**CUCM Comparison Doc 4.1 - 8.6(1a)**

Cisco CallManager 4.1(3)

•Operating System Installation Guidelines

•Upgrading to Cisco CallManager Release 4.1(3)

•Call Pickup and Group Call Pickup Enhancements

•Call Transfer Enhancements

•New Feature Support with Cisco IP Manager Assistant

•New Service Parameter to Enable Music On Hold and Annunciator Duplex Streaming

•Updated Support for Cisco IP Phones

•New Hunt Pilot Configuration Parameters for Automated Alternate Routing

•Support for Annex M1 over H.225 Trunks and H.323 Gateways

•Security Enhancements

•Bulk Administration Tool Enhancements

•Dialed Number Analyzer Enhancements

•New and Changed Information for Cisco CallManager Serviceability

•New and Changed Information for Cisco CallManager Serviceability

•New and Changed Information for Third-Party and SDK Applications

http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/rel\_notes/4\_1/cucm-rel\_notes-413.html#wp1708005

CCM 4.2.x New Features

New and Changed Information for Cisco CallManager Features CCM 4.2

The following sections describe new and changed information related to Cisco CallManager features:

Multilevel Precedence and Preemption Enhancements

Device Mobility

Cisco CallManager Attendant Console and Call Pickup Limitation

\*\* Call Pickup Notification

\*\* Directed Call Park

IPMA Enhancements

Cisco Messaging Interface Enhancements

Call Forwarding Enhancements

Password Management Features

\*\* Log Out of Hunt Groups

\*\* Voice-Mail-Related Changes to Hunt List and Hunt Pilot

New MLPP Service Lockout Enterprise Parameter

Phone Button Template

Call Diagnostics and Voice Quality Metrics

Overlap Sending and Receiving for H.323 Gateways

SDL Traces

Cisco CallManager T1 CAS Hookflash Transfer Support

From this doc;

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod\_release\_note09186a00805f1470.html#wp2430214

Here is the related CCM 5.x and CCM 6.x info :)

Cisco CallManager 5.x Features

Call Forwarding Enhancements

Call Park

Call Pickup Group

CTI and CTI Super Provider

Dynamic Host Control Protocol (DHCP)

International Dial Plans

LDAP Directory Enhancements

Licensing

Personal Directory Enhancements

Phone NTP References for SIP Phones

Presence

Roles and User Groups

RSVP

Service Enhancements

SIP Line Side (Endpoints) Support

SIP Route Pattern

SIP Third-Party Phones

SIP Trunk Enhancements

TFTP Enhancements

URL Dialing

Video Telephony Enhancements

Voice Quality Metrics

http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/rel\_notes/5\_0\_1/501cmrn.html

Cisco Unified Communications Manager 6.0 feature enhancements:

•AAC/iLBC Voice Codec Support

•Advanced Ad Hoc Conference

•Audible Message Waiting Indicator

•Barge Enhancements

•Call Diagnostics and Voice Quality Metrics

•Call Forward Enhancements

•Call Forward All Calling Search Space Backward Compatibility

•Call Forward Overriding

•Call Pickup Notification

•Cisco Messaging Interface Enhancements

•Cisco Unified Communications Manager T1 CAS Hookflash Transfer Support

•Connected Number Display

•Credential Policy and User Authentication

•CTI Enhancements

•Device Mobility

•Directed Call Park

•Do Not Disturb

•Hold Reversion

•Intercom

•Licensing Enhancements

•Log Out of Hunt Groups

•Overlap Sending and Receiving for H.323 Gateways

•MGCP T.38 Enhancements

•Privacy on Hold

•Programmable Line Keys

•SCCP Optimization

•SDL Traces

•SIP Endpoints Support

•SIP Third-Party Phones Enhancements

•SIP Trunk Enhancements

http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/rel\_notes/6\_0\_1/601cmrn.html#wp44812

CUCM 7.0 New Features;

When you reinstall or restore data to a first node in a cluster, you no longer need to reinstall the subsequent nodes in the cluster.

The Apply a Patch option of the installation program now supports applying full patch ISO upgrade patch file, in addition to the previously supported ES and SU patch types.

