Cisco Unified IP Phone 8961, 9951, and 9971 Release Notes for Firmware Release 9.3(1)

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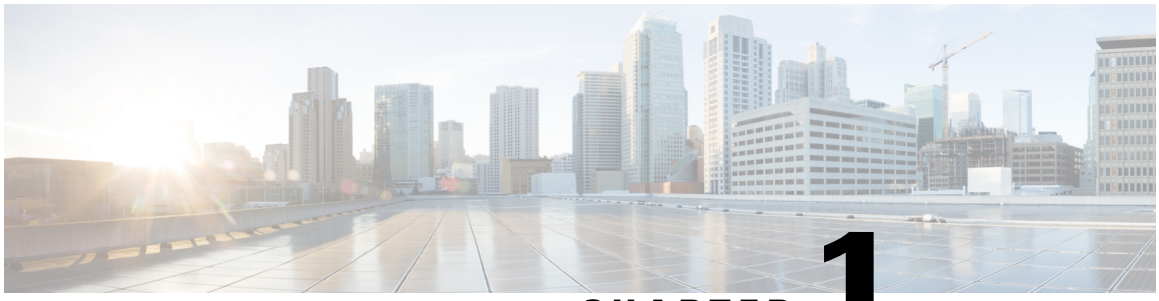

CHAPTER 1

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Introduction

These release notes support the Cisco Unified IP Phones 8961, 9951, and 9971 running SIP firmware release 9.3(1).

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

Table 1: Cisco Unified IP Phones, Cisco Unified Communications Manager, and Firmware Release Compatibility

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

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 describes the features that are new or have changed.

Features Available with Firmware Release

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

Assured Services SIP

The Assured Services for SIP Lines (AS-SIP) feature provides the ability for users to place priority calls and, if necessary, preempt lower-priority phone calls.

These AS-SIP enhancements are supported on the following SIP phones:

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

This feature is also supported on third-party telephones.

Where to Find More Information

- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Display Survivable Remote Site Telephony Message

The Display Survivable Remote Site Telephony (SRST) Message feature displays a message to the users on the phone screen when communication with the Cisco Unified Communications Manager fails. This message alerts users that some of the features of their phones are no longer available.

This feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Device Invoked Recording

The Device Invoked Recording feature enables users to control the recording of phone calls using the Record button on the phone.

Users see a status indicator on the phone display, showing when a conversation is being recorded.

The Device Invoked Recording feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Edit Speed Dial Without Restart

The Speed Dial Without a Restart feature makes it easier maintain an updated collection of Speed Dial numbers by:

- Administrators can add, modify, or delete a Speed Dial number from the Cisco Unified Communications Manager Administration page.
- Users can add, modify, or delete a Speed Dial number from the Cisco Unified Communications Manager User Options web pages.

The phone is not required to restart in order to accept these changes.

This feature does not require any specific configuration.

This enhancement is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Extension Mobility Cross Cluster Enhancement

The Extension Mobility Cross Cluster (EMCC) Enhancement feature preserves the Product Specific Configuration settings for the phone. By so doing, security policies are maintained, network bandwidth is preserved and network failure is avoided within the visiting cluster (VC).

The feature is supported on the following SIP 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 9.0 (SIP)

Handset Bass Adjustment

The Handset Bass Adjustment feature allows the phone to operate with a reduced bass tone rather than with full bass. Reduced bass removes low frequencies, which can improve muffled voices or insufficient volume on handsets. There are no administrator or user controlled settings for this feature.

The feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Pause In Speed Dial

The Pause in Speed Dial feature enables users to set up the speed dial feature to reach destinations that require a Forced Authorization Code (FAC), Client Matter Code (CMC), dialing pauses, and additional digits (such as a user extension, a meeting access code, or a voicemail password) without manual intervention. When the user presses the speed dial, the phone establishes the call to the specified DN and sends the specified FAC, CMC, and DTMF digits to the destination with dialing pauses inserted.

