Communication Manager 4.0.4 SP#3.02 Release Notes

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Preventing toll fraud

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

Avaya fraud intervention

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

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Providing Telecommunications Security

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

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

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

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

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

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

Responsibility for Your Company's Telecommunications Security

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

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

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

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

TCP/IP Facilities

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

Standards Compliance

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

Federal Communications Commission Statement

Part 15:

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

Canadian Department of Communications (DOC) Interference Information

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

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

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

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

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Contents

Changes delivered to Communication Manager 4.0.4 SP #3.02

Communication Manager 4.0.4 SP #3.02 Release Notes

The **Communication Manager** service packs are cumulative and changes in **Communication Manager** 4.0.4 SP#1, SP#2, SP#3, and SP#3.01 are included in **Communication Manager** 4.0.4 SP#3.02. The changes delivered to **Communication Manager** 4.0.4 SP #3.02 are grouped as follows:

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

Product Support Notices

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

To read the PSN description online:

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

   The alphabetical list of documentation is displayed.

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

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

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 4.0.4 SP #3

Problem	Keywords	Workaround
This change allowed four new capabilities associated with 1X Mobile. 1. Allowing **ASAI** to originate a call from an unregistered **IP** station. 2. Allowing calls to terminate to a square bridged **CTI** station. 3. Allowing **ASAI** to originate a call from a bridged appearance administered on a **CTI** station. 4. Allowing a **CTI** extension to have an off-premise mapping associated with it.	083400	
When the duplication link went down, a major alarm was logged immediately without any resolution to retire the alarm.	091287	

Problems fixed in Communication Manager 4.0.4 SP #1

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

Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***1 of 21***

Problem	Keywords	Workaround
SIP Station was showing **Administered WithOut Hardware** (**AWOH**) Station's name and number instead of **Vector Directory Number's** (**VDN's**) name and number when call is placed to **VDN** which is routed to **AWOH**. This problem used to occur only for **SIP Stations.**	063430	
The maintenance audit for **Music-on-Hold** incorrectly identified problems when **Tone-on-Hold** was administered.	071699	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***2 of 21***

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 used to occur 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** and **S8730** servers running in software duplication mode.	071816	
In a **H.323 QSIG** network of **I55** and **Communication Manager Servers**, in some scenarios, an **I55** user did not get the usual display update when a far-end **I55** user put the call on hold.	072014	
When a **Session Initiation Protocol** (**SIP**) user completed an attended transfer or an attended conference of an incoming call to a hunt group which has a **SIP** endpoint as its member, the call was dropped.	072326	
The `list history SAT` command could result in a system restart, requiring a manual linux reboot to recover.	072747	
Certain internal conditions might cause a service-affecting restart (level 1, 2, or 4).	072848 082120 082221 082284 082616	
A **CLAN** link down followed quickly by link up could result in a **TN799** (**CLAN**) being left in an out-of-service state. This can be caused by an administration change that disables and enables the **IP Interface**.	072871	Disable the interface, wait for 10 seconds, and then re-enable the interface.
When an **H.248** gateway registered with a **Communication Manager Server**, the interval to bring **DCP Phones** associated with that gateway into service could be as long as 10 to 20 minutes.	073319	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***3 of 21***

Problem	Keywords	Workaround
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.
When agent **A** on switch **A** calls across a **QSIG** trunk to agent **B** on switch **B** and agent **B** transfers the call back across the **QSIG** trunk group to switch **A** to a number that routes out of switch on another trunk group, when path replacement happens the call may not be tracked properly by **CMS**.	074063	
If **A1** calls **B1** over a **QSIG** trunk and **B1** answers the call and transfers it to a **local VDN** which vectors into a 10 seconds music step and then the call is routed to a remote station which answers it, then path replacement may fail resulting in poor voice quality.	074281	
In certain call scenarios, calls covered to a **Modular Messaging** adjunct might have encountered a non-integrated greeting, instead of getting a specific party's voice mail box.	074295	
When the **IPSI** socket sanity timeout was set higher than three, certain outages could cause **Communication Manager** to execute a spontaneous **IPSI interchange** of the **IPSIs** at three seconds instead of at the **IPSI** socket sanity timeout value.	074344	
Calls made on a **CMS** measured trunk group administered for **Network Call Redirection** and sometimes all-trunks-busy were incorrectly tracked and counted by **CMS**.	080038	
Using the `backup -t` command to look at the backup results shows no failures, only successes. This can erronously cause alarms to be resolved that still exist. This has been corrected to show proper status.	080207	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***4 of 21***

Problem	Keywords	Workaround
On call into a **Meeting Exchange bridge** from an **IP Phone** at a remote gateway (**G700**), the first time the passcode was entered the digits were sent incorrectly and the passcode was rejected.	080327	
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	Execute a **System Warm Reset**.
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	Ensure that **Path Navigator** endpoints initiate calls at a bandwidth lower or the same as the **Communication Manager** bandwidth.
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 be used 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** (**Integrated Services Digital Network**) trunk with calls involving **Crisis Alert** was sometimes incorrect causing the call to fail.	080479	
Eliminate traps and **system resets** that may occur in systems using the **Dial Plan Transparency** (**DPT**) feature.	080542	
If a `Status Station` or `Status Trunk` command was entered for a user on a call that involved more than 10 other ports (For example, due to the effects of bridging or group paging), incorrect data was written into a **Communication Manager** table, corrupting some loss values. Those bad loss values then caused **TDM** bus maintenance tests to receive incorrect results, forcing some **TDM** bus timeslots to be taken out of service and reducing call capacity.	080735	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***5 of 21***

Problem	Keywords	Workaround
Reduce the **USB** alarms on **S8500B/C** and **S8400** servers.	080942	
Certain memory-corruption scenarios for **H.323 IP** traffic were not cleaned up properly by the **IP_ENDPOINT** maintenance audit.	080956	
When a call was made with the **Busy Indicator** button and was covered from the **called-to** Station to a third Station, when the third Station answered the call and was connected to the calling Station, the busy indicator for the covering Station was out. When the answering Station hung up, the busy indicator on the calling Station for the covering Station incorrectly turned on.	080972	
Avaya One-X Desktop Edition 1.x failed to register with **Communication Manager**.	080990	
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** through **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** shows name and number of **user B** instead of **user A**.	081048	
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:** If it is plugged back within six seconds then the button labels may not get updated.	081094	
If an **IP** agents phone was connected to an **H.248 Media Gateway** and there were no zip tones enabled, the agent was sometimes connected to a **VOA** announcement while on an active call (outside of the usual contexts of when **VOA** is expected such as at the start of call, or when the **VOA repeat** button is selected).	081126	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***6 of 21***

