Cisco Unified IP Phone 8961, 9951, and 9971 Release Notes for Firmware Release 9.4(1)

First Published: November 05, 2013

Last Modified: March 27, 2015

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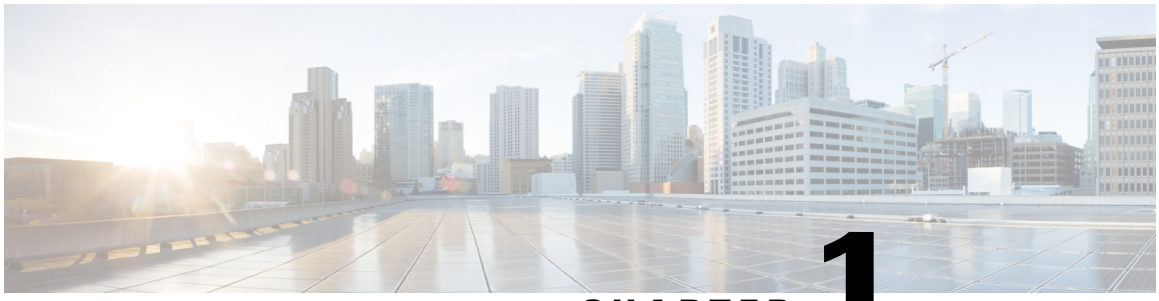

CHAPTER 1

Cisco Unified IP Phone 8961, 9951, and 9971 Release Notes for Firmware Release 9.4(1)

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Introduction

These release notes support the Cisco Unified IP Phones 8961, 9951, and 9971 running Firmware Release 9.4(1).

The following table lists the Cisco Unified Communications Manager release and protocol compatibility for the Cisco Unified IP Phones.

Cisco Unified IP Phone	Protocol	Cisco Unified Communications Manager
Cisco Unified IP Phones 8961, 9951, and 9971	SIP	Cisco Unified Communications Manager Release 7.1(5) and later

Note Some features in this release may require a specific version of Cisco Unified Communications Manager. The feature description identifies if the feature requires a different version of Cisco Unified Communications Manager from the version identified in the above table.

Related Documentation

Use the following sections to obtain related information.

Cisco Unified IP Phone 8900 Series Documentation

Refer to publications that are specific to your language, phone model, and Cisco Unified Communications Manager release. Navigate from the following documentation URL:

http://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8900-series/tsd-products-support-series-home.html

Cisco Unified IP Phone 9900 Series Documentation

Refer to publications that are specific to your language, phone model, and Cisco Unified Communications Manager release. Navigate from the following documentation URL:

http://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phones-9900-series/tsd-products-support-series-home.html

Cisco Unified Communications Manager Documentation

See the *Cisco Unified Communications Manager Documentation Guide* and other publications that are specific to your Cisco Unified Communications Manager release. Navigate from the following documentation URL:

http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/tsd-products-support-series-home.html

Cisco Business Edition 5000 Documentation

See the *Cisco Business Edition 5000 Documentation Guide* and other publications that are specific to your Cisco Business Edition 5000 release. Navigate from the following URL:

http://www.cisco.com/c/en/us/support/unified-communications/business-edition-5000/tsd-products-support-series-home.html

Cisco Virtualization Experience Client Documentation

Refer to publications that are specific to your language. Navigate from the following documentation URL:

http://www.cisco.com/c/en/us/support/collaboration-endpoints/virtualization-experience-client-6000-series/tsd-products-support-series-home.html

New and Changed Features

The following sections describe the features that are new or have changed in this release.

Note Some features may require the installation of a Cisco Unified Communications Manager Device Package. Failure to install the Device Package before the phone firmware upgrade may render the phones unusable.

Features Available with Firmware Release

The following sections describe the features available with the Firmware Release.

