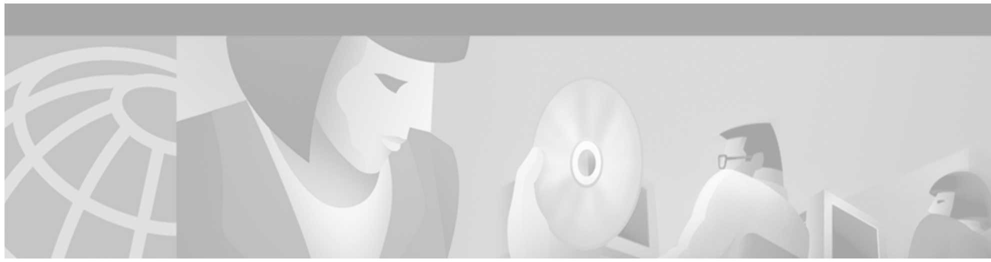

Release Notes for Cisco CallManager Release 3.1

August 2, 2001

These release notes describe the new features and caveats for Cisco CallManager Release 3.1.

For a list of the open and resolved caveats for Cisco CallManager Release 3.1(1), see "Resolved Caveats - Release 3.1(1)" section on page 17 and "Open Caveats" section on page 20. These release notes are updated every maintenance and major release.

Use these release notes in conjunction with the *Installing Cisco CallManager Release 3.1* document, located on Cisco Connection Online (CCO), and the Cisco Documentation CD-ROM. The *Installing Cisco CallManager Release 3.1* document comes with your CDs or convergence server.

Access the latest software upgrades and release notes for Cisco CallManager 3.1 on Cisco Connection Online (CCO) at

http://www.cisco.com/cgi-bin/tablebuild.pl/callmgr

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Contents

These release notes discuss the following topics:

Introduction

Cisco CallManager, a network business communication system, provides high-quality telephony over IP networks. Cisco CallManager enables the conversion of conventional, proprietary, circuit-switched PBXs to multiservice, open LAN systems.

System Requirements

Make sure you install and configure Cisco CallManager Release 3.1 on a Cisco Media Convergence Server.

You may also install Cisco CallManager on a Cisco-approved Compaq server configuration or a Cisco-approved IBM server configuration.

Caution The installation will not complete if you do not follow the exact configuration.

Access the correct Cisco-approved server configuration for IBM server or Compaq server at

http://www.cisco.com/go/swonly

For system hardware component information and system requirements, refer to *Installing Cisco CallManager Release 3.1*.

IBM xSeries 340 and 330 Server Recommendations

Cisco recommends that if you are deploying an xSeries 340 server with a 20/40 GB DDS/4 4-mm tape drive (marketing part number for tape drive 00N7991), update your tape drive firmware to the latest version 8.160 with a release date of 2/19/01. This upgrade improves the performance of your tape drive.

Cisco recommends that if you are deploying the IBM xSeries 330 or 340 servers, update your Advanced Systems Management Processor (ASMP) firmware if necessary.

For the xSeries 340, the ASMP firmware load should be v1.15 dated 4/16/2001, and for the xSeries 330, the ASMP firmware load should be v1.04 dated 4/9/2001. The firmware upgrade ensures UM Services compatibility.

Access the correct server configuration and firmware location for IBM server or Compaq server at

http://www.cisco.com/go/swonly

Determining the Software Version

To determine the software version of Cisco CallManager 3.1, open Cisco CallManager Administration; then, click **Details** on the main Cisco CallManager Administration page. The following information displays:

- Cisco CallManager System version
- Cisco CallManager Administration version
- Database information and database DLL versions

Compatibility Matrix

Table 1 lists minimum versions with which Cisco CallManager Release 3.1(1) has been tested. Previous versions of Cisco CallManager will not work with the versions listed below.

Table 1 Compatibility Matrix for Release 3.1(1)

Component/Application	Version Tested for 3.1(1)
Cisco IP Phone Productivity Services	1.1(1)
Cisco Unity	2.4(6.135)
Cisco Unity TSP	3.0(0.7)
Cisco CallManager Extended Services (including Cisco IP AutoAttendant and Extension Mobility)	2.2(1)
Cisco Customer Response Application (including Cisco IP IVR, IPICD, AutoAttendant, and Extension Mobility)	2.2(1)
Cisco Conference Connection	1.1(1)
Cisco IP SoftPhone	1.2(1)
Cisco Personal Assistant	1.2
IPCC/ICM	Known incompatibilities with this release exist. Do **not** upgrade to Cisco CallManager Release 3.1(1).
Cisco Administrative Reporting Tool (ART)	1.1(1)
Cisco Bulk Administration Tool (BAT)	4.2(1)

Table 1 Compatibility Matrix for Release 3.1(1)

Component/Application	Version Tested for 3.1(1)
Cisco DPA 7610 Voice Mail Gateway	1.2(1)
Cisco DPA 7630 Voice Mail Gateway	1.2(1)
CAT OS: Cisco Catalyst 6000	6.1(1b), 6.2(1), and 6.2(2)
IOS: Cisco VG200	12.2(1)XN
IOS: Cisco Catalyst 4224 Voice Gateway Switch	12.1(5)YE
IOS: 2600/3600	12.2(2t) and 12.2(1)XN
Firmware: Cisco IP Phone 7960	P00303010030
Firmware: Cisco IP Phone 7940	P00303010030
Firmware: Cisco IP Phone 7910	P00403010030
Cisco IP Conference Station 7935	P005S301
Cisco Access Digital Trunk Gateway DT-24+	D00303010013
Cisco Access Digital Trunk Gateway DT-24+	D00303010013
Cisco Access Digital Trunk Gateway DE-30+	D00303010013
Analog Access - AT	A001B031
Analog Access - AS	A001B031
Analog Access WS-X6624	A00203010010
Digital Access WS-X6608	D00403010017
Conference Bridge WS-X6608	C00103010004
Media Termination Point WS-X6608	M00103010004
Java Telephony Application Programming Interface (JTAPI)	1.2(1)
Telephony Application Programming Interface (TAPI)	3.1(0.22)

Related Documentation

The following list contain related documents for Cisco CallManager Release 3.1.