Busy Lamp Field Pickup

\*\*•Call Forward All Loop Prevention and Breakout

•Calling Party Normalization

•Cisco Emergency Responder Location Management Support in Application Server Configuration Window

•Cisco Extension Mobility Feature Safe

•Cisco Unified Communications Manager Attendant Console Support in 7.0

•Cisco Unified IP Phone Expansion Module 7915 and 7916 Support

•Cisco Unified Mobility—Cisco Unified Mobile Communicator

•Cisco Unified Mobility—Dial-via-Office Reverse Callback

•Cisco Unified Mobility—Directed Call Park via DTMF

•Cisco Unified Mobility—SIP URI Dialing

•Cisco Unified Mobility—Time-of-Day (ToD) Access

•Cisco Click-to-Conference Plug-In with IBM SameTime

•Directed Call Pickup

•Do Not Disturb Call Reject

•G.Clear Codec Support

•G.729a and G.729b Codecs Over SIP Trunks

•Enhanced IP Phone Services

•International Escape Character + Support

•Local Route Groups

•Microsoft Windows Server 2008 Active Directory (AD) Support for LDAP Synchronization

•Non-Urgent Translation Patterns

•Privacy Headers for SIP Trunks

•SIP Support for Cisco Unified Communications Manager Features

•SIP T.38 Interoperability with Microsoft Exchange

•Trusted Relay Points

\*\*•Cisco VG202 and VG204 Gateway Support in Cisco Unified Communications Manager Administration

•Voice over Secure IP for SIP Trunks

–Multilevel Precedence and Preemption Enhancements

Cisco Extension Mobility (EM) equivalency eliminates the phone-model dependency of phone button templates. The following factors determine the model equivalency among the various phones:

•Various features that the phone models support

•Number of buttons that the phone models support

EM equivalency includes the following support feature for the Cisco Unified IP Phones:

•Feature Safe on Phone Button Template—Phones can use any phone button template that has the same number of line buttons that the phone model supports.

Release 7.0(1) of Cisco Unified Communications Manager enhances the existing Extension Mobility (EM) equivalency mechanism. The equivalency enhancement works across phone types as follows:

•7940 SCCP, 7941 SCCP, 7942 SCCP, and 7945 SCCP models, which are equivalent, can share an EM profile.

•7940 SIP, 7941 SIP, 7942 SIP, and 7945 SIP models, which are equivalent, can share an EM profile.

•7960 SCCP and 7961 SCCP models, which are equivalent, can share an EM profile.

•7962 SCCP and 7965 SCCP models, which are equivalent, can share an EM profile.

•7960 SIP, 7961 SIP, 7962 SIP, and 7965 SIP models, which are equivalent, can share an EM profile.

•7970 SCCP and 7971 SCCP models, which are equivalent, can share an EM profile.

•7970 SIP, 7971 SIP, and 7975 SIP models, which are equivalent, can share an EM profile.

The enhancement works for all phone models that are equivalent and requires no administration tasks to activate.

Directed Call Pickup

The Directed Call Pickup feature allows a user to pick up a ringing call on a directory number (DN) directly by pressing the GPickUp softkey and entering the directory number of the device that is ringing. Cisco Unified Communications Manager uses the associated group mechanism to control the privilege of a user who wants to pick up an incoming call by using Directed Call Pickup. The associated group of a user specifies one or more call pickup groups that have been associated to the pickup group to which the user belongs.

Import and Export Enhancements for the Bulk Administration Tool

The Import/Export tool in BAT includes new updates to support the export of Cisco Unified Communications Manager configuration details. It also has a new feature to validate .tar import files.

Backup of CAR Database

The CAR and CDR Disaster Recovery Service (DRS) now integrates into the Disaster Recovery System (DRS). The DRS includes the backup of the CAR database, pregenerated reports, and the CDR preserved flat files.

Call Forward All Loop Breakout and Prevention

Cisco Unified Communications Manager 7.0(1) introduces enhancements for the following Call Forward All features:

•Call Forward All Loop Prevention—Prevents the end-user from configuring a Call Forward All destination on the phone that will create a Call Forward All loop or that will create a forward chain with more hops than the existing Forward Maximum Hop Count service parameter allows.