CGI ModeInfo

The CGI ModeInfo feature provides information that can be used for troubleshooting phone problems. Using a web browser, the phone information can be requested in XML format. Access the information using the following command:

http://<ip address of phone>/CGI/Java/ModeInfo

Example: http://xx.xx.xx.xx/CGI/Java/ModeInfo

In order to request the information, a user must be associated with the phone and web access must be enabled.

This feature has no user or administrator impact.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

CTL and ITL Status Display and Report

The CTL and ITL Status Display and Report feature enables the Cisco Unified IP Phone to send information to the Cisco Unified Communications Manager so that it is stored in the shared memory for reporting and alarm purposes.

The CTL and ITL Status Display and Report must be configured by a system administrator.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

CTL/ITL Signature

The CTL/ITL Signature feature enhances security by using secure hash algorithm (SHA1) in the CTL and ITL files. The SHA1 hash replaces the MD5 hash.

There are no impacts to the administrator or user for this feature.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

E-SRST Service Improvements

The E-SRST Service Improvements feature enables Video, Shared Line, and BLF Speed Dial on the phone when communications with the controlling Cisco Unified Communications Manager are broken. This feature builds on the existing Survivable Remote Site Telephony (SRST) feature.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Gateway Recording for SIP

The Gateway Recording for SIP feature improves call monitoring by making it possible to record calls using Cisco Voice Gateway. This allows you to record calls made on Cisco Jabber, a Cisco IP Phone (SIP), or calls made on a mobile device.

Note This feature requires Cisco Unified Communications Manager 10.0 or later.

Gateway Recording must be configured by a system administrator.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Hide Wi-Fi User Interface Setting

The Hide Wi-Fi User Interface Setting feature removes the Wireless Setup option from the Network Setup menu on the phone when Wi-Fi is disabled from the Cisco Unified Communications Manager.

The feature is supported on the following phones:

- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

RTCP Always On

The RTCP Always On feature simplifies the phone administration by removing the need to set the RTCP field. RTCP is always turned on for phones.

There is no user or administrator impact.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Unified Font Size Enhancement

The Unified Font Size Enhancement feature allows users to change the font size used for the call history, call session, phone line label, and key expansion module line label using a single menu item.

The font size affects the following labels:

- call screen (caller name and feature key label)
- caller name in Call History
- line label on phone screen
- line label on key expansion module

The following table describes the configuration setting and the corresponding font sizes.

Configuration setting	Line label	Call session: Caller name	Call session: Caller number	Call history	Key expansion module
Huge	26 point	26 point	22 point	26 point	23 point
Large (default)	23 point	23 point	22 point	23 point	18 point
Medium	20 point	20 point	22 point	19 point	18 point
Small	16 point	16 point	20 point	19 point	16 point

There is no administrator impact for this feature.

This feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Mnager 10.0 (SIP)

Features Available with Latest Cisco Unified Communications Manager Device Pack

The following sections describe features in the release which require the new firmware and the latest Cisco Unified Communications Manager Device Pack.

For information about the Cisco Unified IP Phones and the required Cisco Unified Communications Manager device packs, see the following URL:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/compat/devpack_comp_mtx.html

Confidential Access Level

The Confidential Access Level (CAL) feature controls whether a call can be completed based on the CAL configuration in the Cisco Unified Communications Manager.

Note This feature requires Cisco Unified Communications Manager 10.0 or later.

When CAL is enabled, the user sees information about the call in a CAL message. The phone displays the CAL message for the duration of the call.

If a call fails due to an incompatible CAL, the phones display a failure message. The administrator sets up the failure message that the user sees.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Configurable RTP/sRTP Port Range

The Configurable RTP/sRTP Port Range feature provides a configurable range of ports (2048 to 65535) for Real-Time Transport Protocol (RTP) and secure Real-Time Transport Protocol (sRTP). The default RTP/sRTP port range is 16384 to 32764.

Note This feature requires Cisco Unified Communications Manager 8.6.2 or later.

The administrator configures the RTP and sRTP port values in the SIP Profile.

