** measured agent is dropped by Wait Answer Supervision Timeout (**WAST**). Before the associated trunk drops, an agent answers the **ACD** call. **CMS** identifies this as "two calls connected" and ignores the call.	090217	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***9 of 16***

Problem	Keywords	Workaround
If an agent with IQ/**CMS** measured and unmeasured skills answers a call in an unmeasured skill and then transfers that call, **Communication Manager** does not report the Agent-LoginID of the controlling party to IQ/**CMS**.	090248	
Incoming calls using **SIP** trunks may have resulted in no talk path when answered if codec negotiation failed.	090254	
As per RFC3262,PRACK shall only be rejected with 481 transaction/call leg not found, and in all other scenarios 200OK should be sent for PRACK. **Communication Manager** was processing PRACK and in case of successful processing was sending 200OK.This behaviour was changed and for all scenarios 200OK shall be sent for PRACK irrespective of processing of PRACK except when call leg/transaction is not found	090260	
Tandem calls with an incoming R2MFC trunk failed if the "Incoming Tone (**DTMF**) ANI:" field was configured as ANI*DNIS* or *ANI*DNIS* on page 3 of the trunk group form for incoming R2MFC trunks. The problem was only seen when the maximum digits in the **ARS** analysis form was greater than the number of digits dialed.	090272	
Communication Manager reset observed while using multiple service observers feature.	090273	
When there was a mix of calls on the same **SIP** trunk group between two **Communication Managers** where some calls had User-to-User headers with **UCID**s and some calls had no User-to-User header then the wrong **CID** may have been displayed on the phone.	090279	
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	
When using GEDI interface provided in Avaya Site Administration, the ROOM STATES fields on page 3 of the system-parameters hospitality form could not be changed.	090304	
In a mixed dialplan environment having extensions with different lengths starting with the same digit, the Message Waiting Indicator (**MWI**) did not light at the shorter length extensions.	090310	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***10 of 16***

Problem	Keywords	Workaround
The `list ip-interface` command could have returned an **EECCR** due to translation corruption caused by having a trunk group 2000 administered in the system. The only workaround is to not have trunk group 2000 administered in the system. However, once the **EECCR** occurred, removing TG 2000, would not correct the problem. Avaya support would need to fix it.	090337	
Filesyncs of the Web Access mask from a Main to an **LSP** or **ESS** (specifically S8300Bs, S8500Bs and S8500Cs) are now supported.	090338	
In case, multiple calls came on an Administration Without Hardware (**AWOH**) station, having Bridged Appearances (**BA**) on multiple stations, and one of the calls was answered, the display on the other stations having the **BA** of the same station, was blank instead of showing the information of next oldest ringing call. This problem was specific to "Avaya Digital Terminal for Japan" (J24) sets. This problem would not visible if "Bridged Idle Line Preference" field on the station form is set to "n".	090348	
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	
With 2410 and 2420 endpoints running firmware version 5, for calls made using speed dial buttons, some of the **DTMF** digits are dropped.	090354	
After a Port Network COLD reset, control network outages could lead to a WARM restart of the port network and boards in that port network not being inserted. The boards would stay out of service until an audit in Periodic Maintenance runs and inserts them causing loss of service. The problem can be fixed by executing a COLD reset of the port network.	090390	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***11 of 16***