To include dialing pauses in the speed dial, the user must specify a comma (,) in the speed dial string. Each comma indicates a pause of 2 seconds. The comma also acts as a delimiter between destination digits, the FAC, CMC, and additional DTMF digits. The comma as delimiter is useful in the following cases:

- Differentiates overlapping dial patterns (for example 9.xxx from 9.xxxxx)

- Differentiates overlapping FAC or CMC (for example, 8787 from 87879)
- Identifies the destination number when using variable-length dial patterns (for example 9.!)

Be aware of the following requirements when you include FAC and CMC in the speed dial string:

- FAC must always precede CMC in the speed dial string.
- A speed dial label is required for speed dials containing FAC and DTMF digits.
- Only one comma is allowed between FAC and CMC digits in the string.

For any additional DTMF digits specified after the FAC and CMC, the phone dials these additional digits (with pauses) after the call is connected.

This feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

PLK Support for Queue Statistics

The PLK Support for Queue Statistics feature enables the users to query the call queue statistics for hunt pilots and the statistics display on the phone screen.

The programmable line key Queue Status can be configured by the administrator. When the user presses Queue Status, the phone displays the Queue Status screen. The Queue Status screen includes hunt pilot directory number, number of callers in queue, and the longest call waiting time in queue.

The statistics information does not update automatically. The user must press the Update softkey to view updated statistics. To exit from the queue display screen, the user presses the Exit softkey.

The feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

RTCP Hold On SIP

The RTCP Hold For SIP feature ensures that held calls are not dropped by the gateway. The gateway checks the status of the RTCP port to determine if a call is active or not. By keeping the phone port open, the gateway will not end held calls.

This feature has no administration or user impacts.

The feature is supported on the following SIP phones:

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

Secure Extension Mobility Cross Cluster

Secure Extension Mobility Cross Cluster (EMCC) enables a user in one cluster (using an encrypted/authenticated Cisco Unified IP Phone with TFTP Encrypted Config/Digest Authentication enabled) to log in to another cluster when two cluster are both in mixed mode.

Configure Cisco Extension Mobility on Cisco Unified IP Phones before you configure EMCC.

The feature is supported on the following SIP phones:

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

Where to Find More Information

Cisco Unified Communications Manager Features and Services Guide, chapter "Cisco Extension Mobility Cross Cluster"

SIP Phone No Alert Name in Placed Calls History

The SIP Phone No Alert Name in Placed Calls History feature displays the alert name in the Placed Calls history when the phone is in a translation pattern or call redirection state. Currently these calls appear on the calling party's call history as Unknown. With this enhancement these calls appear as the callee's Alert Name.

This enhancement does not require any specific configuration.

The feature is supported on following phones (SIP):

- 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 Manager 9.0 (SIP)

Uniform Resource Identifier Dialing

The Uniform Resource Identifier (URI) Dialing feature enables the user to place calls using alphanumeric URI address as a directory number, for example, bob@cisco.com. The user must enter the URI address to select the contact.

The phone screen displays the call information for the URI call. The call history record the URI call information in the Call History and the Details page.

The user cannot place calls by URI address using the soft keypad.

URI Dialing has the following feature requirements:

Onhook Call Initiation

The user must press the ABC softkey to switch the input method to URI Dialing mode using the keypad.

Off-Hook Call Initiation

The user can place calls using URI Dialing if the URI address is stored in the speed dial list or call history.

Redial

Press the Redial button to call the most recently dialed URI address.

Speed Dial

The user can configure a URI address as a speed dial entry to place a call.

Session Bubble

When the user dials or receives a call through URI Dialing, the call bubble displays the complete URI address.

Incoming Call Notification

The incoming call alert notification supports the URI address display.

Missed, Placed, and Received Call History

The URI Dialing logs are saved in the call history.

Dial URI from Call History

The user can select the URI address from the call list to place a call. The user can navigate to the URI call history or enter the URI Dial mode to place a call.

Default Domain

The user can enter the complete domain name and override the default domain.

Call History Filter

While the user enters the URI address to place a call through URI Dialing, the call history appears based on the characters entered.