This feature has no user impact.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to find more information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Configurable TLS Resumption Timer

TLS Session Resumption enables a TLS session to resume without repeating the entire TLS authentication process. TLS Session Resumption can significantly reduce the time taken for TLS connection to exchange data.

TLS Session Resumption Timer must be configured by a system administrator.

This feature has no user impact.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

FIPS 140-2 Level 1 Support

The Federal Information Processing Standard (FIPS) 140-2 Level 1 Support feature provides a secure, encrypted environment to conform to the United States Department of Defense Unified Capabilities Requirements (UCR) 2008 standard. Contact Cisco for further information about FIPS and product compliance.

This feature is enabled by the system administrator.

This feature is available for the following phones:

- Cisco IP Phone 8961
- Cisco IP Phone 9951
- Cisco IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0

IPv6 Support

The IPv6 Support feature provides support for expanded IP addressing on Cisco IP Phones. The phones support IPv6 in standalone or in dual-stack configurations. In dual-stack mode, the phone can communicate using IPv4 and IPv6 simultaneously, independent of the content.

Note This feature requires Cisco Unified Communications Manager 10.0 or later.

The default configuration is dual-stack mode.

The feature is controlled using the IP Addressing Mode and the IP Addressing Mode Preference for Signalling parameters in the Common Phone Profile: **Device** > **Device Settings** > **Common Phone Profile** of the Cisco Unified Communications Manager Administration window.

The IP Addressing Mode and the IP Addressing Mode Preference for Signalling parameters have the following options:

- IPv4 Only
- IPv6 Only
- IPv4 and IPv6 (default)

Note The IPv6 HTTP browser and some XSI commands require HTTP authentication. Check the Cisco Unified Communications Manager for IPv6 HTTP Authentication support.

This feature has no user impact.

This feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Line State Display Enhancement

The Line State Display Enhancement feature enables the user to see the Remote-in-Use message when there is a call alert on a shared line.

The Line State Display Enhancement must be configured by a system administrator.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0 (SIP)

Separate Audio and Video Port Range Configuration

The Separate Audio and Video Port Range Configuration feature enables customers to improve Quality of Service (QoS) by configuring audio traffic and video traffic on different ports.

The ports are configured from one of the following Cisco Unified Communications Manager windows:

- **System** > **Enterprise Phone Configuration**
- **Device** > **Device Settings** > **Common Phone Profile**
- **Device** > **Phone** > **Phone Configuration**

The existing RTP port range parameters continue to be used for audio traffic. For video traffic, two new fields in the Cisco Unified Communications Manager Administration are available:

- Start Video RTP Port
- Stop Video RTP Port

The following rules apply when configuring the fields:

- Each field must contain a number between 2048 and 65535.
- The value in the Stop Video RTP Port field must be larger than the value in the Start Video RTP Port field.
- The difference between the Start Video RTP Port field and the Stop Video RTP Port field must be at least 16.

After the Start Video RTP Port and Stop Video RTP Port are configured, the phone uses ports within the port range for video traffic. The audio traffic continues on the port range configured in the SIP Profile for the audio RTP port.

If the audio and video port ranges overlap, the ports carry both audio and video traffic. If the video port range is not configured correctly, the phone uses the configured audio ports for both audio and video traffic.

This feature has no user impact.

The feature is supported on the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0

Serviceability for SIP Endpoints

The Serviceability for SIP Endpoints feature provides a quick and easy way for administrators to capture debug information from the Cisco Unified IP Phones. With this feature, administrators can:

- Remotely run debug commands on a phone or group of phones without logging into each device separately.
- Collect the phone log without logging into each phone separately.
- Troubleshoot phones by using multiple debug categories with the **debugsh** (debug telephony) command

With this feature, administrators set the following parameters in the **Device** > **Device settings** > **Common Phone Profile**, **System** > **Enterprise Phone Configuration**, or **Device** > **Phone** window:

- Log Profile - values: Preset (default), Default, Telephony
- Remote Log - values: Disable (default), Enable
- Log Server - IP address (IPv4 or IPv6 address)

The Log Server address can include a port number.