Problem	**Keywords**	**Workaround**
Call to a vector, originated from non-authorative domain doesn't go to coverage. Specific call scenario is: 1) Incoming call to **VDN** from direct **SIP** trunk between Service Provider (SP) and **Communication Manager** (CM). 2) CM routes the call back to SP because Domain name in From header of the INVITE does not match Authoritative Domain field in ip-network-region form of the **SIP** trunk. Notes, it is normal for From header to have different domain from the one in ip-network-region since the call can be forwarded by SP from other **SIP** domain. 3) SP sends the call back to CM and CM send "302 Loop detected by CM" respond, which SP forward it back to CM. 4) CM got "Loop detected by CM" for the call it routed to SP on step 2 so it know the call now need to be process locally and route the call to the **VDN** 5) Call hit a route-to vetor step with coverage enable. 6) Route-to step route the call to a station that have coverage path set 7) station does not answer the call and the call never goes to coverage so station stays alerting.	090391	
Customers recording calls using NICE and **AES** integration might not have calls recorded. The problem appears when NICE is rebooted and can affect different stations each time.	090431	
Failover from the main server to an Enterprise Survivable Server (**ESS**) or vice-versa would cause a flood of "keep-alive" messages from the IP phones to go to the newly active server. This would cause the newly active server to experience high occupancy. Thereby, delaying the recovery of the system.	090439	
Entering "**`list ip-tti-stations xxxx`**" (where x is a numeric value of length 1 or more) at the **SAT** (System Access Terminal) caused the command to fail. If the value was less than 3 digits, the **SAT** would output pages of useless data. If the numeric value was 3 or more digits, the system would lock up, requiring an interchange or power cycle to recover.	090443	
Using the System Access Terminal (**SAT**), customers could not add a loudspeaker paging zone on the 'Loudspeaker Paging' form. They would see the following message upon form submission: "Error encountered, can't complete request; check errors before retrying"	090470	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***12 of 16***

Problem	Keywords	Workaround
Calls did not go to the EC500 when the Media Gateway to which the desk phone was connected was unregistered.	090472	
If an agent is logged in with two skills and a supervisor tries to remove the first skill using a feature access code (**FAC**), the skill removal does not work. "monitor bcms skill 1" still shows skill 1 as "AVAIL" for the agent and "status station" still lists this skill as "AI" for agent.	090483	
The voicemail system reported the wrong calling party number in some "transfer into voicemail" scenarios.	090507	
A message about a non-service-affecting issue was printing in **Communication Manager** logs too often and consuming log space.	090508	
A Call tore down if Service Observed station put call on hold and Music On Hold played on the call.	090509	
With the **QSIG** customer option 'Value-Added (**VALU**)' feature enabled the ability for the called party to join into a call that has been answered by a member of the called party's coverage path was inhibited. This issue did not occur if "QSIG Value-Added" field is disabled on the trunk form.	090511	
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.
As per RFC 3261, port in **URI** is disallowed for From/To header.	090580	
When executing the SAT command "`list skill-status`", the value for the "Service Level" field does not change and always has the same value as the Hunt-Group value for Service Level Target Percentage.	090623	
When **Communcation Manager** experienced unusually high SIP signaling traffic, causing internal buffer congestion, problems occurred with **Communcation Manager**'s management of signaling connections, leading to dropped calls.	090639	
Display of the system logs using the System Log web page or the `logc` command would sometimes indicate that there were no data to display but direct access to the log by the use of an editor or bash command would indicate that there were current entries.	090648	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***13 of 16***

Problem	Keywords	Workaround
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	"Auto-Answer" mode of the station from which EAS agent login be configured to either "acd" or "all".
When administering the extended-user-profile form, the 'Allow Only' field would not allow 2 separate ranges to be entered together, for example, 4001-4009,4015-4020. The error message "Second value of a range must be greater than the first value of the range" was given when the form was submitted with 2 ranges. A work around for this condition is to insert a single instance between the 2 ranges. For example, 4001-4009,4015-4020 could be entered as 4001-4008,4009,4015-4020.	090669	
In S8510 servers, status of redundant power supplies was not displayed correctly.	090680	
A call that resulted in Network Call Redirection (**NCR**) over a measured **SIP** trunk with the **SIP** REFER METHOD was not measured by the switch originating the call. It should be measured as an outbound call on that trunk.	090682	
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** responded with 200 OK, a=inactive, and null IP address/port and did not begin a transition to fax until receiving a re-INVITE with a=sendrcv.	090687	For the far-end not to 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 continued dialing remaining digits, the displays showed digits interspersed.	090740	
A station was calling a monitored station, whereas the talk path used an **IGAR** connection. On the monitoring station the alerting state of the monitored station was not indicated. When the monitored station went off hook, the active state was indicated on the team button on the monitoring station.	090751	
When a call with a service observer encounters a party that was not allowed to be observed, and the answering party was auto answer, the answering party did not hear zip tone.	090758	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***14 of 16***

