Cisco Unified IP Phones

The following table contains the firmware versions for the Cisco Unified IP Phones for Cisco Unified Communications Manager Release 9.0.

Phone family	Release number
Cisco Unified IP Phone 6900 Series	9.3(1)
Cisco Unified IP Phone 7900 Series	9.3(1)
Cisco Unified IP Phone 8941 and 8945	9.3(1)
Cisco Unified IP Phone 8961, 9951, and 9971	9.3(1)
Cisco Unified Wireless IP Phone 792x Series	1.4(2)
Cisco Unified SIP Phone 3905	9.2(2)

The next sections describe the features for Firmware Release 9.3(1).

Cisco Unified IP Phones

The following table contains the firmware versions for the Cisco Unified IP Phones for Cisco Unified Communications Manager Release 9.0.

Phone family	Release number
Cisco Unified IP Phone 6900 Series	9.3(1)
Cisco Unified IP Phone 7900 Series	9.3(1)
Cisco Unified IP Phone 8941 and 8945	9.3(1)

Phone family	Release number
Cisco Unified IP Phone 8961, 9951, and 9971	9.3(1)
Cisco Unified Wireless IP Phone 792x Series	1.4(2)
Cisco Unified SIP Phone 3905	9.2(2)

The next sections describe the features for Firmware Release 9.3(1).

Cisco Unified IP Phone 6900 Series features

The following table shows the new features that the Cisco Unified IP Phone 6900 Series support with Firmware Release 9.3(1).

Feature	Supported on Cisco Unified IP Phones 6901 and 6911	Supported on Cisco Unified IP Phones 6921, 6941, 6945, and 6961
Device Invoked Recording		X
Extension Mobility Cross Cluster Enhancement		X
Headset Sidetone Control		X
Line Status for Call Lists		X
PLK Support for Queue Status		X
Ring Cadence Localization	X	X
RTCP Behavior On Hold	X	X
View Call Logs From Shared Line		X

The following sections describe the features introduced in Firmware Release 9.3(1) for the Cisco Unified IP Phone 6900 Series.

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 phones (SCCP and SIP):

- Cisco Unified IP Phone 6921

- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Extension Mobility Cross Cluster Enhancement

The Extension Mobility Cross Cluster (EMCC) Enhancement feature preserves the network and security configurations on 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 SCCP and SIP phones:

- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

Cisco Unified IP Phone 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)

Headset Sidetone Control

Headset Sidetone Control lets the user adjust the headset tone levels and can be accessed in the Preferences menu. The users can adjust headset levels to one of the four following settings:

- High
- Normal
- Low
- Off

This feature only applies to wired headsets and does not apply to Bluetooth or wireless headsets.

There is no configuration requirement.

This feature is supported on the following Cisco Unified IP Phones (SCCP and SIP):

- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941

- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)

Line Status for Call Lists

The Line Status for Call Lists feature enables the user to see the availability status of monitored line numbers in the Call History list. The administrator enables or disables this feature using the Line Status for Call Lists parameter in the Enterprise Parameters Configuration window.

When the Line Status for Call Lists parameter is Enabled, the phone line numbers in the Call History register Line Status notifications and an icon appears next to each Call History item in the Call History list. The icon notifies the user that the lines are in one of the following states:

- Idle
- Busy
- DND

When the Line Status for Call Lists parameter is Disabled, the phone line numbers in the Call History list do not register the Line Status notifications.

This feature is supported on the following Cisco Unified IP Phones (SCCP and SIP):

- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

PLK Support for Queue Statistics

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

The programmable line button 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 is not updated automatically. The user must press the Update button to view updated statistics. To exit from the queue display screen, the user presses the Exit button.

This feature is supported on the following Cisco Unified IP Phones (SCCP and SIP):

- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager (SCCP and SIP)*
- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*

Ring Cadence Localization

The Ring Cadence Localization feature allows IP Phones to use either a North American ring cadence or a Japanese ring cadence. In the Cisco Unified Communications Manager Administration, the administrator sets the Ring Locale field to be either Default or Japan from the Common Profile and phone-specific profile windows.

When Ring Locale is set to Japan, the user does not see the Ringtone entry in the Preferences menu.

The Cisco Unified IP Phone 6901 and 6911 do not have a Preferences menu.