- The format for IPv4 Log Server is: address:<port>@@base=<0-7>;pfs=<0-1>.
- The format for IPv6 Log Server is: [address]:<port>@@base=<0-7>;pfs=<0-1>.

Note When the Serviceability feature is enabled, it impacts call quality when the phone is using Wi-fi. The system administrator should decide if the feature should be enabled.

This feature is available for the following phones:

- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 10.0

Installation

Install Firmware Release on Cisco Unified Communications Manager

Before using the Cisco Unified IP Phone Firmware Release 9.4(1) with Cisco Unified Communications Manager, you must install the latest firmware on all Cisco Unified Communications Manager servers in the cluster.

Procedure

Step 1 Go to the following URL:
http://www.cisco.com/cisco/software/navigator.html?mdfid=268437892&flowid=5293

Step 2 Depending on your phone model, choose **Cisco Unified IP Phones 8900 Series** or **Cisco Unified IP Phones 9900 Series**.

Step 3 Choose your phone type.

Step 4 Choose **Session Initiation Protocol (SIP) Software**.

Step 5 In the Latest Releases folder, choose **9.4(1)**.

Step 6 Select one of the following firmware files, click the **Download** or **Add to cart** button, and follow the prompts:

- cmterm-8961.9-4-1-9.cop.sgn
- cmterm-9951.9-4-1-9.cop.sgn
- cmterm-9971.9-4-1-9.cop.sgn

Note If you added the firmware file to the cart, click the **Download Cart** link when you are ready to download the file.

Step 7 Click the + next to the firmware file name in the Download Cart section to access additional information about this file. The hyperlink for the readme file is in the Additional Information section, which contains installation instructions for the corresponding firmware:

- cmterm-8961.9-4-1-9-readme.html
- cmterm-9951.9-4-1-9-readme.html
- cmterm-9971.9-4-1-9-readme.html

Step 8 Follow the instructions in the readme file to install the firmware.

Install Firmware Zip Files

If a Cisco Unified Communications Manager is not available to load the installer program, the following .zip files are available to load the firmware.

- cmterm-8961.9-4-1-9.zip
- cmterm-9951.9-4-1-9.zip
- cmterm-9971.9-4-1-9.zip

Firmware upgrades over the WLAN interface may take longer than upgrades using a wired connection. Upgrade times over the WLAN interface may take more than an hour, depending on the quality and bandwidth of the wireless connection.

Procedure

Step 1 Go to the following URL:
http://www.cisco.com/cisco/software/navigator.html?mdfid=268437892&flowid=5293

Step 2 Depending on your phone model, choose **Cisco Unified IP Phones 8900 Series** or **Cisco Unified IP Phones 9900 Series**.

Step 3 Choose your phone type.

Step 4 Choose **Session Initiation Protocol (SIP) Software**.

Step 5 In the Latest Releases folder, choose **9.4(1)**.

Step 6 Download the relevant zip files.

Step 7 Unzip the files.

Step 8 Manually copy the unzipped files to the directory on the TFTP server. See *Cisco Unified Communications Operating System Administration Guide* for information about how to manually copy the firmware files to the server.

Cisco Unified Video Camera Firmware

The Cisco Unified Video Camera is supported on Cisco Unified Communications Manager Release 7.1(5) and later.

Limitations and Restrictions

Caveat CSCul01945 - Http Service Failed and Returned with Host Not Found

You may encounter a Host Not Found error message when using a HTTP service like Extension Mobility. This issue may impact Extension Mobility or other HTTP services and is due to an increased network delay.

This issue has been resolved in Engineering Special Release 9.4(1)ES3 and the fix will be included in the next Cisco.com release, when that release becomes available.

For additional information, see CSCul01945 on Cisco Bug Search.