Call Forward All

The user can configure the Call Forward All destination using the speed dial or call history entries.

Transfer

The user can initiate a Transfer call using URI dialing if the URI address is stored in the Speed Dial list or Call History.

Ad Hoc Conference

The user can initiate a conference call and add multiple parties using URI Dialing if the URI address is stored in the speed dial list or call history.

Privacy

The user can hide the display of the URI address information. For more information on Privacy, see *Cisco Unified Communications Manager Features and Services Guide 9.0* and *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*.

Busy Lamp Field Speed Dial

The user can monitor the state (in-use or idle) of a call using URI Dialing associated with speed dial or call history.

Call Back

The user can initiate a directory number call when the URI Dialing target becomes available.

Features Compatibility

The URI address speed dial or redial is disabled under Meet Me conference and Group Call Pickup features.

Cisco Unified Communications Manager Express and Survivable Remote Site Telephony

When the phones are connected to the Cisco Unified Communications Manager Express and Survivable Remote Site Telephony (CME/SRST), the URI Dialing functionalities are disabled. The ABC softkey does not appear on the phone screen.

This feature is supported on the following SIP phones:

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

Note Wait for the ABC softkey to appear before you proceed with URI dialing.

Where to Find More Information

- *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9851, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Unique Call ID Display

The Unique Call ID Display feature ensures that all calls with same group call ID display the same call ID on all the phones with the same shared line DN. Displaying the same call ID on all phones ensures that all users with the same shared line DN can identify the correct active call.

There is no administrator impact to this feature.

The feature is supported on the following SIP 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 Manager 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (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/en/US/docs/voice_ip_comm/cucm/compat/devpack_comp_mtx.html

Default Wallpaper Control

The Default Wallpaper Control feature allows the phone administrator to enable or disable the user preference to change the wallpaper on the phone. The phone administrator can also specify the default wallpaper image for all end users.

This feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

RTCP Control for Video

The RTCP Control for Video feature gives the administrator the flexibility to enable the phones to transmit and receive Real Time Control Protocol (RTCP) packets for audio and video streams in a video call. Using RTCP, instead of Real Time Transport Protocol (RTP), provides feedback and statistics that assist in phone system support. The choice of protocol does not impact the users. By default, the feature is disabled.

The administrator enables or disables the RTCP for video field from one of the following Cisco Unified Communications Manager windows:

- **Device** > **Phone**
- **Device** > **Device Settings** > **Common Phone Profile**

The feature is supported on the following SIP 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 Manager 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (SIP)*

sRTP Secure Video

The sRTP Secure Video feature adds more media (audio and video) encryption capabilities and gives the administrator the flexibility to choose RTCP authentication tag length between 32 bit (default) and 80 bit from Cisco Unified CM Administration.

The administrator enables or disables (default) the 80-bit SRTCP field from one of the following Cisco Unified Communications Manager windows:

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

The feature is supported on the following SIP phones:

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

This feature has no user impact.

Where to Find More Information

Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (SIP)

Simplified New Call Bubble

The Simplified New Call Bubble feature provides a simplified window for the user to place an off-hook call. The administrator enables or disables the feature in the Phone Configuration window using the Simplified New Call UI field. By default, the feature is disabled.

When the user start dialing a call with the Simplified New Call Window, the phone does not display possible phone number matches from the call history.

The feature is supported on the following SIP 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 9.0 (SIP)*
- *Cisco Unified IP Phone 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager 9.0 (SIP)*

Single Tunnel for Cisco VXC VPN

The Cisco VXC VPN feature provides integrated VPN functionality for Cisco Virtualization Experience Clients (Cisco VXC) 2111 and 2112. The feature enables VPN tunneling for the Cisco VXC 2111 and Cisco VXC 2112 clients when they are attached to a Cisco Unified IP Phone 8961, 9951, or 9971.

You can configure the Cisco Unified IP Phone and the attached Cisco VXC client to share one VPN tunnel or to use two separate tunnels. To enable the VXC VPN feature, you must set up the VPN feature for the attached IP Phone in Cisco Unified Communications Manager Administration, using the submenus under the **Advanced Features** > **VPN** menu path. In addition, you must populate the Enable VXC VPN for MAC field, using the Phone Configuration Window (**Device** > **Phone** > **Phone Configuration**).