Problem	Keywords	Workaround
Under rare configurations, **ESS** return to main could lead to lost synchronization.	081247	Ensure that the entries in the **change sync atm** form used an existing DS1 instead of an ATM-SW entry for a sync reference.
If a **SETUP** message received with invalid content calls remained in queue even though agents were available.	081261	
The call from different multinational locations did not shuffle eating up media resources. To reproduce this problem, enable **multinational** feature and configure 2 locations with different location-parameters and tone generation plan. Configure a port network in region 1 location 1 and a gateway in region 2 location 2. **DCP Phone** in location 1 calls **IP Phone** in location 2 which has auto-answer enabled for all calls. The call got answered, but was not shuffled.	081286	
There were display issues for minor digits in firmware version on **status Station <Extension>** and **list registered ip-stations** form.	081301	
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	
Toshiba SIP phone-A calls bridge appearance or **Vector Directory Number** administered on **Toshiba SIP phone-B** over **PRI trunk**. When call is answered on **Station-B** it displays trunk name instead of **Station-A**'s name. Problem is visible only if **Send Number** is set to **y** and **Send Name** is set to **n** on trunk form.	081351	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***7 of 21***

Problem	Keywords	Workaround
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 1, 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	
Following problems are reported for Station **64XX** series: 1. When headset button is **ON**, calling party information gets cleared after 30 seconds even if call is ringing. 2. The **64XX** Station sets with the **Headset** button **ON**, are not updating their display after getting a missed call. 3. Two calls are ringing on a Station **64XX** having **Headset** button administered. First call is answered with **Headset** button and transferred to another Station. After transfer is complete, Station **64XX** shows time and date even though second call is ringing.	081420	
In the case of a video IP trunk between two **Communication Manager**s where **G.722-64K** is included at both ends in the `ip-codec-set` forms, an audio-only call across the trunk from an endpoint that supports **G.722-64K** to an endpoint that does not, the call would appear to succeed but will not get audio.	081459	Remove **G.722-64K** from the `ip-codec-set` form from the endpoint where it is not supported.
BTD (**Busy Tone Disconnect**) trunks involved in a conference call were not disconnected when the corresponding far end party dropped.	081460	
Rarely, the **TN775 EPN Maintenance Board** would get into an alarmed state that could be cleared only to return in 15 minutes or less.	081481	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 *8 of 21*

Problem	Keywords	Workaround
When a user made a call across a **SIP trunk** to an extension that does not exist on the far end, and the far end has 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	
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**. The issue always arises.	081598	
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 *talkpath*.	081640	Disable the autoanswer on digital **CallMaster**.
A customer could not execute **`change abbreviated-dialing system`** command due to translation corruption caused by a bad **Customer Defined Button Label** file.	081651	
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** did not answer, and the first coverage point of **Station C** is an **IP trunk,** that covered call would drop.	081727	
A distorted/double ringback tone was heard on an **Avaya SIP Phone** for an outgoing public network call passing through **Nextone**.	081737	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***9 of 21***

Problem	**Keywords**	**Workaround**
A customer who made a call from a Path Navigator registered video system (For example, **Polycom VSX**) through 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**. OR • 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. OR • Don't call audio endpoints through trunks from **Path Navigator Systems** (this is not a particularly useful thing to do as the point of Path Navigator is to support video).
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***10 of 21***

Problem	Keywords	Workaround
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	
Attendant was not receiving second wakeup reminder call for **vip-wakeup**. This problem was occurring only when **do-not-disturb** button was activated on guest station and **Cancel Do-Not-Disturb for Wakeup Calls?** in **system-parameters hospitality** field is set to **y**.	081749	• Keep **Do-not-disturb** button deactivate throughout OR • **Cancel Do-Not-Disturb for Wakeup Calls?** is set to **n**.
When a public network incoming call to a pickup group member is answered using **call-pickup feature** button by another member of the same pickup group, after the call is covered on that member, the display does not show the calling number correctly. It shows some invalid character in it. You will see this problem when you use **call-pickup feature** with coverage to answer the public network call with or without **Calling Party Number** (**CPN**) prefix defined for the pick-up member.	081750	
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	
Two parties **SIP** calls did not shuffle when using **G.726 codec**.	081767	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***11 of 21***

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**) and **0005** is the **ACD Hunt Group**.	081783	Agents should login to the **ACD** by manually entering the **FAC** and the rest of the information.
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.
When misoperation alerting is turned on, calls to voice mail or coverage point does not drop intermittently, hanging the port. Steps and administration to reproduce the problem: 1. Enable **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 calls another Station. Don't answer the call. 3. Analog Station puts the call on hold and calls x-ported analog Station which has coverage path administered. 4. Coverage point is ringing. Analog Station hangs up. 5. The coverage call is not dropped.	081785	Turn off **misoperation alerting**.
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	
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 talkpath* between the **DCP Phone** (A) and **IP Phone** (D).	081823	To establish the *talkpath* for one of the two parties on the call to hold/unhold the call.
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***12 of 21***

Problem	**Keywords**	**Workaround**
An incoming **SIP** trunk call would sometimes fail to complete if it was routed to an **ISDN trunk** through **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 previously 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 expectation is that basic video call setup should proceed between the two vendors equipment through standards based **H.323** (**H.245**) video trunking that is 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) through **IP trunk** to **Avaya Communication Manager**. Fast busy tone is observed. 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.
A call from an **Avaya SIP Phone** to **Cisco**, through **NexTone**, was dropped when it was transferred to another **Avaya SIP** through **NexTone**.	081859	
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	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***13 of 21***

Problem	**Keywords**	**Workaround**
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 *talkpath*.	081875	Disable the autoanswer on digital **CallMaster**.
IF ● An incoming call arrives through 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 through 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	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***14 of 21***

Problem	Keywords	Workaround
When a **SIP user** did an attended transfer of an incoming call to a **Hunt Group**, which has 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**.
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**.	081919	
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	
A race condition existed between a new **Music-on-Hold (MoH)** connection being established across network regions through **Inter-Gateway Alternate Routing** 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	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***15 of 21***

Problem	Keywords	Workaround
The duplication link drops whenever either server (**Active** or **Standby**) in a duplicated pair, running in software duplication mode, undergoes a software reload. This is expected behavior. However, when the link drops and the software reload occurs, a major duplication alarm (#2) was sometimes generated. This often occurred on **ESS Servers** following file synchronization. System behavior was otherwise unaffected. **Server(s) impacted**: **S8720** and **S8730 Servers** running in software duplication mode.	081943	
Extend Call was getting dropped whenever the **Hangup** button was pressed.The **IP softphone** 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	
When a **Customer Interaction Express** adjunct transfers a call back to **Communication Manager**, a break in *talkpath*, 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	
Customers with multipoint **Polycom VSX**s 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	
While troubleshooting problems, Avaya representatives may have needed to use utilities to collect data or traces. Under certain conditions the use of this tools resulted in a warm start.	082007	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***16 of 21***

Problem	Keywords	Workaround
The **SAMP firmware update** failed, if the modem was connected to the SAMP's USB port and incoming calls were enabled.	082015	
If an **IP Station** was on multiple calls, where it was receiving **VoIP** from a Media Gateway and the Media Gateway's link experienced a disruption that results in the Media Gateway re-registering with the main, under certain internal conditions, the system may have encountered a reset system 2.	082060	
If a **SIP trunk** between a **Communication Manager** and a **SIP Enablement Server** was used for outgoing calls then the calls may have been dropped approximately three minutes after they were established.	082061	
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, now 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 previously held call, the call drops.	082080	
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	
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	Use any of these special keywords in user-defined message translations.
Notify messages (message-summary) bombarded on **Communication Manager** from **SIP Enablement Server** caused overload 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	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***17 of 21***