- *Cisco CallManager Document Locator for Release 3.1(1)*
- *Quick Start Guide for Cisco CallManager Release 3.1*
- *Installing Cisco CallManager Release 3.1*
- *Rack-Mount Conversion Kit Installation*
- *Upgrading Cisco CallManager Release 3.1*
- *Backing Up and Restoring Cisco CallManager Release 3.1*
- *Cisco CallManager Administration Guide*
- *Cisco CallManager System Guide*
- *Cisco IP Phone Administration Guide for Cisco CallManager*
- *Serviceability Administration Guide*
- *Personal Directory Configuration Guide*
- *Cisco WebAttendant User Guide, Release 3.1*
- *Cisco CallManager 3.1 JTAPI Developer's Guide*
- *Cisco CallManager 3.1 TAPI Developer's Guide*
- *Cisco CallManager 3.1 Extension Mobility API Developer's Guide*
- *System Error Message*
- *Software License Agreement*

New and Changed Information

The following sections contain new and changed software features for Cisco CallManager Release 3.1.

New Software Features in Release 3.1(1)

Survivable Remote Site Telephony (SRST) Support

Survivable Remote Site Telephony, a service enabled within IOS, contributes substantially to the ability of Cisco's Enterprise IP Telephony System (EIPTS) to support a single, centralized call-processing model for multiple distributed sites.

The service allows remote site IP telephony components to continue to provide service when the WAN connection between Cisco CallManagers at a central site fails. Therefore, service at a remote site survives the broken connection to the central site Cisco CallManager.

Note While the primary enabler of the service is IOS, modifications to Cisco IP phone firmware enhance the functionality. This firmware ships as IP phone firmware with CallManager Release 3.1. Cisco IP Phones 7910, 7940, and 7960 provide for SRST support.

Shared Resource Management Enhancements

Cisco CallManager Release 3.1 includes four enhancements for shared resource utilization in a cluster. These enhancements contribute substantially to the ability to support a single, centralized call-processing model for multiple distributed sites.

Note A shared resource designates a device or application that is shared among multiple users. In the context of this feature, shared resources include conference bridge resources (hardware or software), transcoder resources, and music on hold resources.

Single Shared Resource per Cluster—A single shared resource may now share its services among all Cisco CallManager servers in the cluster.

In previous versions of Cisco CallManager, a single shared resource could not share its service among multiple Cisco CallManager nodes in a cluster. For example, for any cluster requiring conference bridge service, a conference bridge resource had to be configured for each Cisco CallManager in the cluster. This constraint no longer exists.

Improved Transcoder Device Efficiency—This enhancement applies to networks configured with software-only voice applications [messaging IP-interactive voice response, (IP-IVR)] that are configured to transmit and receive G.711 audio.

The Cisco CallManager using a low-bandwidth codec such as G.729 usually sets up calls placed across a low-bandwidth WAN. A call across a WAN from a Cisco IP phone to the software-only voice application therefore required a transcoder resource to transcode G.729 to G.711. Cisco CallManager 3.0 would introduce the transcoder into the Real-Time Protocol (RTP) stream as part of the call setup. However, the system was constrained in that a transcoder was inserted in the call even if the two endpoints of the call both had G.729 capabilities. This inefficient use of transcoder resources by Cisco CallManager no longer exists in Release 3.1(1). Cisco CallManager only inserts a transcoder into the call if either endpoint device does not contain a low-bandwidth codec.

Topological Association of Resources—This enhancement allows endpoint devices (Cisco IP phones, gateways, Cisco SoftPhones) to associate with locally positioned, shared-resource devices. Therefore, when a phone initiates an adhoc conference call, it can use the resources of its associated, locally positioned bridge device.

Note Topological association applies to transcoders, conference bridges, and music on hold devices. Media Resource Groups and Media Resource Lists as new Cisco CallManager constructs allow for load balancing and redundancy for these devices.

In previous versions of Cisco CallManager, the system was unaware of the location of shared resources with respect to phones and gateways in a cluster. In a centralized call-processing environment, all transcoders and conference bridges had to be centrally located. Therefore, a conference bridge with remote site participants would require Real-Time Protocol (RTP) streaming from remote site to the central conference bridge resource, consuming costly WAN bandwidth.

Small-Site, Affordable DSP Resource Module — Prior to Cisco CallManager 3.1, Digital Signal Processor (DSP) services module products that were available for transcoding and mixed codec conference bridge service were limited to the Catalyst 6000 Voice Services Module. With the advent of the three shared resource enhancements described in this section, placement of DSP services cards at remote sites became practical. However, because the Catalyst 6000 Voice Services Card was designed and positioned for large sites, it does not produce an affordable option for these services. New, smaller scale, standalone DSP services modules will be made available shortly after Cisco CallManager 3.1 releases to more affordably provide transcoder and conference bridge services at small- to medium-sized remote sites.

Note Cisco CallManager 3.1 provides administrative support for one DSP services card - a network module form factor, which will enable the service in the VG200. Additional DSP services cards will follow for the Catalyst 4000 AGM and the Catalyst 4224. 2600 and 3600 network module devices with this capability will follow introduction of the VG200 network module capability.

Music on Hold Multicast and Unicast Streaming Service

Music on Hold, an application that may be installed to an MCS server, streams Real-Time-Protocol (RTP) audio in either unicast or multicast streams from the application server to the endpoint device.

Administrators can stream music-on-hold audio to all IP phones, all Cisco VoIP gateways, and Cisco IP SoftPhone. Endpoint devices that support receiving multicast for music on hold include Cisco IP Phone models 7910, 7940, 7960, Cisco Catalyst 4000 Access Gateway Module (AGM) gateways, Catalyst 4224 gateways, and VG200 gateways.

A dedicated MCS server can stream as many as 250 music-on-hold streams (unicast or multicast). Any server can stream from up to fifty separate logical sources, each with its own continuously looping source .wav file. A fifty-first source - a sound card - may provide a real-time streaming source. Audio codec formats for any stream include G.711, G.729A, and high-fidelity audio (see description for this new enhancement). A translation utility included with the application allows translation from common formats such as .mp3 to the supported audio codecs.