Problem	Keywords	Workaround
When **IP** phone on **Communication Manager** called **DECT** phone via **QSIG** trunk and **DECT** phone happened to be switched-off, call 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	
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	
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	
After making multiple calls and holding/unholding them, once in a while `403 Forbidden` message would be sent while trying to put the call on hold and the "Music on Hold (**MOH**)" was not being played, while the call was on hold. This problem was specific to **SIP** stations and did not occur consistently.	090824	
Under certain circumstances involving H.323 stations which cannot shuffle, H.248 Media Gateways, interconnected network regions, bridging, and the Hold feature, two users from two different calls could hear each other. For example: Network regions Y (with station '2' administered) and X (with station '1' administered) were interconnected. Station 2 had a bridged appearance for station 1. An incoming trunk call was answered by Station 1 via a Media Gateway administered in region Y. Similarly, an incoming trunk call was answered by Station 2 via a gateway adminstered in region X. Station 1 pressed HOLD, then pressed the call appearance of Station 2. The first trunk call could then hear the second trunk call.	090830	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***15 of 16***

Problem	Keywords	Workaround
Under certain circumstances, an incoming call to H.323 **IP** Agent which can shuffle, could result in no *talkpath*. Specific scenarios as follows: 1. Incoming call on an H.248 controlled media gateway in network region X. 2. Call terminated to a Vector Directory Number with a vector that included a music step and a "queue to skill" step. The music played from network region Y. 3. An IP agent which can shuffle, in network region X was in "Aux mode" when the call arrived. 4. When the agent became available, the caller and agent would not hear each other.	090838	
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 would be specific to "Avaya Digital Terminal for Japan" (J24) sets.	090857	
CallMaster V or 64xx stations did not clear the display when it was on a call with headset and transferred the call.	090883	
Called **SIP** station showed "UNKNOWN NAME".	090885	
Customer having Application Enablement (**AE**) server running software version 4.2 (or 4.2.1) and is using Device Media and Call Control (**DMCC**) functionality of the **AE** server then `getCallInfoRequest` would fail if the associated extension was administered as **CTI** type (also known as phantom user) on **Communication Manager**.	090894	Use regular **IP** extensions instead of using **CTI** (or phantom) extensions.
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	
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Table 4: Fixes delivered to Communication Manager 5.1.2 SP#2 ***16 of 16***

Problem	Keywords	Workaround
When calls made to Vector Directory Number (**VDN**), which was routed to an Administration Without Hardware (**AWOH**) station, was 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" was enabled on the stations in the coverage answer group. This problem would be specific for "Avaya Digital Terminal for Japan" (J24) sets.	091029	
Some **SIP** phones may not hear **DTMF** tones from the other side of the call when shuffling was enabled.	091034	
Incoming call over **SIP** trunk with non-authoritative domain will not be rejected.	091049	
With the Time-To-Service (**TTS**) feature the socket failure retry mechanism failed. This resulted in dial tone failure.	091050	
Intra-switch Call Detail Record was not produced for calls made to **SIP** stations.	091063	
Under certain circumstances, activating a feature which allowed someone to talk to a particular party in a call (for example, "whisper page") on an H.248-controlled media gateway could cause internal **Communication Manager** memory corruption, potentially leading to system instability.	091453	
Issues associated with the following keywords were also fixed in **Communication Manager** 5.1.2 SP #2: 091516, 083476, 083266		
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Problems fixed in Communication Manager 5.1.2 SP#3