The feature is supported on the following SCCP and SIP phones:

- 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

Where to find more information

- *Cisco Unified IP Phone 6901 and 6911 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 6901 and 6911 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

- *Cisco Unified IP Phone 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

RTCP Behavior On Hold

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 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

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 (SCCP and SIP):

- Cisco Unified IP Phone 6921
- Cisco Unified IP Phone 6941
- Cisco Unified IP Phone 6945
- Cisco Unified IP Phone 6961

Where to find more information

- *Cisco Unified IP Phones 6921, 6941, 6945, and 6961 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phones 6921, 6941, 6945, and 6961 User Guide for Cisco Unified Communications Manager Guide 9.0 (SCCP and SIP)*

Cisco Unified IP Phone 7900 Series features

The following table shows the new features that the Cisco Unified IP Phone 7900 Series support with Firmware Release 9.3(1).

Feature	Supported on Cisco Unified IP Phones 7906 and 7911	Supported on Cisco Unified IP Phone 7931	Supported on Cisco Unified IP Phone 7941, 7942, 7961, 7962	Supported on Cisco Unified IP Phone 7945, 7965, 7970, 7971, 7975
Device Invoked Recording	X	X	X	X
Extension Mobility Cross Cluster Enhancement	X	X	X	X
Headset Recording			X (except 7941 and 7961)	X (except 7970 and 7971)
Headset Sidetone Control			X	X
HTTP Firmware Upgrade	X	X	X	X
PLK Support for Queue Status				X
RTCP Behavior On Hold	X	X	X	X
Secure Extension Mobility Cross Cluster	X	X	X	X
SIP Phone No Alert Name	X	X	X	X

The following sections describe the features introduced in Firmware Release 9.3(1) for the Cisco Unified IP Phone 7900 Series.

Device Invoked Recording

The Device Invoked Recording feature enables users to control the recording of phone calls using the Record softkey 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 SCCP and SIP phones:

- Cisco Unified IP Phone 7906G
- Cisco Unified IP Phone 7911G
- Cisco Unified IP Phone 7931G

- Cisco Unified IP Phone 7941G
- Cisco Unified IP Phone 7941G-GE
- Cisco Unified IP Phone 7942G
- Cisco Unified IP Phone 7945G
- Cisco Unified IP Phone 7961G
- Cisco Unified IP Phone 7961G-GE
- Cisco Unified IP Phone 7962G
- Cisco Unified IP Phone 7965G
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7971G-GE
- Cisco Unified IP Phone 7975G

Where to find more information

- *Cisco Unified IP Phone 7906G, and 7911G Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7906G, and 7911G User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7931G Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7931G User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7962G, 7961G-GE, 7961G, 7942G, 7941G-GE, and 7941G Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7962G, 7961G-GE, 7961G, 7942G, 7941G-GE, and 7941 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7975G, 7971G-GE, 7970G, 7965G, and 7945G Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7975G, 7971G-GE, 7970G, 7965G, and 7945G User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Extension Mobility Cross Cluster Enhancement

The Extension Mobility Cross Cluster (EMCC) Enhancement feature preserves the network and security configurations on 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 SCCP and SIP phones:

- Cisco Unified IP Phone 7906
- Cisco Unified IP Phone 7911

- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975

Where to find more information

- *Cisco Unified IP Phone 7906 and 7911 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7931 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7962, 7961, 7942, and 7941 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7975, 7971, 7970, 7965, and 7945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Headset Recording

The Headset Recording feature adds the audio from the phone handset into the phone headset. If conversations are recorded using the auxilliary port of the headset, the feature ensures that audio from the headset (the agent) and the phone handset (the supervisor) are captured in the recording.

To support this feature, a new field, Headset Recording, is introduced in the Cisco Unified Communications Manager.

For phones associated with Cisco Unified Communications Manager 8.6 and earlier, this feature requires a dev pack.

The feature is supported on the following SCCP and SIP phones:

- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7975

Where to find more information

- *Cisco Unified IP Phone 7941, 7942, 7961, and 7962 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7945, 7965, 7970, 7971, and 7975 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

HTTP Firmware Upgrade

The HTTP Firmware Upgrade feature enhances the firmware download process by first attempting to download all files using HTTP. If the phone rejects the HTTP connection, the upgrade process uses the existing TFTP download process

There is no user impact to this feature.

The feature is supported on the following SCCP and SIP phones:

- Cisco Unified IP Phone 7906
- Cisco Unified IP Phone 7911
- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975

Where to find more information

- *Cisco Unified IP Phone 7906 and 7911 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7931 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7941, 7942, 7961, and 7962 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7945, 7965, 7970, 7971, and 7975 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and 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 information appears on phone screen.