Problem	Keywords	Workaround
When audit **552** (MO_FTING) runs, under some internal condition (For instance, corrupt data) the system may undergo a restart level 1 and further get escalated to levels 2 and 4. This will lead to an interchange of servers in a duplex system. During this process calls may get dropped.	082146	Disable audit **552**.
During a **SIP** call, **Communication Manager** resets because of an internal error.	082152	
When **Headset** button is **ON**, for an incoming call the calling party information display is cleared after 30 seconds even as the call is ringing.	082153	
Extend call was getting dropped on pressing hangup from **IP Sotfphone**.	082206	
A denial event was sometimes reported while listening to a music source. This occurred only when: 1. Tenant partitioning was enabled. 2. **Facility Access Test** feature was used to listen music source. 3. Music sources were not stored sequentially.	082320	
Multiple link bounces of a gateway can cause problems with recovery of **D-channel** links for **ISDN PRI Interfaces**. The second link bounce can get interpreted as a short link bounce and cause **Communication Manager** and the **H.248 Media Gateway** to go out of synchronization.	082328	
When **Communication Manager 4.0.3** was configured to run **Call Center 3.0**, the system provided access to only 999 vectors instead of the 2000 vectors that are available in **Call Center 3.0** for the **S87xx** and **S85xx** media server platforms. 2000 vectors became available in **Communication Manager 3.1** as a standard vector capacity when the **Call Center** release is **3.0** or greater.	082332	
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	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***18 of 21***

Problem	**Keywords**	**Workaround**
Under specific conditions, every Station in a call (For example, the primary parties plus bridged users) may not alert when the **Inter-Gateway Alternate Routing** (**IGAR**) feature is 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 **Inter Gateway Alternate Routing**, a delay of four to five seconds in *talkpath* was observed. Call from **IP DECT** in Location 1 to another Station in Location 2 works fine.	082376	
Call to **listed-directory-numbers** (**LDN**) will fail when administered above 15th position. Steps to reproduce: 1. Administer **listed-directory-numbers** at 16th position or above administer **attendant-group**, **Tenant** and **listed-directory-numbers** to route to attendant, **Enable Night service** on attendant. 2. Make a call to **listed-directory-numbers** group. 3. Call will fail.	082389	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***19 of 21***

Problem	Keywords	Workaround
For a **6402D** type **DCP phone**, when the **IP Softphone** field was enabled and the multimedia mode was set to **enhanced** on the **Station** form, the display was truncated when the **6402D desk phone** was used. If a softphone was used, the display was fine. If the **softphone** field was disabled and **multimedia** mode was set to **basic** the desk phone display was fine. This problem occurred always when a **6402D type DCP phone** has **IP Softphone** field set to **y** and received an external call or an internal call from another Station with a 7-digit extension.	082433	
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 service observer use to observe an active call on the agent wherein the agent is involved in a **single step conference** then the service observer used to display **calling to Conference 2 so**. It should show **calling to called so**.	082475	
When the `trace-route ipaddress` command is run many times, the **SAT** session will hang and the **Communication Manager** may go through a warm restart.	082494	
Station A on **Communication Manager** has a coverage path, this coverage path has a cover point as coverage answer group. Now this coverage answer group has all its member as **IP DECT Station**. When an external call is made to **Station A**, it will go to cover when nobody answers it and if any of the **IP DECT Stations** in coverage answer group are switched off then rest of the **IP DECT** members of the coverage answer group will not be alerted.	082497	
Attendant was not receiving second wakeup reminder call for **vip-wakeup**. This problem was occurring only when **do-not-disturb** button was activated on guest Station and **Cancel Do-Not-Disturb for Wakeup Calls**? in **system-parameters hospitality** field is set to **y** and extended **Do-not-disturb** is set to **terminate** at the same time as that of **vip-wakeup**	082507	
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, administered **mg-recovery** rule instead of **blank**.	082514	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***20 of 21***

Problem	Keywords	Workaround
Occasionally, in systems utilizing **H.248** gateways, a memory error could occur (not externally noticeable in system behavior).	082516	Disable the audit that monitors the state of **H.248** terminations.
On inter-gateway incoming calls to agents, the agent may have heard *crosstalk* if the agent did a hold and unhold of the call.	082522	
IP phones with **EC500** mapping do not alarm if they fail *keepalives*.	082543	
● On **SAT** form **change station** page 3 of a monitored **Station A,** one kind of enhanced call forwarding is filled with an extension number which is the extension number of monitoring **Station B** and **Active** flag is set to **yes**. The **Enhanced Call Forwarding** button on monitored **Station A** indicates that at least one kind of enhanced call forwarding is active, but the **Team** button on the monitoring **Station B** doesn't change its appearance to indicate that monitored **Station A** has at least one active enhanced call forwarding towards the monitoring **Station B**. To have this misbehavior the followed setting are necessary: ● Monitoring **Station B** has a **Team** button assigned which points to monitored **Station A**. ● Monitored **Station A** has 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	
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 Networks** restart for some of the port networks.	082710	
The transfer to voice mail feature access code does not work for all scenarios if the voice mail system is trunk (**PRI**, **H323**, **SIP**) integrated and **Communication Manager** is translated to disallow trunk to trunk transfers.	082843	
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Table 2: Fixes delivered to Communication Manager 4.0.4 SP #1 ***21 of 21***

Problem	Keywords	Workaround
In an **NFAS** arrangement with **Backup D channels**, with the **D-channels** on two 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 two 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. Thereafter, 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.	082846	
A system with **H.248** media gateways and ephemeral caching enabled would frequently go through resets unnecessarily.	083005	Turn **OFF** ephemeral caching or disable the maintenance internal data audit.
When transfering a call to an endpoint that goes to coverage, which is a **VDN**, the transfer is denied.	083297	Change the coverage path to avoid use of **VDN**.
Server resets could occur as a result of **Communication Manager SIP** trunk traffic.	083322	
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Problems fixed in Communication Manager 4.0.4 SP #2

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

Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***1 of 15***

Problem	Keywords	Workaround
For the calls originated by Telecommuter Softphone using feature buttons such as **Last Number Dial**, **Abbreviated Dial**, **Autodial** would dial the destination number without waiting to service link come up.	061716	
Failing modems on the server did not cause alarms.	061759	
For outgoing calls made to a trunk from a **SIP** phone under certain conditions the first second of voice path was lost.	063394	
The recipient heard double the number of expected **DTMF** tones in certain **DTMF** modes.	063677	
A 9630 station was administered with off-pbx mapping to a station on another switch or to a mobile phone in another network region. An incoming call to this 9630 station was answered by the mobile/off-pbx station. The mobile/off-pbx station conferenced another party using the "Conference on Answer" Feature Name Extension setting which was administered by the command **`change off-pbx-telephone feature-name-extensions set`**. The second call-Appearance on the 9630 set (principal station) remained active even after all the calls were dropped. This could be cleared by selecting the second call appearance for a call and then hanging up.	070111	
For Media Gateway systems with a Local Survivable Processor, if this Media Gateway had an analog or **DCP** board with no ports assigned, an Error Type 23 warning alarm, indicating a board was administered, but not physically installed, would likely appear within 15 minutes of the Media Gateway automatically returning to the Main server from the Local Survivable Processor (**LSP**).	070877	Execute a test board long until the alarm is cleared.
Crisis alert calls to attendent consoles went through tenant partitions.	071106	
A call made from a **SIP** endpoint to a **DCP** endpoint got dropped after sometime.	071293	
If a call was active on a **SIP** phone running SPARK firmware, and it rebooted, after reboot, the call was still present but unusable as the user could not bridge back onto the call.	072222	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***2 of 15***