You can set up the Cisco VXC VPN and the phone VPN to use the same tunnel or separate tunnels, in the following configurations:

- Dual Tunnel: Cisco VXC VPN traffic and phone VPN traffic use separate tunnels with the same access credentials.

To ensure the highest quality of service for the phone voice and video services, Cisco recommends the Dual Tunnel setting, which is the default setting. With two VPN tunnels, the host Cisco Unified IP Phone can provide prioritization of CPU and memory resources to the data that associates with the phone voice and with video functions over the data that associates with the Cisco VXC VPN tunnel.

- Single Tunnel: Cisco VXC VPN traffic and phone VPN traffic share one tunnel.

  All data travels over a single VPN tunnel by sharing the available phone processor and memory resources across the voice, video, and Cisco VXC services. The IP phone does not prioritize data handing of one service over another. As a result, possible performance degradation of the IP phone voice and video media handling and UI functions may occur due to IP phone CPU loading.

You can configure the feature to prompt the user only once for access credentials (using the Phone VPN Sign In window), or once each for the phone VPN (using the Phone VPN Sign In window) and the Cisco VXC VPN (using the VXC VPN Sign In window).

This feature is supported on the following SIP 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 9.0 (SIP)

Minimum Cisco VXC Firmware Release Required

To support the VXC VPN feature, the Cisco VXC clients must be running the following minimum firmware releases:

- Cisco VXC 2112: ICA Firmware Release 7.1_118
- Cisco VXC 2111: PCoIP Firmware Release 4.0 (Q3CY12)

Network Guidelines

The following are network guidelines for the Cisco VXC VPN feature implementation:

- The MTU size in the phone VPN profile is a configurable value. The default value is 1290.
- The maximum MTU value on the phone itself is hardcoded at 1406.
- The MTU value must be less than or equal to 1406, but it should not be less than 576, because some IIS and virtualization servers do not accept values less than 576.
- You must set up the firewall to allow the MTU value that you specify in the phone VPN profile.
- If the phone cannot download the certificate file or the phone configuration file, check for the allowed packet size in the network.
- If the Cisco VXC VPN cannot establish a tunnel, then ping the VPN concentrator IP address with a packet size (load) to match the MTU value that the VPN profile specifies.

- If the ping fails, try another ping that specifies no load. If the ping still fails without the load, check the routing configuration.
- If the ping fails only with the load included, check the firewall to ensure that it is configured to allow the required MTU.
- Perform a traceroute to the VPN concentrator IP address, and then ping each route with the load to determine the source of the issue.
- Ensure the Don't Fragment (DF) bit is not set on the server, network, or IP phone VPN tunnel.

Static IP Fallback for Cisco VXC 2111

At power on, the Cisco VXC 2111 attempts to obtain an IP address using DHCP. The Static IP Fallback setting on the Cisco VXC 2111 determines the client behavior if it cannot obtain an IP address using DHCP.

The following table describes how the Static IP Fallback setting affects the Cisco VXC 2111 operations. The default setting is determined by the version of firmware that was originally running when the client was shipped.

Table 2: Cisco VXC 2111 Static IP Fallback Behavior

Firmware on Shipped Client	Default Static IP Fallback Setting	Cisco VXC 2111 Behavior If DHCP Unsuccesful at Power On
Firmware release prior to 3.5.1	On	After 2 minutes, the client stops sending DHCP requests and uses a self-defined static IP.
3.5.1 firmware or later	Off	The client keeps sending DHCP requests indefinitely.

The Static IP Fallback setting will be configurable with Cisco VXC Manager release 4.9.1 (available Q3CY12).

Note If the client is factory reset, the setting will revert back to the original default setting listed.

View Call Logs From Shared Line

The Call History for Shared Line feature offers enhanced viewing of shared line activity in the Cisco Unified IP Phone call history. In addition to logging missed calls for a shared line, this feature will log all answered and placed calls on a shared line.