The programmable line button 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 is not updated automatically. The user must press the Refresh button to view updated statistics. To exit from the queue display screen, the user presses the Exit button.

The feature is supported on the following SCCP and SIP phones:

- Cisco Unified IP Phone 7906
- Cisco Unified IP Phone 7911
- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975

Where to find more information

- *Cisco Unified IP Phone 7945, 7965, 7970, 7971, and 7975 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7945, 7965, 7970, 7971, and 7975 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

RTCP Behavior On Hold

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 7906
- Cisco Unified IP Phone 7911

- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7945
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975

Where to find more information

- *Cisco Unified IP Phone 7906 and 7911 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7931 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7941, 7942, 7961, and 7962 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 7945, 7965, 7970, 7971, and 7975 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Secure Extension Mobility Cross Cluster

The Secure Extension Mobility Cross Cluster (EMCC) feature enables a user configured in one cluster to log into a Cisco Unified IP Phone in another cluster. The users from a home cluster log into a Cisco Unified IP Phone at a visiting cluster. The visiting cluster can log into home cluster in secure mode.

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

The feature is supported on the following phones (SCCP and SIP):

- Cisco Unified IP Phone 7906
- Cisco Unified IP Phone 7911
- Cisco Unified IP Phone 7931
- Cisco Unified IP Phone 7941
- Cisco Unified IP Phone 7942
- Cisco Unified IP Phone 7961
- Cisco Unified IP Phone 7962
- Cisco Unified IP Phone 7965
- Cisco Unified IP Phone 7945

- Cisco Unified IP Phone 7970
- Cisco Unified IP Phone 7971
- Cisco Unified IP Phone 7975

Where to find more information

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

Cisco Unified IP Phone 8941 and 8945 features

The following list identifies the new features that the Cisco Unified IP Phone 8941 and 8945 support with Firmware Release 9.3(1).

- Call Log Filter Enhancement
- Extension Mobility Cross Cluster Enhancement
- Headset Sidetone Control
- Line Status for Call Lists
- Pause In Speed Dial
- PLK Support For Queue Statistics
- RTCP Control For Video
- sRTP Secure Video
- SIP Phone No Alert Name
- Secure Extension Mobility Cross Cluster
- Video Disable

The following sections describe the features introduced in Firmware Release 9.3(1) for the Cisco Unified IP Phone 8941 and 8945.

Call Log Filter Enhancement

The Call Log Filter Enhancement feature assists the user make a call by checking the call history as the user dials and presenting a list of entries that match the digits as they are input. The user can select one of the displayed numbers instead of entering the complete telephone number.

To disable call log filtering, the administrator enables the Simplified New Call UI field. By default, call log filtering is enabled.

The feature is supported on the following phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Device Invoked Recording

The Device Invoked Recording feature enables users to control the recording of phone calls using the Record programmable line key 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 phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 8941and 8945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Extension Mobility Cross Cluster Enhancement

The Extension Mobility Cross Cluster (EMCC) Enhancement feature preserves the network and security configurations on the phone. By so doing, security policies are maintained.

The feature is supported on the following SCCP and SIP phones:

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)

Headset Sidetone Control

The Headset Sidetone Control feature enables users to adjust the headset sidetone level, from the phone Preference screen. This menu item provides four different headset sidetone levels:

- High
- Normal
- Low

- Off

Note This feature only applies to wired headsets and does not apply to Bluetooth or wireless headsets.

This feature is supported on the following Cisco Unified IP Phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0.

Line Status for Call Lists

The Line Status for Call Lists feature enables the user to see the availability status of monitored line numbers in the Call History list. The administrator enables or disables this feature using the Line Status for Call Lists parameter in the Enterprise Parameters Configuration window.

When the Line Status for Call Lists parameter is Enabled, the phone line numbers in the Call History register Line Status notifications and an icon appears next to each Call History item in the Call History list. The icon notifies the user that the lines are in one of the following states:

- Unknown
- Idle
- Busy
- DND

When the Line Status for Call Lists parameter is Disabled, the phone line numbers in the Call History list do not register the Line Status notifications.

This feature is supported on the following Cisco Unified IP Phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Pause In Speed Dial

The Pause In Speed Dial feature allows 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.!)

Note Including FAC and CMC in the speed dial string has the following requirements:

- 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 8941
- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manage 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and 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 information appears on phone screen.

The programmable line button 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 is not updated automatically. The user must press the Update button to view updated statistics. To exit from the queue display screen, the user presses the Exit button.