Note The MCS servers do not ship with sound cards. If you choose to use a sound card, you will have to purchase it separately. Cisco has tested the Sound Blaster PCI 16 sound card and recommends it for use with the MCS 7835 and MCS 7835-1000. The MCS7825-800 requires a PCI 2.2 card; therefore, no recommended or supported sound card exists for this server model.

Issue with Music on Hold Using Locations-Based Call Admission Control

If you use locations-based call admission control, users at remote sites cannot (i.e, across a WAN link) use Music on Hold. Remote site users cannot use this feature because bandwidth calculations across locations boundries do not take into account Music on Hold streams. In place of Music on Hold, these users receive Tone on Hold, and bandwidth calculations will be correct.

Note When using Locations, only users in “Location 0” (i.e., at the hub) can receive Music on Hold streams. All others only receive Tone on Hold.

Issues with Conference Bridge Using Locations-Based Call Admission Control

Because the conference bridge resources cannot be configured with a location value, it always assumes it exists in the hub. This means that if a conference bridge is configured at a remote site (i.e., not at the hub), bandwidth would be removed from the available bandwidth, even though no bandwidth is in use across the link.

Consequently, calls may be rejected across the WAN link due to insufficient bandwidth when, in reality, enough bandwidth exists. Therefore, do not install a conference bridge at a remote site unless you can artificially adjust the bandwidth on the locations link to make more bandwidth available.

High Fidelity Audio Support

Any Cisco IP Phone 7910, 7940, and 7960 can transmit and receive audio that is sampled at 16 kHz and at a resolution of 16 bits per sample. The improved fidelity for calls through the phone handset exceeds 5-kHz audio range to the ear. This range provides a substantial improvement over a traditional 3.1 kHz audio during calls over TDM-based PBXs. As with the G.711 and G.729 codecs supported by these same phones, Cisco CallManager automatically controls the high-fidelity audio codec.

Extension Mobility Support

Extension mobility allows any user to log in to any Cisco IP Phone 7940 or Cisco IP Phone 7960. Once logged in, the user default profile, including class-of-service restrictions, primary directory number, speed dials, and productivity services apply to the phone. The user may log out manually or allow the system to log them out after timer expiration. The extension mobility application ships separately from Cisco CallManager software.

T1-CAS Support in Selected VoIP Gateways

Selected gateway interfaces including VG200, Catalyst WS-6808-T1, Catalyst 4000 AGM, Catalyst 4224, and DT-24+ gateways support T1 channel associated signaling. T1-CAS support includes E&M (ear and mouth) types 1, 2, 4, and 5.

T1/E1 Primary Rate Interface (PRI) Support for Selected VoIP Gateways

Cisco now offers T1 and E1 PRI protocols support for VG200, Catalyst 4000 AGM, and Catalyst 4224 gateways.

Call Preservation for Selected VoIP Gateways (MGCP)

Calls from any endpoint to a gateway controlled by either H.323 or Skinny Gateway Control Protocol gateways may fail when the controlling Cisco CallManager service disrupts during the call. Using Media Gateway Control Protocol (MGCP) as a substitute for H.323 or skinny gateways allows for more efficient call preservation.

In earlier versions of Cisco CallManager, MGCP support was provided for Cisco VG200 (FXS, FXO), 2600 (FXS, FXO), and 3620, 3640 and 3660-series (FXS, FXO) gateways. Two completed projects coincide with the release of Cisco CallManager 3.1 to extend MGCP support to a wider range of gateway/

time division multiplexing (TDM) interface combinations. Cisco CallManager administrative interfaces include additional MGCP configuration support for the following gateways (TDM protocols supported appear in parentheses):

- IAD2400 (FXS, FXO)
- VG200 (E1 PRI, T1 PRI, T1-CAS)
- Catalyst 6000 WS-6608-T1/E1 (T1 PRI, E1 PRI, T1-CAS)
- Catalyst 6000 WS-6624-FXS (FXS)
- Catalyst 4000 AGM (FXS, FXO, T1 PRI, E1 PRI, T1-CAS)
- Catalyst 4224 (FXS, FXO, T1 PRI, E1 PRI, T1-CAS)
- Cisco Access Digital Trunk Gateway DT-24+ (T1 PRI, T1-CAS)
- Cisco Access Digital Trunk Gateway DE-30+ (E1 PRI)

On-Hook Dialing

In Cisco CallManager 3.1, users may dial the phone without picking up the handset. The digit string initiates after the user goes offhook through the speakerphone, headset, or handset. This behavior adds to the existing dial behavior where the user dials the phone after going offhook.

Support for Centralized Voice Messaging Application with Multiple Cisco CallManager Clusters

Three enhancements to Cisco CallManager 3.1 signaling provide the support for an interface between a single voice-messaging application to multiple Cisco CallManager 3.1 clusters.

Redirected Dialed Number Identification Service (RDNIS)—Collectively, RDNIS support displays the last redirected number as well as the originally dialed number to and from configured devices and applications. While specifically designed and tested for support of voice-messaging applications, support for other applications and devices is available subject to testing and certification. Consult design guidelines for details of support.

Call Forward Number Expansion—On-net enterprise dial plans typically comprise four- or five-digit plans. Within a single site, the likelihood of digit overlap -- where two directory numbers within the same dial plan are identical -- stays minimized or eliminated because the direct inward dial (DID) number range provided by the PSTN service provider is likely from the same office and therefore nonoverlapping. However, the multisite capabilities offered by centralized call processing result in the possibility of overlap. While dial plan enhancements in Cisco CallManager 3.0 allowed operation with overlapping dial plans, delivery of dialed number information to voice-messaging systems did not accommodate overlap. Cisco CallManager 3.1 provides this overlapping dial plan support.

Call Forward Reason Code Delivery—Most voice-messaging systems can deliver prerecorded audio messages to callers. For example, mailbox owners can customize Cisco Unity mailboxes for calls forwarded on no answer, busy, or immediately on any condition. Cisco CallManager 3.1 now provides the reason code for call forwarding the voice messaging system for each call.

Serviceability Enhancements

A number of enhancements to Cisco CallManager and device instrumentation, real-time data collection and monitoring tools exist to monitor the health of the Cisco CallManager and related devices and applications. Broader instrumentation of Cisco CallManager, IP phones, gateways provides extended real-time information and alarms. Real-time Information Server (RIS), a Cisco CallManager real-time information collector is installed to Cisco CallManager nodes to collect real-time information from CCM about associated devices.