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

Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *1 of 14*

Problem	Keywords	Workaround
Updates for secondary dial tone were not updated to Media Gateway for the first time the administration changes were made on "tone-generation" form.	073125	Submit the changes on "tone-generation " form once again.
If a call was delivered to an agent using H.248 Media Gateway resources, and the call then received an unusually rapid answer via third-party call control from an adjunct software application, the call did not have *talkpath* between the agent and the calling party.	073291	
When **Communication Manager** sent Re-Invite for Display Change or session refresh to the other end point and if the end point responded 200 OK with a different port than the previous one, the existing call should not be dropped.	073783	
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	
The display on the calling user showed ANSWERED BY when in fact the user that was called over the **SIP** trunk was busy and the calling user was hearing busy tone. This issue came when call was made over a **SIP** trunk and the called party was busy.	080138	Set "ANSWERED BY" field on trunk group form to "no".
Calls set up using **Communication Manager**'s codec list in ip-codec-set form instead of setting up calls with the **SIP** endpoints preferred codec list.	080213	
DTMF tones may not be heard when **Communication Manager** was connected to certain non-Avaya equipment when shuffling was involved.	080890	
When a **Communication Manager** user dialed an extension on non-Avaya system using an H.323 trunk or used an abbreviated dialing string containing ~w and a remote access extension and authorization codes in it, then sometimes the call failed.	081214 091031	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***2 of 14***

Problem	**Keywords**	**Workaround**
Station A has Enhance Call Forward feature button activated. This button could be deactivated by pressing the feature button followed by button 2 (for deactivation) and button 0 (for all call forwarding). But if the feature was deactivated by pressing the keys/buttons rapidly, then the feature got deactivated but the lamp of the feature button remained steady. The problem was seen on the telephones which had the lamps associated with feature buttons like the telephone models 96xx, 6408D+, 8410D, etc. The problem was seen only when the buttons were pressed rapidly without keeping time interval of even a second.The problem could be obviated by repeating the process (press the feature button followed by 2 and 0) or Pressing the button slowly keeping around one second time gap in each button press.	081259	
Previously, A PKTINT fatal fault was a major alarm. Now, A PKTINT fatal fault is a WARNING alarm for the first 3 minutes, and a MINOR alarm after three minutes.	081703	
On system features page *Don't Answer Criteria For Logged Off IP/PSA/TTI Stations? y* This feature caused ringback to be played to the caller when the destination station was not in service. However, if EC500 were administered for the station and disabled, then the caller would hear busy.	081741	
All long distance fax calls leaving **Communication Manager** on a **ISDN-PRI** trunk failed 100% of the time. Fax calls to local area code succeeded 100%.	081779	
The **`change mst`** command was allowing the submission of the form with the Start Trigger value as Y without accepting a value for the Message Type.	081948	
An **IP** trunk call between two 'Avaya **Communication Manager** switches' failed half of the time when both trunk ends had both, **SRTP** and no encryption settings enabled, in the far-end region's codec set and one switch did not have **SRTP** capable resources.	081956	
In some cases, attempting to add an EC500 XMOBILE station with Data Module set to "n" would result in the error: "`Error Encountered, can't complete request`".	082191	
When the Customer attempted to add the 102nd **CLAN** in the system, the **SAT** locked up and the command could not be aborted.	082368	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***3 of 14***

Problem	Keywords	Workaround
The output of the System Logs web page was different from the output of the `logv shell` command even though the same pattern filters were used.	082575	
The system could not direct high traffic volume towards H.248 media gateways efficiently. This would cause unnecessary delays in call setup, including tones and talk-path, and could possibly have caused call failures under high call volume.	082991	Direct traffic towards Port Networks OR Add media gateways to network regions with a high call load.
Some **SIP** entities may lose *talkpath* or had their calls dropped when interacting with **Communication Manager** after calls were shuffled to a direct media connection.	083071	
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 given.	083103	
DCP terminal maintenance could get in a state where terminal lamp and button refreshes, as well as date and time updates, were never sent to the terminal. This could have a negative effect of clocks being wrong on the terminal or lamps not being refreshed that need to be.	083146	
Dial Plan Transparency (**DPT**) calls failed in the following cases: 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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *4 of 14*