Cisco Unified IP Phone 9971 and Personal Address Book or Fast Dial Features

Users of the Personal Address Book or Fast Dial feature may receive an error message followed by a request to log back into the network. In DNS round robin configurations, the phone does not retain access with the same IP address. In this situation, the phone displays a message to indicate that the user must log into the phone again.

Plantronics Audio 615M Headset and Cisco Unified IP Phone 8961

The Plantronics Audio 615M headset is not compatible with the Cisco Unified IP Phone 8961. You must use an alternate headset type for this IP phone. For more information, see https://tools.cisco.com/bugsearch/bug/CSCth71104.

Plantronics CS50 USB Headset and Cisco Unified IP Color Key Expansion Module

The Plantronics CS50 USB headset causes the phone to request power from the switch even though the headset is self powered. In this case, if a device such as a camera or expansion module is connected and active on the phone, the switch will reject the power request for the headset because the power budget has been exceeded. In this case, the headset cannot be used.

Cisco Unified IP Phones 9951 and 9971 One-Way Video Calls

Because of limitations in the H.264 video signaling standards, Cisco Unified IP Phones 9951 and 9971 may not correctly display video that is received from devices supporting resolutions greater than 640 x 480. In this case, the user sees a black video screen.

To ensure that video from such devices displays properly on the IP phone, configure high definition phones and Cisco Unified IP Phones 8961, 9951, and 9971 into different call regions and limit the video bandwidth to 384 kb/s when calling between regions.

Cisco Virtualization Experience Client 2100

The Cisco Virtualization Experience Client 2100 (VXC 2100) Series are zero clients designed to deliver a user desktop from a centralized host server, providing access to desktop applications as if they were available locally. The Cisco VXC 2100 Series attaches to the Cisco Unified IP Phones 8961, 9951, and 9971 through a spine connector cable.

When running VXC with a single-tunnel option, high traffic to or from VXC may affect the phone's performance. Cisco Unified IP Phones 8961, 9951, and 9971 support 384kbps throughput bandwidth for VXC.

Set VPN MTU to 1406 to reduce packet reassemble in Cisco Unified IP Phones 8961, 9951, and 9971, to improve bandwidth in limited packet rate.

Video and audio applications may not play smoothly through VXC, even on a device plugged directly into a LAN.

USB CD-ROMs and other USB storage devices have limited support. In the current VXC environment, the expected performance is a low bit/transfer rate for USB storage devices and USB CD-ROMs connected to the VXC device.

For more information, see http://www.cisco.com/c/en/us/support/collaboration-endpoints/virtualization-experience-client-2000-series/tsd-products-support-series-home.html.

No UDP for SIP Support

Cisco Unified IP Phones 8961, 9951, and 9971 do not support UDP for SIP. Do not configure the UDP protocol for SIP signaling on Cisco Unified IP Phones 8961, 9951, and 9971.

Multiple Text Messages

If you send multiple text messages to Cisco Unified IP Phones 8961, 9951, and 9971 within a few seconds using CiscoIPPhoneExecute.xml, the phone may reply with an HTTP/1.1 503 Service Unavailable message, and the user may not receive the text message on their phone screen.

Secure Video Bandwidth When Calling Over VPN

When using Cisco Unified IP Phones 8961, 9951, and 9971 for a secure video call over VPN and VXC VPN, the maximum supported bandwidth is 320 kbps.

Turn Off VPN Before Downgrade

The phone will go into an unregistered state if it is downgraded from a 9.3(1) load to a pre-9.2(3) load while VPN is turned on. Turn off VPN before downgrading to a pre-9.2(3) load.

Voice VLAN and IPv6 Limitation

If the PC attached to the PC port of the phone is using IPv6, we recommend that the PC Voice LAN access be disabled. This ensures that the PC can connect to the Voice VLAN.