Administrator can configure the Alarm interface to deliver system events through a number of interfaces, including Event Viewer, Syslog, SDI Trace, and SDL trace, and the Alarm definitions provide system events detailed description.

An Admin Serviceability Tool (AST) comes bundled with the Cisco CallManager (but not as a standalone tool). Administrator can use this tool to monitor the health of the CCM cluster in two ways: by monitoring the vital performance counters on each CCM node in the cluster and by monitoring Device registration status and to which CCM node they are registered. The tool also provides configurable alert functionality to alert the administrator of significant events.

Cisco WebAttendant Enhancements

Cisco WebAttendant and the underlying Telephony Call Distributor have three enhancements. Longest idle hunt group logic adds to the existing linear hunt group logic. When longest idle logic is selected for a hunt group, a call sent to the hunt group goes to the available hunt group member that has been idle for the longest period. When a call passes to a Cisco WebAttendant client and the user interface is not visually focused on the display, the display will automatically “pop-to-top,” regaining primary visual focus on the display. Finally, a keyboard shortcut enhancement adds function keys 'F1' through 'F4' to select among the four main windows on the Cisco WebAttendant client user interface.

Cisco CallManager Redundancy for TAPI/JTAPI Applications

In previous versions of Cisco CallManager, a TAPI or JTAPI application registered to the TAPI or JTAPI service provider on a specific Cisco CallManager node. If that Cisco CallManager failed, the application lost service and could not automatically seek an alternate TAPI or JTAPI service provider.

Cisco CallManager 3.1 provides any application with the ability to automatically restore service in the event that a serving Cisco CallManager fails. A new service, the computer telephony interface (CTI) manager, is installed and runs normally on all Cisco CallManager nodes in a cluster. An application registers to a single CTI Manager for TAPI or JTAPI service. The application identifies alternate CTI Managers for CTI service. Each CTI Manager can receive call-processing services from a primary and an alternate Cisco CallManager server. Make sure applications are written to take advantage of this redundancy capability.

Overlap Sending Support through PRI ISDN

Prior versions of Cisco CallManager supported enbloc sending and receiving as well as overlap receiving through gateways with PRI ISDN protocols configured on the TDM trunk interface. Enbloc signaling sends the entire digit string to the receiving network. In the case of some PSTN switches, the switch can accept and act upon digit strings delivered in a block. However, a small percentage of PSTN switches can accept the alternative to enbloc signaling, which is overlap sending.

Cisco CallManager 3.1 allows the administrator to configure any PRI ISDN trunk interface on a Cisco VG200, Cisco Catalyst 4000 AGM, Cisco Catalyst 4224, Cisco Digital Trunk Access Gateway DE-30+, or Cisco Catalyst 6608-E1 gateway configured with Euro-PRI ISDN protocol to send digits to the connected PSTN switch digit-by-digit. This overlap sending allows the attached switch to accept digits one at a time and to tell the source network (Cisco CallManager through the trunk gateway) to stop sending digits after the switch has determined that it has enough to match its dial plan. This capability, specifically developed to address the German market, may be deployed in any market where overlap sending is required.

Cisco Mobile NETwork (MNET) Integration

Cisco MNET solution comprises a composite mobile network that normally installed within an enterprise. Its initial implementation targets markets in which global system for mobile communication (GSM) cellular networks are popular. It allows an enterprise to provide extended cellular service to users when those users are located within buildings where signal strength from the public mobile cellular networks is not available due to low-signal strength. GSM cellular phones use a codec specifically designed for low-bandwidth, lower power consumption. There are two modes of this codec: GSM-FR and GSM-EFR. The FR codec provides the default codec used normally if the higher fidelity EFR codec either is not available at the other endpoint or if the codec negotiation falls back to GSM-FR. Prior to Cisco CallManager 3.1, IP phones has no access to GSM codecs, neither directly nor through transcoders. Therefore, whenever an enterprise GSM phone user wanted to talk to an IP phone when inside the enterprise buildings, the call had to be hairpinned through an AS5300 gateway for transcoding. While this allowed a connection, its implementation was costly, and the call was subject to low quality due to multiple transcoding legs. Cisco CallManager 3.1 implements GSM-FR and -EFR codecs in the Catalyst 6608-T1 and E1 Digital Signal Processor (DSP) services cards. All eight segments on these cards can be configured as transcoder resources conference bridges or trunk gateways. A call from a GSM phone may be made through a transcoder or conference bridge directly to an IP phone or directly to the WS-6608 segment configured as a trunk gateway.
Cisco CallManager 3.1 administration interface supports plans for implementation in a low-cost standalone gateway at a later release.

Headset-Only Auto-Answer

You can enable or disable this feature on any line appearance of any 79XX series Cisco IP phone. When feature is enabled and a call is placed to an idle phone on which the DN on the configured line appearance has autoanswer enabled, the call automatically gets answered, and the call audio is directed to the headset audio path. If the phone is offhook while a call is made to the configured line, a visual alert lets the user know that another call is being presented to the phone. In this case, an audible alert does not occur.

Personal Directory

Personal Directory provides a personal address book stored in the Cisco CallManager LDAP directory, a Cisco IP phone synchronizer, and two Cisco IP phone services: Personal Address Book and Personal Fast Dials.

The Cisco IP Phone Address Book Synchronizer allows you to synchronize your Microsoft Outlook and/or your Outlook Express address book entries with the directory in Cisco CallManager. From a Cisco IP Phone model 7960 or 7940, you can use the Personal Address Book service to look up entries, make a selection, and press a softkey to dial the selected number.

With the Personal Fast Dials service, you can assign index numbers (from 1 to 99) for quick dialing from your Cisco IP phone. You can assign index numbers either to Personal Address Book entries or to directory entries that you add that do not correspond to the address book. You can assign and remove the Personal Fast Dials entries from your phone or the Cisco IP Phone User Options application.

CiscoWorks2000 Remote Syslog Analyzer Collector (RSAC) Component Not Included

Cisco no longer packages the CiscoWorks2000 Remote Syslog Analyzer Collector (RSAC) component with Cisco CallManager; however, the component will remain on upgrades.