Problem	Keywords	Workaround
When a call was received on a **SIP** trunk without a number, and that **SIP** trunk was part of a group that had it's **Replace Unavailable Numbers** field set to *y* on page 3 of the **change trunk-group** form, the trunk's access code (**TAC**) would be displayed on the receiving endpoint.	072510	
During Extension to cellular (EC500) service, when a cell phone dialed an active appearance select for its principal station, under certain conditions, the call appearance remained active after the call was dropped.	072594	
An incoming **IP** trunk call where the **IP** trunk and called party were in different network regions resulted in no ringback for the caller and no talk path if the call was answered.	073223	
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	
There was a system reset on duplicated systems soon after an upgrade.	073872	
When **Mask CPN/NAME for Internal Calls** on a **COR** form is set to '*y*', **Communication Manager** used to display caller's information on a team button's monitoring station when a call was made to the monitored station and monitoring station pressed the alerting **Team** button.	074108	
A corrupted non-**ACD** hunt group member record was being saved in translations. On a subsequent system re-boot the corrupted record would be recognized causing translation corruption. Once translation corruption occurred, the system was not allowing the saving of translations.	074222	Contact the Services team to have them clean the corrupted data. However, this would not prevent the corruption from re-occurring.
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	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 *3 of 15*

Problem	**Keywords**	**Workaround**
When integrated announcement was used as the music source, when a trunk call was transferred and recalled, the station display incorrectly showed conference.	080393	
When a ringing call on a monitored station was picked up by a monitoring station through a **CTI** application, audible ringing did not immediately stop after the monitoring station answered the call.	080644	
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	
In situations where a station has more than one coverage point and the call covers to the next coverage point, the previous coverage point does not stop ringing when call covers to an Octel voice mail adjunct connected to the **Communication Manager** with **QSIG** value trunk.	081024	
If an **IP** station was on multiple calls, where it was receiving **VoIP** from a Media Gateway and the Media Gateway s link experienced a disruption that resulted in the Media Gateway re-registering with the main, under certain internal conditions, the system sometimes encountered a reset system 2.	081059	
An external application e.g. **AVAYA Softconsole OSPC** expects **CTI** events if a call is picked using the **Team Button Pickup** functionality. The application in this case the **AVAYA Softconsole OSPC** cannot change the displayed status of a phone due to the missing events and present the phone permanently in a busy state.	081067	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***4 of 15***

Problem	**Keywords**	**Workaround**
In case of call forwarding Busy/Don't Answer (DA), Enhanced Call forwarding Busy/DA, **Communication Manager** did not send Adjunct Switch Application Interface (**ASAI**) events of forwarded call. Due to this user did not get an alert message on his application for the redirected call. Now with this fix user can see an alert message whenever call gets redirected due to no answer.	081068	
Outgoing calls made to a trunk from a **SIP** phone resulted in a loss of voice path for the initial second.	081145	
An announcement was not played for Restricted dialed number.	081230	
Abnormal temperature readings for the S8710, S8720, and S8730 servers were indicated only by a major alarm trap. A new minor alarm trap had sent providing earlier warning of such a condition.	081390	
The customer would have no talkpath in some cases. Typically the configuration would be like the one explained below. There are 2 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.	081395	Use an IP trunk instead of the SIP Trunk.
When a customer upgrades from a Prologix to an S8400 **Communication Manager** server platform, they could not add or change hunt groups.	081427	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***5 of 15***

Problem	Keywords	Workaround
If a non-shufflable IP endpoint, getting **VoIP** (Voice Over Internet Protocol) from a Media Gateway, was active and talking on call appearance one (1), then pressed call appearance two (2), getting dial tone, then immediately went back to call appearance one (1), there was no talk path.	081430	
DTMF digits were not heard by non-IP endpoints administered on gateways.	081595	
B-Channel out of service coming back into service state was not reported to **CMS**.	081633	
When the wait answer supervision timer expired on an **ISDN** trunk, the trunk was getting disconnected giving an invalid cause value.	081647	
This problem was specific for the **Avaya Digital Terminal for Japan** also known as J24. A J24 station was configured with Bridged Appearance of two **AWOH** stations. This digital terminal received a call on the first Bridge Appearance. While that bridge appearance was still ringing, it received another call on the second bridge appearance, and the display of the digital terminal was updated to show the information of the second call. It was expected that it continued to show the information of the oldest call.	081725	
While administering Native name and Script tag fields on Avaya Site Administration only Script tag was specified and Native name was blank. This caused translation corruption	081744	
In the case of Call-Fwd Off-Net, the display on a **SIP** phone (Calling Party) showed the **ARS/AAR** code along with the administered/call-forward destination number. This happened always when a **SIP** phone placed a call to a station which forwarded all the calls Off-Net.	081838	
Agents intermittently did not hear zip tone or did not get talkpath after retrieving a call from hold.	081896	
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.
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***6 of 15***

Problem	Keywords	Workaround
User could not hear **DTMF** (Dual-tone multi-frequency) side tone while dialing out using **IP**-Agent.	082034	
Communication Manager in certain cases disallowed the transfer of an active call, when that call was established via **QSIG** diversion with rerouting.	082090	
If an incoming **QSIG** call was transferred into a **VDN**/vector that had as its first step 'wait step hearing music', **Communication Manager** sent to the calling user via **QSIG** an incorrect indication that the call was alerting instead of answered.	082196	
The "-d" option from the testled command was deprecated. Also, when the "-a" option was selected, it did not test the duplication card LEDs.	082302	
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	
The **change modem settings** button on the configure modem web page of configure server will fail with an error when one of the newer Avaya supported modems was attached to the server.	082485	
Caller at some customers heard **DTMF** digits when answering a call whose talk path was set up by the Inter-Gateway Alternate Routing (**IGAR**) feature. The problem happened only if the **IGAR** connection was routed through a non-standard **ISDN** network such that the `ISDN CONNECT` message was delayed by several seconds, though the talk path was set up quickly.	082491	
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	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***7 of 15***