Problem	Keywords	Workaround
Digital Set (**DCP**) port may not be restored to service automatically when the associated softphone was logged off. A new retry mechanism was introduced and attempted to restore the **DCP** port for a maximum of three times, at 1 second, 10 seconds and 1 minute after the the soptphone client was logged off.	083207	
The customer would observe that making a call directly to a cell phone would ring the cell phone, but if the call was extended to the cell phone using EC500 it would not ring. The customer/user must have a correctly configured and enabled EC500.The Trunk Select field of the off-pbx station-mapping form must specify ars or aar. The route pattern used for routing to the (cell) phone number must be set to allow look ahead reroute. The first available trunk capable of handling an EC500 call must be dysfunctional in a way will result in the call being rejected or failing. There must also be another routing pattern entry that can carry a call to the cell phone.	083474	
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	
On doing an **SNMP** walk on the interfaces **MIB** group, the ethernet speed was incorrectly reported.	083688	
Could not login an agent using a **CTI** application that was running and using an autodial button.	083753	
Dial Plan Transparency feature was invoked towards an unplugged **IP** phone causing improper trunk usage.	083845	
Very high volumes of **IP** phone unregistration events in a short period of time could cause the **Communication Manager** application to freeze.	083872	
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	
ISDN call setup retried as a result of glare conditions fail 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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***5 of 14***

Problem	Keywords	Workaround
Music was not played to the calling station when the Look-Ahead Interflow (**LAI**) checks were performed on the **Communication Manager**.	090140	
In cases of feature activations over **SIP** trunks where additional digits were required in addition to the **Communication Manager** feature access code (for example, call forwarding destination digits required in addition to the call forwarding feature access code), the feature activation may not be invoked properly.	090176	
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	
Pickup alert was not sent to other members of the Pickup Group if the called member was an unregistered station.	090276	
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	
If a **SIP** signaling group was in a bypass state, **Communication Manager** did not skip over this signaling group for outgoing calls as quickly as it should have. In fact, every channel of the signaling group was erroneously considered as a candidate for an outgoing call leading to increased levels of processor occupancy.	090341	
Under heavy traffic conditions, **Communication Manager** could enter system overload if one or more H.248 media gateways reported H.248 errors.	090535	
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	
Upon calling a dissociated **IP** phone the caller was not hearing ringback even if the feature "Don't Answer Criteria For Logged Off **IP/PSA/TTI** Stations" was enabled.	090624	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***6 of 14***

Problem	**Keywords**	**Workaround**
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 allowed phones with a call appearance that was in CA_WAIT_ORIG to originate a call on that appearance. This prevented that appearance from being stuck and considered busy by the software.	090655	
If a call-center agent using an H.323 endpoint or softphone was offered a call, but their endpoint wasn't 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	
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" displayed a route-to command with a collected digit of "0" or "#", the character "a" (10) or "c" (12) was displayed instead.	090750	
Busy out on H.323 trunks in certain intermediate call states ws 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	
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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *7 of 14*

Problem	**Keywords**	**Workaround**
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	
Incoming trunk calls across a **SIP** trunk could occasionally fail.	091003	
A team button was configured for a station. A call was picked using this team button. Transfer of this call to the voice mail server failed.	091009	
A call that covered to a **VDN** and routed out to an available agent could not be conferenced. Even though the call was no longer in vector processing and routed to an agent, that agent was not able to conference in the following manner: 1) Given: Incoming call has covered to a **VDN**. That **VDN**/ Vector executed a "queue-to skill" step and routed to the next available agent who is now active on this call. 2) Agent presses HOLD to hold this incoming call. 3) Agent presses new call appearance and dials another station/agent. 4) That station/agent can let the call ring or answer. 5) Agent presses CONFerence and a third call appearance goes active. 6) Agent presses the incoming HELD call appearance. 7) Agent presses CONFerence and is blocked from conferencing. Denial event "1746 Conf/xfer a Vector call" occurs.	091014	
When Call to prime is tranfered to **VDN**/HUNT, the display on transfered party was showing calling party's information.	091025	
Abbreviated dialing containing ~w failed where dialing string contained remote access extension and authorization codes in it.	091032	
Abbreviated dialing having too many digits in the dialed string used to cause **PCD** (Packet Control Driver) congestion.	091033	
System resets could occur on **Communication Managers** using the **ASAI** feature.	091035	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***8 of 14***