With Release 3.1, you only configure the RME server in the CCM Alarm Config page. This will send ccm Syslog events directly to CiscoWorks2000 RME server, and the RSACcomponent is not needed.

For additional information, refer to the System Log Management Chapter in the Cisco CallManager Serviceability Administration Guide or CiscoWorks2000 online documentation a

http://www.cisco.com/univercd/cc/td/doc/product/rtrmgmt/cw2000/index.htm

Important Notes

Ringer Defaults to Chirp 1

When upgrading from Release 3.0(x) to Release 3.1(1), the ringer on the phone resets to Chirp 1.

You must manually reset the ringer on the Cisco IP phone to the desired state.

Resolved Caveats

Resolved Caveats - Release 3.1(1)

Table 2 lists and describes Caveats that were resolved in Cisco CallManager Release 3.1(1).

Note If you have an account with Cisco.com (Cisco Connection Online), you can use the Bug Toolkit to find caveats of any severity for any release.

To access the Bug Toolkit, log on to http://www.cisco.com/support/bugtools

Table 2 Resolved Caveats for Cisco CallManager Release 3.1(1)

DDTS Number	Summary
CSCdr36331	Going from low bit rate to low bit rate, the wrong counter is sent from a transcoder.
CSCdr36406	A <CmdArg>[Object Error]<noCmdArg> error message returns when a user is added via the Cisco CallManager administration page.
CSCdr39403	Database notification does not work in certain situations.
CSCdr43111	Call only forwards once if route point does not have forward on no answer.
CSCdr48076	Cisco IP Phone 7960 transfer button causes malfunction of Resume button.

Table 2 Resolved Caveats for Cisco CallManager Release 3.1(1)

DDTS Number	Summary
CSCdr57791	Users cannot join MeetMe conference if one number is used instead of a range.
CSCdr81135	Devices cannot be associated to user using Netscape; Javascript error displays. Using Netscape 4.7 or 4.73 on different PCs, go to Global Directory, choose user, press Associate Devices, and go to User Device Assignment screen. When you check a device box, at the bottom of the browser, it displays "Javascript error: Type 'Javascript:' into Location for details." When you press update, the device does not get associated with the user.
CSCds20133	Call Forward Back fails when Cisco CallManager attempts to use unavailable Catalyst 4000 Access Gateway Module Call Forward Back resource.
CSCds50413	Missing Disconnected, Established & TransferEnd in conference transfer scenario.
CSCds63434	CALL INIT: User cannot successfully place international calls when using Cisco Softphone.
CSCds67113	Cisco IP phone does not forward Network File System (NFS) datagrams for packet sizes larger than 11832 bytes.
CSCds67777	Blind transfer cannot complete successfully.
CSCds75317	A lineRedirect() call back to calling party causes problems.
CSCds79206	Request exists for capability to hardcode speed and duplex on Cisco IP Phone 7960.
CSCds80198	Call pickup does not pick up a call even when line is Call Forward All.
CSCdt08062	Simple Network Management Protocol (SNMP) incorrectly reports registered phone and gateway counts.
CSCdt11416	Conference master can transfer the ability to add participants.
CSCdt17567	Conference master drops from conference while adding 24th party into the conference.
CSCdt27464	Documents for Call Forward Answer do not mention special case of CSS set to <none>.
CSCdt45861	Failed SetupTransfer request incorrectly modifies existing calls.
CSCdt52841	Cisco WebAttendant documentation calls for **read access** to the wausers share, but opening the database requires **change access**.
CSCdt53938	Upgrade does not put BINs in nondefault TFTPPATH.

Table 2 Resolved Caveats for Cisco CallManager Release 3.1(1)

DDTS Number	Summary
CSCdt54310	Call Detail Records (CDRs) are written incorrectly when digit manipulation is performed on Route List.
CSCdt57230	Cisco CallManager upgrade adds extra ProcessNode for the same Cisco CallManager.
CSCdt62091	Directory number entries remain in database.
CSCdt62292	Call from PSTN to an undefined directory number records 0 in CDR.
CSCdt64163	User cannot delete partition from Cisco CallManager.
CSCdt67779	If more than 100 Call Detail Records (CDRs) are written per the defined interval, the process stops after the first 100 and waits for the next timed interval.
CSCdt94640	Cisco CallManager Release 3.0(8) does not permit the configuration of any ports of slot 6 of a 3660 router configured for Media Gateway Control Protocol (MGCP) if ports of another slot have been configured.
CSCdt95739	Caller ID overwrites a call-parked number.
CSCdu01246	Dial tone persists even after key is entered on Cisco IP SoftPhone.
CSCdu10456	Transfer succeeds after first party drops.
CSCdu23382	Route filters do not display in exported route plan reports.
CSCdu38680	User must manually turn off the server instead of pressing the key as instruction state.
CSCdu52215	User hears no distinctive ring on calls that route through H.323 gateway.
CSCdu52785	User hears no distinctive ring on calls that route through MGCP gateway.
CSCdu57794	If you have region names containing more than 32 characters, upgrading to Cisco CallManager 3.0(10) fails.
CSCdu63638	Memory leak occurs with a consult call when a blind transfer is invoked. When a consult call during a blind transfer is invoked, the consult call handle is not removed, causing a memory leak. A memory leak of around 10 bytes preconsult call initiated during a blind transfer occurs.
CSCdu71408	AS-8 gateway drops the connection if it is on user or network hold.
CMTerminals -- Firmware	
CSCdu44028	Cisco IP Phone 7960 causes slow response when connected to a 10MB NIC card. This problem only occurs when the PC port is running at a different speed from the SW port.

Open Caveats

Open Caveats for Release 3.1(1)

Table 3 describes possible unexpected behaviors by Cisco CallManager Release 3.1(1). Unless otherwise noted, these caveats apply to all Cisco CallManager 3.1 releases up to and including Cisco CallManager Release 3.1(1).

Note If you have an account with Cisco.com (Cisco Connection Online), you can use the Bug Toolkit to find caveats of any severity for any release.