This feature is supported on the following SIP phones:

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

Where to Find More Information

- *Cisco Unified IP Phones 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 9.0 (SIP)*
- *Cisco Unified IP Phones 8961, 9951, and 9971 User Guide for Cisco Unified Communications Manager Guide 9.0 (SIP)*

Installation

Installation Requirements

Before you install the firmware release, you must ensure that your Cisco Unified Communications Manager is running the latest device pack.

Important If your Cisco Unified Communications Manager does not have the required device pack to support this firmware release, the firmware may not work correctly.

For information on the Cisco Unified Communications Manager Device Packs, see http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/compat/devpack_comp_mtx.html.

Install Latest Cisco Unified Communications Manager Release

Before using the Cisco Unified IP Phone with Cisco Unified Communications Manager, your Cisco Unified Communications Manager servers must be running a version of the server software that supports the phones. All Cisco Unified Communications Manager servers in the cluster must support the phones. For information about the minimum Cisco Unified Communications Manager software version that the phone requires, see the introductory sections of these release notes.

For more information on Cisco Unified Communications Manager installations and upgrades, see the documents for your Cisco Unified Communications Manager version at the following location: http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_installation_guides_list.html

To download and install the Cisco Unified Communications Manager version, perform these steps.

Procedure

Step 1 Go to the following URL:
http://www.cisco.com/cisco/software/navigator.html?mdfid=268439621&catid=278875240

Step 2 Choose your Cisco Unified Communications Manager version.

Step 3 Choose the appropriate software type.

Step 4 Hover over the desired file. When the popup window displays, click the **Readme** link to open the readme file.

Step 5 Choose **Download** or **Add to cart** for the desired file.

Step 6 Use the instructions in the readme file to install the updated file on the Cisco Unified Communications Manager.

Install Firmware Release on Cisco Unified Communications Manager

Before using the Cisco Unified IP Phone firmware Release 9.3(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 **9.3(1)**.

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

- cmterm-8961.9-3-1-33.cop.sgn
- cmterm-9951.9-3-1-33.cop.sgn
- cmterm-9971.9-3-1-33.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-8961.9-3-1-33-readme.html
- cmterm-9951.9-3-1-33-readme.html
- cmterm-9971.9-3-1-33-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-8961.9-3-1-33.zip
- cmterm-9951.9-3-1-33.zip

- cmterm-9971.9-3-1-33.zip

Note 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 **9.3(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.

Important Notes

This section contains important information to consider with the Cisco Unified IP Phone 8961, 9951, and 9971.

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.

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.

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.

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.

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.

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.

Caveats

The following sections list the open and resolved caveats, and describe how to get further information on the 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 that use Firmware Release 9.3(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 in Access Cisco Bug Search, on page 20

Table 3: Open Caveats for Firmware Release 9.3(1)

Identifier	Headline
CSCty96858	Call session label is inconsistent under various locale
CSCty90733	Tone is played incorrectly for digits pressing in whisper intercom call
CSCty44400	LED is still lighted after change phone button template
CSCtv04593	Conference list is still shown when it is disabled in FCP
CSCtz98106	Video will flash when call ended in some situation
CSCtz90586	EnergyWise will not work if IP address of phone is changed
CSCtz85424	IP addresses between 128.x.x.x-223.x.x.x are not supported in startMedia
CSCtz82754	Rings chirp1 and chirp2 have pop sounds
CSCtz77224	No standby server info when bring Sub up first then SRST
CSCtz68986	four direction arrow on TFTP input box and list item
CSCtz54715	Call history is listed for all lines but not the selected line