Problem	Keywords	Workaround
For **PRI-DECT** b-isdn termination on one gateway and any other kind on termination at another gateway, on doing hold/ unhold on the PRI-DECT talkpath lost. After the fix, on doing hold/unhold talk path existed.	082615	
If a system had hundreds of unnamed H323 endpoints registered and a network outage occurred that lasted longer than eight minutes then the system did undergo WARM reset 1 and COLD_2 reset	082638	
Calling party number information may be lost for vector adjunct route steps if path replacement occurred for the call while the call was in vector processing.	082681	
In certain cases, calls involving physical ports on H.248 media gateways may experience call failures and/or no-talkpath problems.	082682	
Some customers were seeing multiple instances of extensions when executing the `list monitored-station` command.	082702	
An incoming trunk call was answered by an **AAS** (Auto Available Split) agent and then force transferred to another agent. If the forced transfer failed, the transferring agent was left in a bad state and the incoming trunk call was not dropped even if **Intercept Treatment On Failed Trunk Transfers?** field on page 17 of system-parameters features was set to "*n*".	082758	
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 call could not be redirected to a station using **ARS** when all of the following conditions were met: 1. Call was redirected using **ARS** 2. **ARS** analysis table for the target station was set with different min and max values (for example, min is 4 and max is 8) 3. Number of digits of the target station, was less than the max limit set in the **ARS** table (for example, number where the call is redirected is 12345, that is, 5 digits when max is set to 8) This problem was observed whenever the target number (where the call is redirected) had less digits than the maximum allowed by the ars analysis table.	082793	Either assign the same value to min and max fields in the ars analysis table, or to make the length of the target number the same size as max allowed digits.
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 *8 of 15*

Problem	Keywords	Workaround
When a third party make call was executed using **ASAI**, with only **ARS** code:that is, if an **ASAI** application dials only 9 (assuming 9 is **ARS** code) without dialing any more digits, then the station was bound to dialtone. While the expected behavior was, in such situation, the station would receive intercept signal and after certain time (timeout) call would be dropped (station should be on hook).	082815	
Loudspeaker paging did not work correctly when the pager was in a fiber-connected port network and was paging parties in **IP**-connected port networks.	082854	
96xx **IP** station could not blind transfer the call to station which has call-forward activated.	082859	
Calls may have failed when using **ARS** to route calls to a Cisco **SIP** phone which then blind transfered the call to a **VDN** with number of the "route-to number" set so that the call would terminate at a **SIP** endpoint.	082869	
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	
In some call scenarios a telephone could end up ringing either when it should not or after the call has ended. The scenarios were limited to time-of-day coverage and if the coverage points in the selected coverage path had bridge appearances of the called party. In addition, the administration field **Terminate to Coverage Pts. with Bridged Appearances** must be set to *yes* on the coverage path.	082910	
Under certain unusual circumstances, a software restart could occur.	082920 082931	
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	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***9 of 15***

Problem	Keywords	Workaround
In the past, Integrated Management products did not display "list VDN" data properly.	082946	
Intermittently any analog station could not initiate auto callback to any analog station.	082955	
The performance of systems running software server duplication was not as good as it could have been.	082998	
The **CDR** (Call Detail Recording) output was not generated when all of the following circumstances occurred: - the user dialed using the **ARS**/**AAR** short-cut dialing feature - an authorization code was required to place the call - on the dialplan parameters form the fields **AAR/ARS Internal Call Prefix** and **AAR/ARS Internal Call Total Length** were both left blank.	083013	
Interaction of Coverage of Calls Redirected off-net with Look Ahead Routing (**LAR**) and a busy coverage point caused the caller to hear silence.	083045	
Communication Manager sent back a clearing cause value of "user busy" (instead of "no circuit or channel available") when it is unable to route out of the server due to a lack of compatible trunk resources. Steps to duplicate problem: - administration or feature(s) that must be active: Administer **ISDN PRI** trunk between Switch A & Switch B, and another trunk between Switch B & Switch C. Administer **UDP** remote extension on Switch B to make call to Switch A & Switch C extensions - specific order of events): Make a call from Switch A to Switch C via Switch B when all member of trunk (which Switch B to Switch C) are busy. Switch B send the DISCONNECT ISDN message with cause value 17 User Busy and list trace tac shows Denial event 1012 Destination Unavailable on Switch B.	083095	
On enabling look ahead routing with rehunt scheme, trunk lockups seen if look ahead routing failed on a member and the call successfully routed on a subsequent member.	083138	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***10 of 15***

Problem	Keywords	Workaround
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. The problem can be reproduced by as follows: Administer an **AAS** agent with **AAS** skills on the system. Remove one of the skills through a **FAC**. On the **SAT**, enter **`monitor bcms skill`**. The display shows the old status (the skill is still logged in for the agent). Adding a skill via a **FAC** results in the same behavior.	083142	Make the skill changes using **CMS**. OR "busy station" and "release station" to update the status.
Sometimes AVAYA **IP** Endpoints would fail to register after registration of Non-AVAYA Endpoints.	083144	
For tenant partitioned switches, a trunk call to an attendant that timed out and went to night service failed to ring at the night service station the first time after night service was activated. Subsequent calls to attendant in night service went fine.	083170	
ASAI adjunct route coupled with Look Ahead Interflow (**LAI**) caused tracking of the call by **CMS** to be aborted. The customer scenario that caused the problem was as follows: - An incoming call to **VDN**-1/vector-1 interflows to **VDN**-2/ vector-2 which does an adjunct route to give control to an **ASAI** adjunct. - The **ASAI** adjunct directs the call to an agent. - The agent starts a conference, putting the call on hold and calling **VDN**-3/vector-3. - This interflows to **VDN**-4/vector-4 and again does an adjunct route giving control to the **ASAI** adjunct. - The **ASAI** adjunct sends back a route request, sending the call out on a trunk to the **PSTN**. - While **Communication Manager** is waiting for feedback from the **PSTN** on the outgoing call, the agent completes the conference call, joining the incoming call and the outgoing call together. - **Communication Manager** then receives the ALERT or PROGRESS message back from the PSTN, causing **Communication Manager** to send an unexpected message to CMS. This results in the calls being ignored by CMS.	083202	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 *11 of 15*

Problem	Keywords	Workaround
The Network Region Audits that were run as result of a WARM restart after an interchange, could not handle some out-of-bounds data appropriately and resulted in an Escalated COLD_2 restart.	083270	
Avaya Communication Manager was not compliant with RFC 3262 in the sense **Avaya Communication Manager** queued final and other provisional response till PRACK was received in use cases where 100rel was supported for reliable provisional response.	083326	
If an H.323 **IP** station was connected to an 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	
Under particular circumstances when an incoming public call was routed over a multifrequency trunk and the call routes to the auto attendant, only a small portion of the announcement was heard before the call was delivered to the final destination.	083352	
On a **Communication Manager** with **LSP**s, sometimes the **LSP**'s Keep Alive Registration Request (**KARRQ**) could cause a restart on **Communication Manager**.	083369	
On a **Communication Manager** with **IP** trunks, sometimes the **IP** trunk call could cause a restart on **Communication Manager**.	083370	
If an **IP**/**SIP** signalling group was administered with the following attributes: **Direct IP-IP Audio Connections? y** and **DTMF over IP: rtp-payload** then under certain circumstances, when a user would press digits on his/her phone, those digits would not be heard on the far end of the corresponding trunk. In particular, if a **IP** station (which can shuffle) called over the trunk, no digits pressed would be heard on the far end of the trunk.	083386	
Under rare conditions, invalid internal data associated with a **CMS** SPI link may have caused a warm start to occur.	083390	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***12 of 15***