To access the Bug Toolkit, log on to http://www.cisco.com/support/bugtools.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCds67777	Blind transfer cannot complete successfully. **Workaround**: None exists.
CSCds75317	Redirecting a call back to calling party causes problems. **Workaround**: Do not redirect the call back to the caller.
CSCdt26108	User cannot open two lines on one computer telephony integration (CTI) port. The TSP does not support the Cisco CallManager configuration of two more lines per CTI port device. **Workaround**: Configure the Cisco CallManager to have only one line per CTI port device.
CSCdt57230	Cisco CallManager upgrade adds extra process node for the same Cisco CallManager. If Cisco CallManager is installed on Windows 2000 as a workgroup member and Cisco CallManager joins a domain and if upgraded to a new version, the installation adds a new server instance to servers list. **Workaround**: After adding the machine to a domain, the customer must change the name of the server in the Cisco CallManager Admin windows.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdt66284	Simple Network Management Protocol (SNMP) leaks. If you are experiencing this problem, Cisco recommends that you apply the Microsoft hot fix. **Workaround**: As a workaround, have the Print Spooler service running at all times or prevent querying Print-related fields in the LanManagerMIB. You can also restart SNMP service when the memory usage exceeds a certain limit(>50 MB). **Step 1** To start the Print Spooler, choose **Start > Settings > Control Panel > Administrative Tools > Services**. **Step 2** Choose the **Print Spooler** service from the list. **Step 3** Right-click and press **Start**. **Step 4** Restart the **SNMP** service.
CSCdt86546	Redirect to busy route point succeeds, but the caller receives a busy tone. **Workaround**: None exists.
CSCdt95739	Caller ID overwrites call parked number. In a site where incoming number of calls is high, if a call is placed on park and an incoming call comes in immediately, the call park number gets overwritten, and the user cannot see the number. **Workaround**: None exists.
CSCdu06412	Incorrect interpretation of the disconnect cause occurs. A route list consists of two route group. One route group uses local gateway (local call), and the other route group uses remote gateway (long distance call). One route group that uses local gateway fails due to User Busy on the called side. **Workaround**: Use the route group that chooses local gateway.
CSCdu13391	Mobile Office Network GSM phone cannot call through 2600 gateway to POTS. This occurs when configuring the phone as a gateway insteadof an H.323 client. **Workaround**: Configure Mobile Office Network GSM phone as H.323 client.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu14186	Transferred outbound calls appear as inbound in call detail records. When a call to the PSTN is made from an IP phone (7960 in this case) it is recorded as outbound. If the call is then transferred to another IP phone the second portion of the phone call is recorded in CDR as an inbound call originating from the PSTN. **Workaround**: None exists.
CSCdu14950	Out call voice-mail ports are locking up after a few hours of reset. **Workaround**: Reset gateway.
CSCdu22012	Call forward from Cisco CallManager Admin allows invalid number entry. This works as designed. No way exists to validate destinations from CCMAdmin. Phones validate destinations using the runtime configuration, and even those destination can become invalid if the routing or network configuration changes. **Workaround**: None exists.
CSCdu31271	User cannot add G.729 call to a software conference bridge. **Workaround**: None exists. This works as designed. Software conference bridge only supports G.711.
CSCdu32362	CPU running at 100 percent with svchost.exe consuming most of the CPU time. This occurred under a heavy call load. Application was unable to accept calls within 4 seconds. **Workaround**: None exists.
CSCdu34488	User cannot complete call over H.323 gateway when in 729 region. **Workaround**: None exists.
CSCdu35665	Cisco CallManager information in CTI manager for applications-controlled IP phones is wrong. **Workaround**: When an application controls IP phones, if Cisco CallManagers are added or deleted from the Cisco CallManager Group, or Device Pool configuration is modified, restart the application.
CSCdu38120	Counter does not appear to function properly during a conference call. **Workaround**: None exists.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu38419	H.245 packet precedence is not getting set properly. **Workaround**: None exists.
CSCdu43682	30VIP second line does not go offhook correctly when VM/SD is pressed. **Workaround**: Go on hook and press line button for second line to get dial tone.
CSCdu44827	Equipment possibly sending incorrect restart code across 6608 port. This behavior seems to happen after the datalink has been lost between the gateway and the PSTN. **Workaround**: Resetting the gateway clears up the problem if it is observed in the Cisco CallManager traces.
CSCdu47183	The delay before dialing timer for groundstart CAS is always set to 80-90 ms. **Workaround**: No need exists to set this timer. It's always set to use 80-90 ms in the hardware to meet the EIA464 requirement.
CSCdu47196	A call through a PSTN on Cisco CallManager 3.0(X) cluster will be dropped if Cisco IP phone on Cisco CallManager 3.1(1) puts the call on hold. The called dropped because of interoperability between Cisco CallManager 3.1(1) and Cisco CallManager 3.0(X) when Music on Hold occurs between clusters. The held call drops immediately when MOH is attached. Cisco CallManager drops the held call is dropped due to StartMediaTransmission with zero IP address and port number to the gateway when one way streaming is established by MOH. **Workaround**: Enable MTP check box in the H.323 intercluster trunk configuration page at cluster running Cisco CallManager 3.1(1).
CSCdu47838	Cisco CallManager does not apply the calling party transform mask when a call is forwarded out to a gateway, and the gateway Calling Party Selection field is set to last or first redirected number. Hence, the calling party number presented to the PSTN is the local IP Phone extension number. **Workaround**: Use Caller ID Dn field to force the outgoing CLI mask on the gateway. However, this workaround does not work in case of multitenancy.
CSCdu48051	Cisco CallManager sends RST for TCP SYN when queue is full. **Workaround**: None exists.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu48440	Cisco CallManager stops when memory management initializes large number of regions. This Cisco CallManager worked with 14 regions; then, 55 "dummy" regions were added for testing purposes. After this was done, the Cisco CallManager service repeatedly stopped and then restarted every couple of minutes. The "Regions Initialization Time" Cisco CallManager service parameter was set at the default value of 30. **Workaround**: If you set the **Regions Initialization Time** Cisco CallManager service parameter to a value of 90, you can support at least 40 regions. However, 50-70 regions may require a larger timeout value, or you may need to limit the number of regions to the 40-50 range.
CSCdu49686	Call forward no answer happens even after call is redirected. Call forward no answer is active even after a active call is redirected. This causes the call, which is not answered in the redirected location, to be forwarded to the call forward no answer number. **Workaround**: None exists.