Problem	Keywords	Workaround
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** will slow down during Automatic Call Distribution (**ACD**) calls.	091066	
Under heavy traffic loads, communication with H.248 media gateways could become severely restricted, causing slow response times and user-interface delays of several seconds.	091097	
The Call Pickup feature has a special algorithm to determine which call is 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	
Calls between some **Communication Manager**'s via **SIP** trunk and/or with **SIP** endpoints may fail a short time after call initially connects.	091145	
Under certain circumstances for calls involving two port networks (**PN**s), or a **PN** and an H.248 media gateway, **Communication Manager** could experience a restart.	091174	
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	
Activation of Send All Calls (**SAC**) from a 96xx **SIP** phone may fail if the "Enbloc Dialing without **ARS FAC**" feature was enabled.	091178	
When a Vector Directory Number (**VDN**) was called over a trunk, or from **SIP** phones, after answer the display on the calling party showed the information of the **VDN**. It should have shown the information of the answering party on the display.	091181	
Data for the g3trunksta **MIB** group displayed garbage values when a walk was performed on the g3mib.	091186	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***9 of 14***

Problem	Keywords	Workaround
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	
When the duplication link went down, a major alarm was logged immediately without any resolution to retire the alarm.	091287	
Under certain conditions, an internal **Communication Manager** error may result in a system restart.	091292	
For calls involving multiple H.248-controlled media gateways and 'shufflable' H.323 stations assigned an 'audix-rec' button, if the AUDIX ONE-STEP RECORDING feature was administered as "Apply Ready Indication Tone To Which Parties In The Call? initiator", users may experience one-way talk path when using the 'audix-rec' feature.	091299	
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	
If a user invoked AUDIX recording or the whisper-page feature, they could experience a one-way talk path if the call involved an Inter-Gateway Alternate Routing (**IGAR**) connection between a port network and an H.248-controlled media gateway.	091363	
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 Connect 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	
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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***10 of 14***

Problem	Keywords	Workaround
Changing ping parameters on page one of the "system-parameters ip-options" form on an S8300 caused invalid TTR-LEV alarms to appear.	091427	
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	
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	
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	
While transfering a call over **SIP** trunk (session initiation protocol), call should not be dropped.	091532	
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	
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 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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 ***11 of 14***

Problem	**Keywords**	**Workaround**
When an external call was made to a Busy **IP DECT** station, caller was getting reorder tone instead of getting busy tone.	091586	
Communication Manager could experience a system restart with H.323 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 was stopped, started, reset, busied out or released.	091604	
This issue had 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 came on a **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. 3. When calls made to a **VDN**, which was routed to an **AWOH** station, was 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	
Conditions: a) An **ASAI** adjunct initiates a call to an **OOS** station on the same server but in a different Network Region. b) The call invokes Dial Plan Transparency to reach the **OOS** station. c) The **ASAI** adjunct drops the call early, before the call is 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 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *12 of 14*

Problem	Keywords	Workaround
Calls failed to conference after covering and routing from a **VDN** to a valid extension.	091668	
The voicemail adjunct reported the wrong calling party number in some "transfer to voicemail" scenarios involving X-ported stations.	091670	
IP agent calls were getting dropped in certain scenarios involving high call traffic.	091687	
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	
Denial event 2076 "**IP** RRJ-Call Mismatch endpt" sometimes included incorrect data.	091757	
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	
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	
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	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *13 of 14*

Problem	Keywords	Workaround
An Avaya **Communication Manager** user could not login on the IA770 to retrieve voice messages.	091815	
All the **IP** stations on the server were abruptly rebooting at the same time.	091818	
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	
From load 733 onwards, 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	
In case of Calling ID blocking, Cellular Service Provider originated call from mobile phone was not working	091936	
Port number shall not be sent as 0 in From header for a **SIP** tandem call scenario.	091985	
SDES **SRTP** call did not work across **SIP** trunk.	092047	Either disable **SRTP** or shuffling on **SIP** trunk calls.
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	
Whenever a call 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.	092095	
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Table 5: Fixes delivered to Communication Manager 5.1.2 SP#3 *14 of 14*