Problem	Keywords	Workaround
With the **Wait Answer Supervision Timer** set to *yes* on the **System-parameters Feature-Related** form ("change system feat"), a call came into Agent 1. Agent 1 transferred the call to Agent 2. Agent 2 did not answer the call. After 50 seconds (the Wait Answer Supervision Timer), an abandon message was sent to **CMS**. Before the following idle message could be sent to **CMS**, an audit started. This caused **CMS** to lose data.	083410	
When a TIME message was sent to **CMS** slightly before the minute (59 seconds) and a different message was sent to **CMS** after the minute (0 seconds) but before the next TIME message was sent, then **CMS** behaved as though the data collection clock had reset. This was reported in the Real Time Exception Log.	083440	
Some **SAMP** related server commands may have timed out resulting in `WriteXML Failed` errors being returned when the command was run.	083444	
Causing overload events on **Communication Manager**, which was resulting into no dialtone episodes.	083451	
Attendant Vectoring (a vector with **`queue-to attd-grp`** or **`queue-to attendant`** vector step commands) did not report the redirection to an attendant to **IQ** or **CMS**. This resulted in incorrect reports.	083459	
Permission values entered into the **isdn dcs-qsig-tsc-gateway P** field on the **USER PROFILE** and **USER PROFILE BY CATEGORY** forms would change when the **data-module D** field was changed by the user.	083468	
A service observer was not connected to a call when it was observing a station that answered the call through a personal CO line (**PCOL**).	083500	
When a call was transferred to a logical agent the call-log entry of the station where the agent had logged-in, did not update the incoming calls properly.	083557	
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	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 *13 of 15*

Problem	Keywords	Workaround
On rare occaions after an upgrade, command `status pnc` would incorrectly show the State of Health: as 'partially functional' with the 3rd entry in the Inter **PN** Index: having a non-zero value.	083561	
If incoming call from R2MFC/Analog trunk to station and another station pressed team button to monitor the call, display showed calling number or trunk-group name.	083568	
If incoming call from R2MFC/Analog trunk to station and station on pickup-group alerted with calling number or trunk-group name, then display showed different trunk-group name instead of Calling Number.	083586	
Call Detail Recording (**CDR**) account codes were not properly logged for an outgoing call if the outgoing call route preference had look ahead routing translated for the route preference.	083613	
Call center agents may not disconnect properly if the call center agent was routed to via a Vector Directory Number (**VDN**), which has its **Return Destination** field administered.	083679	
This change fixes a problem where Watchdog was recording multiple alarms on mdmtty starting and stopping.	083701	
When an **IP** phone attempted a blind or un-attended transfer of an active call to an another **IP** phone, under certain conditions the other party did not hear the ringback.	083723	
Service Packs cannot always be removed.	083749	
A condition existed on the customer **Communication Manager** that caused **IGAR** to be used to setup the voice path for a call between two phones. The destination phone had **EC500** administered and enabled. The caller did not hear ringback on the call although it alerted the destination. When the call was answered there was a normal voicepath.	083801	
Trunk to trunk transfer was not denied. This occurred always when **Trunk-to-Trunk Transfer** feature on **system-parameters features** form was set to *none* and a station that was on a trunk call, had a Single Step Conference (**SSC**) party listening to it, tried to transfer this trunk call over another trunk call.	083810	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***14 of 15***

Problem	Keywords	Workaround
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	
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	083874 083871	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.
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	
Station A had a Microsoft Office Communicator (**MOC**) client associated with it and called Station B. B did not answer and so the call went to B's coverage path, which means that the call went to a sip-adjunct hunt group of the Microsoft Unified Messaging (**MSUM**) Voicemail Server. Station A then received a reorder tone that it should not receive.	083932	
On some incoming calls to a **SIP** endpoint the display would not have contained the calling parties information.	083961	
When an incoming call to the attendant was transfered to a digital station which had a bridged appearance on a **SIP** station the system may have experienced a WARM reset.	090022	
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Table 3: Fixes delivered to Communication Manager 4.0.4 SP #2 ***15 of 15***

Problem	Keywords	Workaround
If an announcement was played when an attempt was made to connect to a call using an **ASAI** Single Step Conference request (for instance, by Witness call recording), the request was then allowed to proceed.	090027	
Nightly maintenance generated chronic PS-RGEN errors, primarily with error codes 257 and 513, on G650 power supplies. These appeared to be false errors because manual testing produced no failures.	090051	
If cabinet 1, 2 and 4 were administered, circuit packs resided in carrier A of cabinet 4, and the command list configuration carrier 3a was executed on the **SAT**, the circuit pack information for carrier 4a was displayed.	090072	
The **Avaya IQ** reporting adjunct could not track a transferred call in which the first call leg was unmeasured, the second leg was a measured **ACD** call, and the transfer was completed while the **ACD** call was ringing an agent.	090214	
A trunk call ringing at a **CMS** measured agent was dropped by Wait Answer Supervision Timeout (**WAST**). Before the associated trunk drops, an agent answered the **ACD** call. **CMS** identifies this as "two calls connected" and ignored the call.	090217	
If an agent with **IQ**/**CMS** measured and unmeasured skills answered a call in an unmeasured skill and then transferred that call, **Communication Manager** did not report the Agent-LoginID of the controlling party to **IQ**/**CMS**.	090248	
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.	090309	
Under certain conditions, **Communication Manager** systems with mixed fiber/**IP** Port Network connectivity may experience a restart. Disabling Port Network media-processor board audits may prevent the reset.	090654	
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Problems fixed in Communication Manager 4.0.4 SP #3

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

Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***1 of 13***

Problem	Keywords	Workaround
A call using G.726 and **SRTP** encryption, made over an **IP** trunk sometimes had no *talkpath*.	071525	
A trunk call to an **IVR** that subsequently went to coverage continues to hear Music on Hold (**MoH**) and never went to the coverage point. The scenario is as follows: A trunk call arrives at a vector, which sends the call to an **IVR** using a converse-on step - The **IVR** places the call on hold and dials an extension - The original call receives music on hold - If the dialed extension does not answer, but instead goes to coverage, the original call continues to receive music	071680	
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	
30% of R2MFC trunk calls tandemed to an **ISDN** trunk failed to complete.	080633	
If an incoming R2MFC trunk call to an **IP** station was forwarded over an **ISDN** trunk, the call failed sometimes.	080745	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***2 of 13***

Problem	Keywords	Workaround
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	
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	
When a **Communication Manager** user dialed an extension on non-Avaya system using an H.323 trunk, then sometimes the call failed. **Note:** This fix, along with the fix for 091815 changes **DTMF** event behavior. **DTMF** events on **IP** trunks no longer default to using Q.931/H.225 INFO messages with keypad information elements to send **DTMF** information. With non Avaya equipment, **Communication Manager** now opens H.245, or if H.245 is already open, **Communication Manager** sends the **DTMF** information as an H.245 alphanumeric string or an H.245 tone event, depending on what the non Avaya equipment has advertised for capabilities. This could require administration changes in **Communication Manager**.	081214	
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 could be seen on the telephones which had the lamps associated with feature buttons like the telephone models 96xx, 6408D+, 8410D, etc. The problem could be seen only when the buttons were presses rapidly without keeping time interval of even a second.	081259	Repeat the process (press the feature button followed by 2 and 0) or Press the button slowly keeping around 1 second time gap in each button press.
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***3 of 13***