CSCdu53300	The device pool and Cisco CallManager group information in CTI Manager is inconsistent. If device pool, Cisco CallManager group, or Cisco CallManagers in a Cisco CallManager group are modified when no CTI application is logged on, CTI Manager will not have the recently modified Device Pool and Call Manager Group information. **Workaround**: When the CTI application is logged on, reset the devices that are affected by the change in device pool, User hears no distinctive ring on calls that route through H.323 gateway or Cisco CallManagers.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu54196	Simple Network Management Protocol (SNMP): You cannot set cdpInterfaceEnable to true or false. **Workaround**: Alternatively, Enable/disable cdp via the Win2k Device Manager as follows: **Step 1** In Windows 2000 Control Panel Menu, double click "System" and select "Hardware"; then, select "Device Manager." **Step 2** Go to "View" and choose both "Devices by connection" and "Show hidden devices." **Step 3** Double click "Cisco Discovery Protocol"; then, choose "Driver" tab. **Step 4** Choose "stop" to disable the cdp or "start" to enable the cdp. Make sure CDP is enabled at all time for CiscoWorks2000 to discover the CCM server.
CSCdu56651	Call from Mobile Office Network phone through Cisco Access Digital Trunk Gateway DT24+ fails to establish. **Workaround**: None exists.
CSCdu58207	CTI Manager CPU reaches 100 percent during second failover and failback. This happens in a cluster environment with two Cisco CallManagers. One of the Cisco CallManager is a publisher and the other Cisco CallManageris a subscriber. The condition can be summarized with the following steps: 1. CM1 failover to CM2 2. CM2 failback to CM1 3. CM1 failover to CM2 4. CM2 failback to CM1 Both CM1 and CM2 CPU usage goes up to 100 percent after executing step 4. The perfmon shows the CTI Manager for both CM1 & CM2 goes 100 percent. **Workaround**: Shutdown and restart the Cisco CTI Manager and the Cisco CallManager Service in the subscriber Cisco CallManager. The CPU for the both CM1 and CM2 will go back to normal.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu58609	Modifying route list parameter within existing route pattern fails. **Workaround**: Restart the Cisco CallManager service.
CSCdu61559	Cisco CallManager install fails due to integrated install and/or install shield issues. A combination of factors caused this problem: • The extracted InstallShield setup files from a previous incomplete/aborted install exist in the %TEMP% folder • InstallShield Engine, C:\Program Files\Common Files\InstallShield\Engine\IKernel.exe, version mismatch **Workaround**: **Step 1** Delete all the files from the folder pointed to by the environment variable %TEMP%; typically, C:\Documents and Settings\Administrator\Local Settings\Temp **Step 2** Delete the folder C:\Program Files\Common Files\InstallShield\Engine **Step 3** Rerun the install.
CSCdu63020	Stopping the CMI service causes a call to go to open tree for voice-mail number. **Workaround**: None exists.
CSCdu64779	A 1-second delay occurs on request to the OLC with call forward all through an H.323 gateway. An issue exists in the H.245 negotiation between the IOS gateway and Cisco CallManager: the Cisco CallManager is waiting one second (after receiving the H.245 OLC request from GW) before sending the H.245 OLC request. **Workaround**: None exists.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu65117	No ringback occurs when parked call reverts to parking party. When a call is parked, the parked user will hear Music on Hold or Tone on Hold. If the parked call is not retrieved within the CallParkReversionTimeout, the phone that parked the call will begin to ring. At this point, the parked party will not hear Music or Tone on Hold any more and will not hear ringback, leading the parked party to think the call has been dropped. **Workaround**: None exists.
CSCdu67281	TSP exits at startup if the computer is on a network that cannot access the Cisco CallManager. **Workaround**: Establish a network connection to the target Cisco CallManager and restart the application that uses the TSP (e.g., Cisco IP SoftPhone).
CSCdu68134	Audio translator produces garbled G.729 audio when converting from .mp3 files. **Workaround**: Convert the MP3 file to a 44.1k 16-bit WAV file then put this new file in the C:\Cisco\DropMOHAudioSrouceFilesHere directory.
CSCdu68370	Completing a transfer call fails when it is set up manually. **Workaround**: None exists.
CSCdu73278	Exception (access violation) occurs on svchost.exe when extension mobility user logs out. **Workaround**: None exists.
CSCdu73294	TSP crashes when Cisco IP SoftPhone tries to conference in the Cisco Conference Connection. **Workaround**: None exists.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu74713	Voice path does not cut through after progress occurs with PI = 1. This problem only occurs with the following signalling: CM - Q931 Setup -> E1/T1 Cat6K -> PSTN CM <- E1/T1 <- Q931 Setp_Ack <- PSTN CM <- E1/T1 <- Q931 Progress with PI = 1 <- PSTN When receiving progress, it's not cutting through, which it should if PI=1,2 or 8. **Workaround**: Insert 'SendingComplete IE' in the outgoing Setup. If user then receives Progress, voice path is already established and user hears the INBAND information (possible ringback).
CSCdu75418	MWI does not light for new partition without restarting Cisco CallManager service. **Workaround**: Stop and start the MWI service.
CSCdu77097	The pilot number is not processing the order of the hunt group correctly in an intercluster environment. **Workaround**: Turn off/on servers or stop/start CTI and TCD services.
CSCdu77512	Cisco CallManager route filter clauses are ignored beyond 1024 character limit. **Workaround**: Stop clauses for the first route filter before the 1024 character limit. Create a second route filter that contains the rest of the clauses. Create two duplicate route patterns and assign one route filter to each route pattern.
CSCdu78298	Music on Hold ignores Locations call admission control, which leads to voice quality degredation. **Workaround**: Remote users are not allowed to receive Music on Hold. They only receive a Tone on Hold. This bandwidth restriction will be resolved in an upcoming maintenance release.
CSCdu78704	Hunt Group does not work after Cisco CallManager failover when Cisco CallManager is stopped. **Workaround**: Stop all the TCD services in the cluster, then start the service one by one. If you still have problem, stop all the TCD services, stop all the CTIManager services, start all CTIManager services, and start all TCD services.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu79855	cgpnVoiceMailBox is incorrect when skinny port goes offhookwith cgpn. **Workaround**: A partial workaround is to blank the PA port's Voice Message Box on the Directory Number Configuration page.
CSCdu81246	Speed dial configuration display text field does not reflect real space. **Workaround**: None exists.
CSCdu81800	Cisco CallManager Admin page gives 500 Internal Server Errors. **Workaround**: Go into Component Services and unload the CCMadmin component.
CSCdu81988	TFTP problems with SRST phone loads. **Workaround**: Reset only a few phones at a time. Once they are fully registered, more phones can be reset until all the phones have been successfully upgraded.
CSCdu82852	Cisco WebAttendant hunt group does not work on new installation. **Workaround**: Restart Telephony Call Dispatcher service.
CSCdu83136	CTI Manager and CCM Services cannot be started. CTI Manager and CCM Service cannot be started after second failover and failback. This happens in a cluster environment with two Cisco CallManagers. One of the Cisco CallManager is a publisher and the other Cisco CallManageris a subscriber. The condition can be summarized with the following steps: 1. CM1 failover to CM2 2. CM2 failback to CM1 3. CM1 failover to CM2 4. CM2 failback to CM1 Both CM1 and CM2 CPU usage goes up to 100 percent after executing step 4. The perfmon shows the CTI Manager for both CM1 & CM2 goes 100 percent. When the CTI Manager and the CCM Services for CM2 is restarted, the service does not come back. **Workaround**: None exists.
CSCdu84212	Modem connections on C6624 FXS are a low speed or unreliable. **Workaround**: None exists.