Identifier	Headline
CSCtz54317	Call from call history list with one touch of list item
CSCtz46348	Still show CUCM domain name as active server when stop DNS and TFTP
CSCtz34504	Cursor is displayed in a wrong location in the intercom dial window
CSCtz30127	DHCP Release turn Yes first then it turn No when setting it to No
CSCty76206	Encrypted phone will not play zip tone after CFwdAll is input
CSCty31537	SenderReportsSent of "show stream active video" is incorrect
CSCtz98112	UI mess under 7.1(5) CCM when alerting name long
CSCtz93333	RT: Can't handle rtprx and rtpmrx with volume value
CSCtz90864	DN is logged in the remote shareline phone when offhook pickup call
CSCtz90108	Appear once press speaker key can't end the call
CSCtz88254	Password Persistence not effect on RT phones
CSCtz85354	RT can't handle senddigits event
CSCtz79477	VXC VPN alert stays in screen while loses focus and softkeys
CSCtz76680	[double shot] No prompt for 4th "double shot" incoming call
CSCtz71381	WIFI toast an alert window overlapping with other UI menu
CSCtz66049	Redial does not work when there's an incoming intercom call
CSCtz43088	UI filter will focus on two items when intercom
CSCtz22269	ConcurrentModificationException when switching between the two calls
CSCtz17167	UI mess when offhook call from admin page
CSCtz11498	UI display incorrect after "Apply Config" followed by restart
CSCty93030	Mute tone instead of digit DTMF tone playing when mute caller
CSCty89938	Call time will count from -3 if lots of speed dial BLF
CSCty69131	remote placed call doesn't log DN in cross cluster if callee no answer
CSCty56877	Phone in trusted net try establish VPN when power on w/ auto net detect

Identifier	Headline
CSCti98208	6911 SIP: No Alarm(rc=18) when secure phone fallback from SRST to CUCM
CSCtz86079	KEM default logging needs to detect crashes
CSCtz86075	3rd KEM reset/crashed while on active call
CSCtz59235	8961 incorrect time after NTP server time is changed
CSCtr51513	ETSGJ-CH: Conference message is showing in ENGLISH instead of JAPANESE
CSCtx90826	the loading of image fail of phone 9951/9971 in option66 case
CSCtz65871	ALL-LANG: SRST: 99xx: Directory is blank when restored from fallback.
CSCti77356	SRST: ALL-LANG: RT Phone: "Abbreviated Dial" softkey will disappear.
CSCti77929	Impossible to transfer a call from SCCP phone to SIP phone.
CSCtz72610	Phone Caches a Service Page Despite Expired Header Set as Page Expired
CSCtz83811	RT: experimental changes
CSCtr93019	Can't answer call through BT headset after hold revert
CSCtr13418	Phone keep alive timer issue in 9.2.(2) phone load
CSCtz98079	EM login logout time performance
CSCtz82195	Bluetooth Device Does Not Reconnect
CSCtz68472	Phone cant enter into debush when no active debugsh
CSCtz71926	Savi 7xx can't reconnect automatically after enable side USB on CUCM
CSCua06066	phone not response after enter debugsh
CSCtw97451	Blackwire C220 "accessory not supported"
CSCtz51849	Group pickup call in call history do not have call duration time
CSCtz93469	Touch screen issue with call history
CSCtt05778	No UI feedback when dialing external speed dial using softkey in onhook
CSCtz45639	A triangle is displayed on title bar of call history window
CSCua08465	Call duration time string is cut off in multi-leg call log detail

Identifier	Headline
CSCua11010	PD displays abnormal when we press return button after log out.
CSCua14036	Softkey displayed as pressed after restart
CSCty31023	Dialed digits lost while making conf or xfer call quickly in video call
CSCua00229	9971 not transmitting RTP stream over the air
CSCua13836	Pip is messed display when swapping pip and remote video
CSCts01615	99xx become abnormal or crash after long period of network impairments
CSCtn89145	Joggling fullscreen selview during VGA video call to CSF softphone
CSCty01167	OpenSSL SSL_CTX_new Uninitialized Buffer Remote Information Disclosure
CSCtz51332	99xx/8961 takes long to to register after EM login in certain stuation
CSCty57481	SRST ip is still displayed on "Stand-by Server" even if srst is down
CSCtz65557	A calls B cannot get any re-order tone but silence for 500
CSCty35673	Xfer call bounces back when picked up @ ~same time as 2nd xfer keypress
CSCti79116	Memory leak during SIP Codenomicon run
CSCtz82603	8961/99XX firmware does not verify source address on Rx unicast RTP
CSCua09003	Phone does not send REFER alarm when it fallback to primary CUCM
CSCtx48830	RT phone lost wireless connectivity during call
CSCty75302	9971 does not connect to wireless when wired in (using PoE)
CSCtz40373	Call be disconnected when reauthen happen for phone
CSCtz23376	No toast prompt up when switch between wired/wireless network
CSCua15471	Sometimes dial window will lose focus and user can't input digit
CSCua17545	Phone may have a white screen when video call is quickly dropped
CSCua25787	Recording Failed toast appears twice in various fonts
CSCua26548	XSI API on CP-9971
CSCua27527	Directed call park timer start from -1 or -2 sometimes