Phone Limitation During SVI Change

Cisco IP Phones use a Switch Virtual Interface (SVI) to manage VLANs. If the SVI changes and the phones require new IP addresses, some phones require a reboot so that the new IP address is used. The following phones must be rebooted in this condition:

- Cisco Unified IP Phone 6901
- Cisco Unified IP Phone 6911
- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961
- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945
- Cisco Unified IP Phone 8961
- Cisco Unified IP Phone 9951
- Cisco Unified IP Phone 9971

Phone Behavior During Times of Network Congestion

Anything that degrades network performance can affect Cisco IP Phone voice and video quality, and in some cases, can cause a call to drop. Sources of network degradation can include, but are not limited to, the following activities:

- Administrative tasks, such as an internal port scan or security scan
- Attacks that occur on your network, such as a Denial of Service attack

To reduce or eliminate any adverse effects to the phones, schedule administrative network tasks during a time when the phones are not being used or exclude the phones from testing.

Caveats

Access Cisco Bug Search

Known problems (bugs) are graded according to severity level. These release notes contain descriptions of the following:

- All severity level 1 or 2 bugs
- Significant severity level 3 bugs

You can search for problems by using the Cisco Bug Search.

Before You Begin

To access Cisco Bug Search, you need the following items:

- Internet connection
- Web browser
- Cisco.com user ID and password

Procedure

Step 1 To access the Cisco Bug Search, go to:
https://tools.cisco.com/bugsearch

Step 2 Log in with your Cisco.com user ID and password.

Step 3 To look for information about a specific problem, enter the bug ID number in the Search for field, then press **Enter**.

Open Caveats

The following table lists severity 1, 2, and 3 defects that are open for the Cisco Unified IP Phones 8961, 9951, and 9971 for Firmware Release 9.4(1).

For more information about an individual defect, you can access the online record for the defect by clicking the Identifier or going to the URL that is shown. You must be a registered Cisco.com user to access this online information.

Because defect status continually changes, the table reflects a snapshot of the defects that were open at the time this report was compiled. For an updated view of open defects, access Bug Toolkit as described inAccess Cisco Bug Search, on page 16.

Table 1: Open Caveats for Firmware Release 9.4(1)

Identifier	Description
CSCtr13418	Phone keep alive timer issue in 9.2.(2) phone load
CSCtr51513	ETSGJ-CH: Conference message is showing in ENGLISH instead of JAPANESE
CSCtr93019	Can't answer call through BT headset after hold revert
CSCtt05778	No UI feedback when dialing external speed dial using softkey in onhook
CSCtz71926	Savi 7xx can't reconnect automatically after enable side USB on CUCM
CSCua13836	Pip is messed display when swapping pip and remote video
CSCud28196	Video color is totally distorted
CSCue77925	The font is enlarged so much that beyond the screen.
CSCuf78358	BT: Draggy digits tone in BT headset when doing off hook dialing
CSCuf85262	unable to end call via BT headset in some scenario
CSCuf93446	TLS connection setup performance downgraded comparing to 9.3.2
CSCug20675	The performance of hold/resume is degraded compared to 9.3.2
CSCug38873	DHCP ACK dropped by phone while dhcp server in dif sub-net and VPN on
CSCug57846	HF EQ Grabs Wrong coefficient set from dspcoefs.txt
CSCug93920	after unmute, video can't be seen on peer side
CSCuh59516	USB Headset: Adjust volume for active call can't be work
CSCuh86295	UI mess when menuname contains carriage returns in CiscoIPPhoneMenu