Problem	Keywords	Workaround
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	
When user initiated a warm restart, then, under certain internal conditions, the system may encounter a reset system 2.	092233	
On rare occasions, the Time Slot Record Audit (**TSRA**) may provide some faulty data as shown on the 'status audits cumulative' form.	092322	
For external inbound calls to an **IP** Agent over a media gateway, the **IP** Agent observed announcement cross talk.	092511	
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	
Issues associated with the following keywords were also corrected in **Communication Manager** 5.1.2 SP3: 091091, 091273, 091289, 091551, 091577, 091701, 091756		
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Known problems

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

Table 6: Known problems in Communication Manager 5.1.2 SP#3

Problem	Keywords	Workaround
When a unicode-customized button label is created with the backup/restore file for an **IP phone**, some characters may be truncated, and the label sent back to the phone is not the label that was edited into the backup/restore file. **Communication Manager** currently allows up to 26 bytes of data for storage of a unicode-customized button label.	081169	

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
AE	Application Enablement
AES	Application Enablement Services
AGL	Alternate Gatekeeper List
ARS	Automatic Route Selection
ASAI	Adjunct Switch Applications Interface
AVD	Alternate Voice/Data
AWOH	Application Enablement Services
BA	Bridged Appearances
BRI	Basic Rate Interface
BSR	Best Service Routing
BTD	Busy Tone Disconnect
CDR	Call Detail Record
CLAN	TN799 Control LAN circuit pack that controls TCP/IP signalling and firmware downloads
CMS	Call Management System
CTI	Computer Telephony Integration
DCP	Digital Communications Protocol
DCS	Distributed Communications System
DECT	Digitally Enhanced Cordless Telecommunications
DID	Direct Inward Dialing
DMCC	Device Media and Call Control
DPT	Dial Plan Transparency
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
EECCR	Error Encountered Cannot Complete the Request, usually indicates data corruption
EPN	Expansion Port Network
ESS	Enterprise Survivable Server
FAC	Feature Access Code
FNE	Feature Name Extension

GUI	Graphical User Interface
HEMU	Home Enterprise Mobility User
IGAR	Inter-Gateway Alternate Routing
IP	Internet Protocol
IPSI	Internet Protocol Server Interface
IPTF	IP Toll Free
ISDN	Integrated Services Digital Network
ISG	Integrated Services Gateways
IVR	Interactive Voice Response
KARRQ	KeepAlive Registration Request
LAI	Look Ahead Interflow
LAN	Local Area Network
LAR	Look Ahead Routing
LLDT	Link Loss Delay Timer
LDN	Listed Directory Number
LSP	Local Survivable Processor
MG	Media Gateways
MIB	Management Information Base
MM	Modular Messaging
MOH	Music on Hold
MWI	Message Waiting Indication
NCR	Network Call Redirection
NFAS	Non Facility Associated Signaling
OSPC	Avaya Softconsole OSPC is an Avaya product, where OSPC stands for Operator Set PC
OSSI	Operations Support System Interface
PAM	Pluggable Authentication Modules
PCD	Packet Control Driver
PN	Port Network
PRI	Primary Rate Interface
PSA	Personal Station Access
PSTN	Public Switched Telephone Network
PBX	Private Branch eXchange
QSIG	International Standard for inter-PBX feature transparency at the Q reference point
RFU	Remote Field Update

RTP	Realtime Transport Protocol
SAC	Send All Calls
SAFE	Self Administration for EC500
SAT	System Access Terminal
SBA	Simulated Bridge Appearance
SBS	Separation of Bearer and Signaling
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SMI	System Management Interface
SRTP	Secure Real-Time Protocol
TAC	Trunk Access Code
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TLS	Transport Layer Security
TSRA	Time Slot Record Audit
TTI	Terminal Translation Initialization
TTS	Time To Service
URI	Uniform Resource Identifier
USB	Universal Serial Bus
USNI	United States Network Interface
VALU	Value-Added
VDN	Vector Directory Number
VEMU	Visitor Enterprise Mobility User
VLAN	Virtual LAN
VOA	VDN of origin Announcement
WAST	Wait Answer Supervision Timeout