Identifier	Headline
CSCua27553	Call Quality is Reduced When Upgrade Phone Load
CSCua30357	RTCP port will keep open when posting RTPRx/RTPTx to phone then end call
CSCua33239	UI mess when end the directed call park
CSCua22427	Top row of pixels not updated when modify SD label of LKEM
CSCts51576	9971 differ from 7945 on rtp sequence jump behavior

Resolved Caveats

The following table lists severity 1, 2, and 3 defects that are resolved for the Cisco Unified IP Phones that use Firmware Release 9.3(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 in Access Cisco Bug Search, on page 20

Table 4: Resolved Caveats for Firmware Release 9.3(1)

Identifier	Headline
CSCtt18467	Phone syslogs prints NOT[ice] level always
CSCtu49998	VXC 2111 and 2112 can't get power from phone and lkem
CSCtu53630	9971 May Intermittently Drop from WLAN and Begin Scanning
CSCtw65250	Video call with TP, PiP is transparent or all green
CSCtw99539	Phone may not switch to new load after upgrade
CSCtx20913	PiP window flashes once or twice sometimes at beginning of video call
CSCtx66216	Bandwidth calcuation not correct when multiple sessions exist
CSCtx82992	Pressing the Back button has no effect when on Edit Dial UI
CSCtx84153	Low performance if configure BLF on RT KEM module registered to CME
CSCtx85612	Lock icon may be displayed on non-secure idle UI

Identifier	Headline
CSCty03846	TFTP Server inputbox is not activated while enable alt-tftp
CSCty10161	Green strip is displayed when switching from full screen to window
CSCty25822	Unable to show video of video call made with SIPp
CSCty34439	99XX phone reboot due to chinese locale
CSCty35299	Same jad or jar file is always downloaded twice by phone
CSCty77693	no toast when calls fwd to 2nd line
CSCty60735	Phone retries using same bad password if it contains space
CSCty83995	Disable Speakerphone Parameter Allows Audio Output on 9971
CSCty91081	FAC and CMC only in speed dial not work
CSCty96383	Phone may freeze if restart it while pressing Release to get new call UI
CSCtz19102	9971 phone - one-way video when called MCU
CSCtz20164	RT phones cannot display low-res video from Jabber for Windows
CSCtz22348	Incorrect DSCP marking on IP Phone 9971 during recording
CSCtz23146	Non-normal used IP can't be displayed in TFTP address
CSCtz25519	CR/LF in QED XML cause could not save device configuration
CSCtz31444	Group pickuped call will be displayed "Unknown Caller" in call history
CSCtz43376	vpn tunnel can't be setup for route info is wrong in kernel
CSCtz45859	Intercom call can't be ended by line triggered by speed dial
CSCtz60278	the function of 'Display on when Incoming Call' can't work
CSCtz64608	RT: All Calls enabled, CID disappears w/ xfer to final called party
CSCtz74624	PiP and call statistics lost in RT video phones
CSCtz77357	Connect USB headset on usb hub 'Max num exceeded' display and not work
CSCtz84937	Mute LED showed lit, but device was unmuted
CSCtz87589	TVS Client should support same size certificate as CUCM

Identifier	Headline
CSCtz96238	Phone still send out video when mute is on

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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