Identifier	Description
CSCui76146	Anonymous and DN in phone call history when phone make a call to cucm
CSCui98311	New call window appears slowly for the first call after display is on
CSCuj15819	Application switching time at phone display degraded compared to 9.3.4
CSCuj30606	call history window quits slowly
CSCuj37873	mute icon is not synced with its status of video mute
CSCuj38581	Occasionally the phone can not connect to the WIFI network
CSCuj70699	[Chinese Locale]modify phone button, phone register failed
CSCuj73452	A BT Headset(Jawbone ICON) can not redial or end a call
CSCuj82795	phone can't install LSC again after stopping LSC install in the middle
CSCuj84943	MOS valures for ISAC be zero in UI and web
CSCuj85339	Phone resets continuously in lldp scenario
CSCuj87326	Callback number not be displayed and dialed
CSCuj91054	8961 UI intermittently freezes when pressing CFAll to BLF/SD
CSCuj93829	resume hard key no response for a while
CSCuj99110	IPv6 2nd TFTP server IP address can not be removed manually
CSCuj99228	URL and URLDown not work correctly
CSCuj99322	Wireless Ipv6 phone registers to sub after dhcp shutdown and start
CSCul01945	http-tcp retransmission packets were sent very fast ; May impact EM service when there is increased network delay

Resolved Caveats

The following table lists severity 1, 2, and 3 defects that are resolved for the Cisco Unified IP Phones 8961, 9951, and 9971 for Firmware Release 9.4(1).

For more information about an individual defect, you can access the online record for the defect by clicking the Identifier or going to the URL that is shown. You must be a registered Cisco.com user to access this online information.

Because defect status continually changes, the table reflects a snapshot of the defects that were resolved at the time this report was compiled. For an updated view of resolved defects, access Bug Toolkit as described in Access Cisco Bug Search, on page 16.

Table 2: Resolved Caveats for Firmware Release 9.4(1)

Identifier	Description
CSCty35673	Xfer call bounces back when picked up @ ~same time as 2nd xfer keypress
CSCuc00963	Digit dropped via offhook dialing when application window exist.
CSCue47224	9971 Phone - Automatic Dial, when press on Edge of Phone Screen frame
CSCue67934	first tvs server down phone does not switch to the second
CSCuf30257	Video call statistics UI disappears on rx-only phone after call ended
CSCug19038	Disable Default user account and SSH Keyfile access
CSCug20992	Process "cvm" takes high CPU rate up to 98% for a long time
CSCug31051	Actional alert is malformed when much info is displayed
CSCug39839	phone crashed when USB cable was connected
CSCug56051	Phone status wrong when hold/resume with shareline several times
CSCug60897	Recording failed for 9971 with wifi connected
CSCug77567	RT phone will not work after multiple MLPP pre-emption calls
CSCug83244	Wrong DSCP for SIPS in IPv4 and Ipv6
CSCug90826	WLAN IPv4 address setting is lost
CSCug91042	Phone could not restart succesfully after moving to another cucm
CSCug91158	The outputs are not correct for CLI tail and head
CSCuh02334	Softkey wrong when header is Call statistics(video)
CSCuh08225	Size of the session index is not consistent
CSCuh10327	Potential Exists to Crash debugsh
CSCuh10334	Image Upgrade Facility Susceptible to Command Injection
CSCuh10339	Memory Leak in debugsh Command Functionality
CSCuh10343	Buffer Overflow Could Result in Restart of webapp

Identifier	Description
CSCuh10357	Arbitrary File Read via debugsh
CSCuh18809	Encrypted phone can't registered while set alt-tftp without ctl itl xml
CSCuh21202	Softkey "x" is still enabled when no digit in the simplifed dial UI
CSCuh23805	9951 SIP can't switch sendrecv video to recvonly video
CSCuh24069	Session index ID are same for different calls on a same line
CSCuh29621	Logout the hunt group will not get the notification toast.
CSCuh31749	Long press nav down key in status windows side bar does not move
CSCuh39649	Fail to answer a pickup call with Speaker when DND-R is enabled on phone
CSCuh40143	Recording failed for the first call after fail over or fall back
CSCuh44626	Audio decryption failure after rollover and re-INVITE
CSCuh45333	Interrupted g729 audio stream results in a hiss/artifact in audio
CSCuh49545	No video on 9951 phone after H/R across BFCP enabled trunk
CSCuh52810	Arbitrary File Download in ServiceabilityServlet
CSCuh57082	Sometimes the unexpected actionable incoming call alert display.
CSCuh59820	Enerywise Wake up Issue on 99xx phones
CSCuh59947	Call history is not cleared after change the line number.
CSCuh68924	Phone will not do EM login while in Remote in used status
CSCuh74432	Not unsubscribe the status of a DN when call history record is deleted
CSCuh74508	USB Headset: Volume button on ip phone can't be work in active call
CSCuh79617	Call not be made by selecting call history record that has []<> in URI
CSCuh79798	Phone UI gets stuck after hide-then-show video
CSCuh83576	Messy code in phone call history when phone make a call to cucm
CSCuh85524	Sometimes softkey is grey by swithing history and UI session
CSCuh91277	No conference window when invite 4th party before 3 party conf completed