Problem	**Keywords**	**Workaround**
Calls forwarded to a sip-adjunct voice mail went into an endless loop.	081262	
The customers **ASAI** applications received an unexpected Trunk Identifier **IE** in the response to an **ASAI** Party ID query for call id's involving **PRI** trunks. The Trunk Identifier **IE** had incorrect information for the call. Process errors accumulated three at a time for each query made.	081608	
Incoming calls to Media Gateways and Outgoing calls from Media Gateways were getting blocked due to resource exhaustion.	081611	
A PKTINT fatal fault was a major alarm. Now, A PKTINT fatal fault is a WARNING alarm for the first three minutes, and a MINOR alarm after three minutes.	081703	
A Nice recorder failed to record a port because the **Communication Manager** thought the phone was busy when it was not.	081710	
Customers monitoring stations with **ASAI** may see incomplete called party numbers when calls were manually placed from the monitored station over an **ISDN-PRI** trunk that was administered with overlap digit sending.	081804	
Under certain circumstances, systems with media gateways would see high occupancy readings, independent of the level of call processing traffic.	081870	
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 arrives at the forwarding station it is not always forwarded but sometimes stays ringing at the forwarding station. Servers impacted: All Linux Media Gateways impacted: Not Specific	082184	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***4 of 13***

Problem	Keywords	Workaround
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?" was set to "y" on "system-parameters features" form. This was 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	
During recovery of thousands of Time-to-service (**TTS**) phones it was possible that requests to establish sockets to the phones would overwhelm the TN799 (**CLAN**) board. In extreme conditions the **CLAN** would reset, forcing recovery of all the existing sockets on the board. The reset of the **CLAN** board delayed the recovery of the phones.	082315	
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	
When there were more than 15 **CTI** links the '`status aesvcs cti-link`' command displayed garbage on the second page when another status command was run at the same time.	082827	
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 contained an inappropriate setting (for example, a digital station loss group was specified for a digital trunk group) then features like Inter-Gateway Alternate Routing may not work as expected. A warning message would 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 was enabled, an incoming call to a **VDN** over a trunk that was in night service mode was routed to the **VDN** and the agent receiving the call could not answer it.	083047	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 *5 of 13*

Problem	Keywords	Workaround
SNMP Walk on g3ipintlist **MIB** Group displayed incorrect data for nodename and slot entries.	083102	
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	
When a given region had multiple interconnections (for example region 1 to 2, region 1 to 3, and region 1 to 4), and that region had failures with more than one of its interconnections, the "**`test failed-ip-network-region`**" command did not work correctly. The test would only do the first failed region pair and not successive region pairs correctly. The command showed garbage for the rest of the region pairs that it tested.	083126	
Dial Plan Transparency (**DPT**) calls failed in the following case: a) Look-Ahead Routing (**LAR**) was enabled on the route pattern set up to handle **DPT/IGAR** trunk calls. b) The calling phone was a **DCP** or analog phone (that is, not H.323 or **SIP**). Also, both **IGAR** and **DPT** calls failed in the following case: a) Look-Ahead Routing (**LAR**) was enabled on the route pattern set up to handle **DPT/IGAR** trunk calls. b) The call was rerouted using a later route pattern preference, because the initial **DPT/IGAR** call failed with an **ISDN** Cause value that triggers **LAR**.	083190	
After making multiple calls 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.	083216	
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	
For restricted call over trunk From: header for INVITE message contained anonymous@anonymous.invalid	083564	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***6 of 13***

Problem	Keywords	Workaround
The **Communication Manager** watchdog process would no longer reset if it had detected 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	
On doing an **SNMP** walk on the interfaces **MIB** group, the ethernet speed was incorrectly reported.	083688	
"`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	
Dial Plan Transparency feature was invoked towards an unplugged **IP** phone causing improper trunk usage.	083845	
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'.
Call was not dropping properly when answered on Covered party if "Simulated Bridge Appearance (**SBA**) at Principal" was turned on and Music On Hold was configured as Integrated music.	083911	Turn off "Simulated Bridge Appearance (**SBA**) at Principal" field on "change system-parameters coverage-forwarding" form.
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***7 of 13***

Problem	Keywords	Workaround
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	
'`display button-labels n`' command displays labels in the supported unicode language when phone was registered. To display the labels in the supported language, **Communication Manager** got language data from the end-point. When the phone was unregistered there was no end-point. Then **Communication Manager** would not get language data from the end-point so displayed in the default English language. Now we have added 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 EC500 sets dialed 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	
A call may get dropped when shuffling a call using a **SIP** (session initiation protocol) phone.	090040	
A system reset might occur when the Enhanced Call Forwarding feature was administered or the translations were saved (automatically or on administration request). Server(s) impacted: All Linux based servers Media Gateway(s) impacted: Not Specific	090108	
ISDN call setup retried as a result of glare conditions failed if Explicit Call Transfer or Two B-Channel Transfer supplementary service was active on the call. This error occured only when glare happened on a call which was setup due to vector ~r route-to step.	090130	
9620 set type could not cancel the call if there was a Single Step Conference party active on the held call.	090131	
Music was not played to the calling station when the Look-Ahead Interflow (**LAI**) checks were performed on the **Communication Manager**.	090140	
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 4.0.4 SP #3 ***8 of 13***

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 could get into a state where they would never come into service after both gateways have taken 2 link bounces that continued 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 **MG**s 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 **MG**s 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	
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	
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	
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.Avaya **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	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 *9 of 13*

Problem	Keywords	Workaround
Pickup alert was not sent to other members of the Pickup Group if the called member was an unregistered station.	090276	
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	
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	
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.	090390	Execute a COLD reset of the port network.
Customers recording calls using **NICE** and **AES** integration might not have calls recorded. The problem appeared when **NICE** was rebooted and could affect different stations each time.	090431	
Entering "**`list ip-tti-stations xxxx`**" (where x is a numeric value of length one or more) at the **SAT** (System Access Terminal) caused the command to fail. If the value was less than three digits, the **SAT** would output pages of useless data. If the numeric value was three 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	
Calls did not go to the EC500 when the Media Gateway to which the desk phone was connected was unregistered.	090472	
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Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***10 of 13***

Problem	Keywords	Workaround
A message about a non-service-affecting issue was printing in **Communication Manager** logs too often and consuming log space.	090508	
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.
Under heavy traffic conditions, **Communication Manager** could enter system overload if one or more H.248 media gateways reported H.248 errors.	090535	
When executing the **SAT** command "**`list skill-status`**", the value for the "Service Level" field did not change and always had the same value as the Hunt-Group value for Service Level Target Percentage.	090623	
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	
Executing the SAT command "**`change ip-network-region`**" may have resulted in overloaded resources to the point of a system reload. Now there are checks in place to limit the amount of resources the "**`change ip-network-region`**" command can use during its execution.	090625	
A call was made to a station A having EC500 feature enabled, this call was 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 does not receive **DTMF**.	090633	
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	
Correct sshd timestamps now appear in the secure log.	090650	
		10 of 13

Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***11 of 13***

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	Don't configure "auto-answer" mode to "acd" or "all" for an agent with Expert Agent Selection.
Busy out on H.323 trunks in certain intermediate call states was not allowed. The busyout command, in this case, would fail.	090793	
Under certain circumstances involving non-shuffable H.323 stations, 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	
An **IP** Softphone in shared control mode failed to login if lots of buttons on the phone and an expansion module were administered.	090878	
CallMaster V or 64xx stations did not clear the display when it was on a call with headset and transferred the call.	090883	
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	
Incoming trunk calls across a **SIP** trunk would occasionally fail.	091003	
		11 of 13

Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***12 of 13***

Problem	Keywords	Workaround
A call that covers to a **VDN** and routes out to an available agent could not be conferenced. Even though the call was no longer in vector processing and had been routed to an agent, that agent was not able 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 which 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 was tranfered to **VDN**/HUNT, the display on transfered party was showing calling party's information.	091025	
Intermittently, certain button pushes (like serv-obs) could be incorrectly denied.	091065	Remove the service observing port and add it back.
The Call Pickup feature had a special algorithm to determine which call was to be picked up next. The pickup display was updated to reflect any changes to the next call to be picked up. The display was not updating properly in case of Enhanced Call Pickup alerting.	091118	
The system restarted when an **ISDN BRI** endpoint was connected to **Communication Manager** and the endpoint sent an **ISDN** message containing an information element with an invalid length field.	091131	
No log entries were displayed with the System Logs web page when multiple views were selected or when a match pattern was entered.	091139	
Data for the g3trunksta **MIB** group displayed garbage values when a walk was performed on the g3mib.	091186	
		12 of 13

Table 4: Fixes delivered to Communication Manager 4.0.4 SP #3 ***13 of 13***

Problem	Keywords	Workaround
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 EC500 user dialed the idle call appearance **FNE** (Feature Name Extension) and then dial 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	
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	
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	
Calls failed to conference after covering and routing from a **VDN** to a valid extension.	091668	
Customers monitored 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	
An Issue associated with the following keyword was also fixed in **Communication Manager** 4.0.4 SP #3: 083476		
		13 of 13

Problems fixed in Communication Manager 4.0.4 SP #3.01

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

Table 5: Fixes delivered to Communication Manager 4.0.4 SP #3.01

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

Problems fixed in Communication Manager 4.0.4 SP #3.02

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

Table 6: Fixes delivered to Communication Manager 4.0.4 SP #3.02 ***1 of 2***

Problem	Keywords	Workaround
Wrong station heard **DTMF** tones when call was initiated using autodial button with ~p and DTMF digits.	092855	
IP Agent observed announcement cross talk on external inbound trunk calls to it, over Media Gateways.	092889	
		1 of 2

Table 6: Fixes delivered to Communication Manager 4.0.4 SP #3.02 ***2 of 2***

Problem	Keywords	Workaround
When an **IP** station conferenced in a non-**IP** station, and the **IP** station audio was encrypted with **SRTP** when the conference was completed, there was no audio on the **IP** station.	093118	
After receiving **DTMF** tones embedded into an incoming **RTP** streams, **Communication Manager** did not forward these **DTMF** tones over a H.323 trunk.	093119	
		2 of 2

Known problems

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

Table 7: Known problems in Communication Manager 4.0.4 SP #3.02

Problem	Keywords	Workaround
When a unicode-customized label is created with the backup/restore file for an **IP** phone, some characters might 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	
On an **S8730**, **S8510**, or **S8500** B/C server with an e1000 driver add-on **NIC**, and **VLAN** is administered on the **NIC**, when the interface shuts down, Linux commands for all interface administration fail. The Ethernet ports used by the e1000 driver add-on **NIC** on each server are: ● **S8500B**: Eth 2 and Eth 3, ● **S8500C/S8510**: Eth 3 and Eth 4, ● **S8730**: Eth 2, Eth 3, and Eth 4. See server installation and configuration documents for more information on the servers and ports.	081712	

Technical Support

Support for Communication Manager is available through Avaya Technical Support.

If you encounter trouble with Communication Manager:

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

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

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

When you request technical support, provide the following information:

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

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

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

Appendix A: Acronyms

AAR	Automatic Alternate Routing
ACD	Automatic Call Distribution
AES	Application Enablement Services
ARS	Automatic Route Selection
ASAI	Adjunct Switch Applications Interface
ATM	Asynchronous Transfer Mode
AWOH	Administered WithOut Hardware
BA	Bridged Appearances
BCMS	Basic Call Management System
BRI	Basic Rate Interface
BSR	Best Service Routing
CDR	Call Detail Recording
CLAN	TN799 Control LAN circuit pack that controls TCP/IP signalling and firmware downloads
CMS	Call Management System
CPU	Central Processing Unit
CTI	Computer Telephony Integration
DCP	Digital Communications Protocol
DCS	Distributed Communication System
DID	Direct Inward Dialing
DPT	Dial Plan Transparency
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
EPN	Expansion Port Network
ESS	Enterprise Survivable Server
ETSI	European Telecommunications Standards Institute
FAC	Feature Access Code
FNE	Feature Name Extension
IGAR	Inter-Gateway Alternate Routing
IP	Internet Protocol
IPSI	Internet Protocol Server Interface

ISDN	Integrated Services Digital Network
IVR	Interactive Voice Response
J24	Avaya Digital Terminal for Japan
KARRQ	Keep Alive Registration Request
LAI	Look Ahead Interflow
LAN	Local Area Network
LAR	Look Ahead Routing
LDN	Listed Directory Number
LED	Light Emitting Diode
LLDT	Link Loss Delay Timer
LSP	Local Survivable Processor
MG	Media Gateway
MIB	Management Information Base
MOC	Microsoft Office Communicator
MOH	Music On Hold
MSA	Multi-Site Administration
MSUM	Microsoft Unified Messaging
MWI	Message Waiting Indication
NFAS	Non Facility Associated Signaling
NIC	Network Interface Card
OSPC	Avaya Softconsole OSPC is an Avaya product, where OSPC stands for Operator Set PC
PCOL	Personal Central Office Line
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	Real-Time Protocol
SAMP	Service Access and Maintenance Processor
SAT	System Access Terminal
SAMP	Server Access and Maintenance Processor
SBA	Simulated Bridge Appearance

SBS Separation of Bearer and Signaling
SDP Session Description Protocol
SIP Session Initiation Protocol
SNMP Simple Network Management Protocol
SRTP Secured Real-Time Protocol
SSC Single Step Conference
TAC Trunk Access Code
TCP Transmission Control Protocol
TDM Time Division Multiplex
TTI Terminal Translation Initialization
TTS Time To Service
UDP User Datagram Protocol (RFC 768)
URI Uniform Resource Identifier
VDN Vector Directory Number
VEMU Visitor Enterprise Mobility User
VLAN Virtual LAN
VOA VDN of Origin Announcement
VoIP Voice over Internet Protocol
VSX A Polycom standard definition video room system
WAST Wait Answer Supervision Timeout