Table 3 Open Caveats for Cisco CallManager Release 3.1(1)

DDTS number	Description
CSCdu84227	Redirect failed while redirect a call from Cisco Access Digital Trunk Gateway DT24+. **Workaround**: None exists.
CSCdu86461	No translation when DN is unregistered (gets wrong mailbox). **Workaround**: None exists.
CSCdu86476	DisplayIE for anonymous device is always displayed as deselected. **Workaround**: None exists.
Firmware	
CSCdu67286	Ringer resets to Chirp 1 on 3.0(X) to 3.1(X) upgrade. **Workaround**: After the phone has been upgraded to CallManager 3.1, manually change the ring back to what it was before the upgrade.

Documentation Updates

The following section provides documentation changes that were unavailable when the Cisco CallManager Release 3.1(1) documentation suite was released.

Changes

Getting Started Title Changes

The *Cisco CallManager Administration Guide* and *Cisco CallManager System Guide* refer to the *Getting Started* publications provided with your phones.

Cisco IP Phone Models 7960 and 7940 User Guide replaces the *Getting Started with the Cisco IP Phone 7940/7960*. This document and the *Getting Started with the Cisco IP Phone 7910* do not ship with the phone but are available on CCO and can be ordered.

Cisco IP Phone 7900 Family Administration Guide Title Changes

The *Cisco CallManager Administration Guid*e and *Cisco CallManager System Guide* also refer to the *Cisco IP Phone 7900 Family Administration Guide*. This document has been renamed to *Cisco IP Phone Administration Guide for Cisco CallManager*.

Remote Serviceability and Troubleshooting Information Changes Book

Serviceability Administration Guide includes instructions to configure remote serviceability and to use the Cisco CallManager Trace for diagnostic traces.

Omissions

Maintaining Cisco IP Phone Services List

Using Cisco CallManager Administration, you define and maintain the list of Cisco IP Phone Services to which users can subscribe at their site. You can also create parameters for each service that require users to enter data in the Cisco IP Phone User Options application before subscribing to that service.

In the 3.1(1) release, you can mask entries in the Cisco IP Phone User Options application, so asterisks display rather than the actual user entry. You may want to do this for parameters such as passwords that you do not want others to be able to view. To mask a parameter entry, check the Parameter is a Password (mask contents) field on the Configure Cisco IP Phone Service Parameter window in CallManager Administration.

Corrections

Cisco CallManager Release 3.1(1) supports Cisco Unity Version 2.4(6.135). The Cisco CallManager System Guide incorrectly states that Cisco CallManager Release 3.1(1) requires Cisco Unity Version 3.0(1).

Obtaining Documentation

The following sections provide sources for obtaining documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at the following sites:

- http://www.cisco.com
- http://www-china.cisco.com
- http://www-europe.cisco.com

Documentation CD-ROM

Cisco documentation and additional literature are available in a CD-ROM package, which ships with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or as an annual subscription.

Ordering Documentation

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- Registered Cisco Direct Customers can order Cisco Product documentation from the Networking Products MarketPlace:

  http://www.cisco.com/cgi-bin/order/order_root.pl

- Registered Cisco.com users can order the Documentation CD-ROM through the online Subscription Store:

  http://www.cisco.com/go/subscription

- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco corporate headquarters (California, USA) at 408 526-7208 or, in North America, by calling 800 553-NETS(6387).

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San Jose, CA 95134-9883

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Technical Assistance Center

The Cisco TAC website is available to all customers who need technical assistance with a Cisco product or technology that is under warranty or covered by a maintenance contract.

Contacting TAC by Using the Cisco TAC Website

If you have a priority level 3 (P3) or priority level 4 (P4) problem, contact TAC by going to the TAC website:

http://www.cisco.com/tac

P3 and P4 level problems are defined as follows:

- P3—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- P4—You need information or assistance on Cisco product capabilities, product installation, or basic product configuration.

In each of the above cases, use the Cisco TAC website to quickly find answers to your questions.

To register for Cisco.com, go to the following website:

http://www.cisco.com/register/

If you cannot resolve your technical issue by using the TAC online resources, Cisco.com registered users can open a case online by using the TAC Case Open tool at the following website:

http://www.cisco.com/tac/caseopen

Contacting TAC by Telephone

If you have a priority level 1 (P1) or priority level 2 (P2) problem, contact TAC by telephone and immediately open a case. To obtain a directory of toll-free numbers for your country, go to the following website:

http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

P1 and P2 level problems are defined as follows:

- P1—Your production network is down, causing a critical impact to business operations if service is not restored quickly. No workaround is available.
- P2—Your production network is severely degraded, affecting significant aspects of your business operations. No workaround is available.

This document is to be used in conjunction with the documents listed in the "Related Documentation" section.