Identifier	Description
CSCuh95110	Speaker led sometimes not on when intercom a busy call
CSCuh95112	Switch to connected intercom does not take effect immediately
CSCuh98573	number can't be input when fast key pressing
CSCui01129	Monitoring session will ended while cancel conf via session key
CSCui04382	Insecure file permissions on 4th-Gen allows privilege elevation
CSCui06106	The line text label is not right alignment for Arabic
CSCui08832	Phone not registered after EMCC login
CSCui17455	99XX phones fail to install LSC, status "Upgrade Failed: Invalid MIC"
CSCui19193	phone UI chaos when swap the call by session key in conference feature
CSCui21656	"buffer overflow" if length of string is greater than or equal to 32.
CSCui26598	Call softkey is disabled in phone detail page when E.164 dn in history
CSCui27301	Phone uses data vlan for voice
CSCui37081	The bluetooth headset still rings on RT when the call has been answered.
CSCui44633	ASSIP MLPP Authorization window error
CSCui54633	English typo in error report of ringtone
CSCui55889	99xx phone freezes during CAPF operation showing keys are generated
CSCui68116	dialout window disappear when invite 4th party before 3party conf comple
CSCui92888	"Separate audio and video mute" should not be displayed on 8961
CSCuj02881	DE36334: Low res video on call to RT Phone after escalating to video
CSCuj05841	message response key performance degrade than 931
CSCuj05937	UI stays at Applications, no response to key pressing
CSCuj08486	[share line] when resume the call, the phone show "private" bubble
CSCuj13699	Incoming alert cann't display URI
CSCuj14989	Audio quality issue on 8961 Phone when padding is present

Identifier	Description
CSCuj27767	select PFK "All Calls", Enter Call History will make phone abnormal
CSCuj27991	PLAR can not end call
CSCuj49252	Pop-up IP Phone service doesn't work intermittently
CSCuj54462	9971 Allow Barging When Ringing not working in 9-3-4-24
CSCuj69058	Jabber can't start/stop deskphone video with 9971 firmware 9-3-4-24

Unified Communications Manager Endpoints Locale Installer

By default, Cisco IP Phones are set up for the English (United States) locale. To use the Cisco IP phones in other locales, you must install the locale-specific version of the Unified Communications Manager Endpoints Locale Installer on every Cisco Unified Communications Manager server in the cluster. The Locale Installer installs the latest translated text for the phone user interface and country-specific phone tones on your system so that they are available for the Cisco IP Phones.

To access the Locale Installer required for a release, access http://software.cisco.com/download/navigator.html?mdfid=286037605&flowid=46245, navigate to your phone model, and select the Unified Communications Manager Endpoints Locale Installer link.

For more information, see the “Locale Installer” section in the *Cisco Unified Communications Operating System Administration Guide*.

Note The latest Locale Installer may not be immediately available; continue to check the website for updates.

Cisco IP Phone Firmware Support Policy

For information on the support policy for Cisco IP Phones, see http://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/unified-ip-phone-7900-series/116684-technote-ipphone-00.html.

Documentation, Service Requests, and Additional Information

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/c/en/us/td/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS Version 2.0.

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