The feature is supported on the following SCCP and SIP phones:

- Cisco Unified IP Phone 8941

- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager (SCCP and 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 both 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 protocol choice does not impact the users.

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 SCCP and SIP phones:

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager 9.0(SCCP and 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 (default) or disables 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 8941
- Cisco Unified IP Phone 8945

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager (SCCP and SIP)*

SIP Phone No Alert Name

The SIP Phone No Alert Name feature makes it easier for end users to identify transferred calls by displaying the original caller's phone number. The transferred call appears as an Alert Call followed by the caller's telephone number.

This enhancement does not require any specific configuration.

The feature is supported on following phones (SIP only):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)

Secure Extension Mobility Cross Cluster

The Secure Extension Mobility Cross Cluster (EMCC) feature enables a user configured in one cluster to log into a Cisco Unified IP Phone in another cluster. The users from a home cluster log into a Cisco Unified IP Phone at a visiting cluster. The visiting cluster fails to log into home cluster in secure mode.

Configure Cisco EMCC on the Cisco Unified IP Phones before you configure secure EMCC.

The following table describes the security level of EMCC when both the home and visiting clusters are release 9.0 clusters.

9.0 visiting cluster	9.0 Visiting phone	9.0 home cluster Mixed Mode	9.0 home cluster Nonsecure Mode
Mixed Mode	Secure	Secure EMCC	Log on fails
	Nonsecure	Nonsecure EMCC	Nonsecure EMCC
Nonsecure Mode	Nonsecure	Nonsecure EMCC	Nonsecure EMCC

The following table describes the security level of EMCC when the home cluster is at release 9.0 and the visiting cluster is at release 8.x.

8.x visiting cluster	8.x Visiting phone	9.0 home cluster Mixed Mode	9.0 home cluster Nonsecure Mode
Mixed Mode	Secure	Not supported	Not supported
	Nonsecure	Nonsecure EMCC	Nonsecure EMCC
Nonsecure Mode	Nonsecure	Nonsecure EMCC	Nonsecure EMCC

The following table describes the security level of EMCC when the home cluster is at release 8.x and the visiting cluster is at release 9.0.

9.0 visiting cluster	9.0 Visiting phone	8.x home cluster Mixed Mode	8.7 home cluster Nonsecure Mode
Mixed Mode	Secure	Log on fails	Log on fails
	Nonsecure	Nonsecure EMCC	Nonsecure EMCC
Nonsecure Mode	Nonsecure	Nonsecure EMCC	Nonsecure EMCC

The feature is supported on the following phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

Where to find more information

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

Video Disable

The Video Disable feature allows the user to control the sending of video on a call. When the administrator has enabled Video Capability in the Cisco Unified Communications Manager Administration, the user can control the video.

The user sees the Video entry in the Preferences menu, and can enable and disable the ability of the phone to send video.

The feature is supported on the following phones (SCCP and SIP):

- Cisco Unified IP Phone 8941
- Cisco Unified IP Phone 8945

This feature is also supported on third-party telephones.

Where to find more information

- *Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*
- *Cisco Unified IP Phone 8941 and 8945 Administration Guide for Cisco Unified Communications Manager 9.0 (SCCP and SIP)*

Cisco Unified IP Phone 8961, 9951, and 9971 features

The following list identifies the new features that the Cisco Unified IP Phone 8961, 9951, and 9971 support with Firmware Release 9.3(1).

- Default Wallpaper Control
- Device Invoked Recording
- Display SRST Message
- Extension Mobility Cross Cluster Enhancement
- Handset Audio Tuning
- Handset Bass Adjustment
- Line Status for Call Lists
- Pause In Speed Dial
- PLK Support For Queue Statistics
- RTCP Control For Video
- RTCP Hold On SIP
- Secure Extension Mobility Cross Cluster
- Simplified New Call Bubble
- SIP Phone No Alert Name
- Speed Dial Without Restart
- sRTP Secure Video
- Unique Call ID Display
- URI Dialing

The following sections describe the features introduced in Firmware Release 9.3(1) for the Cisco Unified IP Phone 8961, 9951, and 9971.

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)*

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)*

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)*

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 Audio Tuning

Handset Bass Adjustment

The Handset Bass Adjustment feature removes some of the low frequencies on a narrowband handset call, which can improve muffled voices or insufficient volume on handsets. The default setting is for reduced bass.

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

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)*

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"

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)*

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)

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)

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)*

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)*