•Call Forward All Loop Breakout—Detects and breaks Call Forward All loops, which allows the call to ring on the phone that would have closed the loop. Prior to this release, the call got cleared if a Call Forward All loop was detected through the Forward Maximum Hop Count service parameter.

Parallel Upgrades from Unified CM Releases 5.x and 6.x to Unified CM Release 7.0(1)

When you upgrade a cluster running a supported version of Cisco Unified Communications Manager 5.x or 6.x to Cisco Unified Communications Manager 7.0(1), begin upgrading the first node first. You can begin upgrading subsequent nodes in parallel after the first node has reached a specified point in the upgrade.

<http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/7_0_1/cucm-rel_notes-701.html>

CUCM 8.0 new features

**Cisco UCS C210 Rack-Mount Server Support**

Cisco Unified Communications Manager Releases 8.0(2c) and later can operate on virtualized servers on VMware ESXi 4.0 and later only. Cisco supports VMware ESXi 4.0 and later on the Cisco UCS C210 Rack-Mount Server in Reference Configurations 1, 2 and 3.

### Changes to Available CLI Commands

The following changes were made to the available CLI commands:

•The **file list sftpdetails** command was removed. In its place, the **file dump sftpdetails** command is now interactive and provides a list of files that the user can dump (display).



•The **utils disaster\_recovery history** command got added. This command displays the history of previous backups and restores.



•The **utils disaster\_recovery show\_tapeid** is disabled for VM deployments.



•The **utils disaster\_recovery show\_backupfiles** command now takes a device name and the tape and network options are removed.



•The **utils disaster\_recovery backup tape** command is disabled for VM deployments.



•The **utils disaster\_recovery restore tape** command is disabled for VM deployments.



•The **utils disaster\_recovery device add tape** command is disabled for VM deployments.



•The **utils disaster\_recovery device add network** command now accepts a server name or IP address (previously the server name was required).



### De-allocation of Transcoder

The de-allocation of transcoder feature enables Cisco Unified Communications Manager to support mid-call de-allocation of unused transcoder resources. Cisco Unified Communications Manager can now de-allocate a transcoder allocated due to codec mismatch, for scenarios when it receives a mid-call invite with updated codec change from the SIP trunk or the SIP line side device.

In previous releases of Cisco Unified Communications Manager, if the transcoder was allocated because of codec mismatch and if a SIP trunk or SIP line side device with incoming re-invite changed the codec, the previously negotiated codecs were retained and the transcoder continued to be used. Also, if the codec completely changed from the previously negotiated one, then it depended on whether the allocated transcoder supported the new codec combination. In case the transcoder was not universal, it was not able to support the call and the call ended.

With the introduction of the de-allocation of transcoder feature, Cisco Unified Communications Manager 8.5(1) is able to free up the expensive transcoder devices by de-allocating these when not required. Based on the mid-call changes, Cisco Unified Communications Manager can now de-allocate transcoders and allocate MTPs instead, if required.

If the new SDP has the previously negotiated codec along with new codecs and if one of the new codecs, after filtering for bandwidth limitations, matches with any of the codec(s) supported by the peer endpoint, the transcoder is de-allocated.

**Exceptions**

There are a few exceptions to de-allocating a transcoder. The de-allocation of transcoder feature does not support the following, if not supported already:

•De-allocation of transcoder when inbound mid call updated capabilities come in from a non-SIP endpoint.



•De-allocation of transcoder when updated video codecs come in the mid-call re-invite.



•Currently transcoders do not support transcoding for video codec mismatch. If a transcoder is present because of audio codec mismatch, then there will be no video on the call.



•The allocation/de-allocation of transcoder resources and MTPs due to DTMF mismatch is already being taken care of by Cisco Unified Communications Manager, and this feature does not modify any existing functionality related to DTMF. Therefore, this feature does not support any unsupported cases of transcoder de-allocation due to DTMF mismatch.



•In scenarios where more than one media resource is allocated for the media leg of a call, the transcoder does not get de-allocated. For example, a scenario where a transcoder as well as an MTP get allocated because of the "MTP required" checkbox being selected on a party.



•This feature is disabled when there is an RSVP agent in the media call leg.



•This feature is disabled when call flows with mid call re-Invites without SDP are received.



### Cisco Unified Communications Manager Call Detail Records

This section contains these subsections:

•[New CDR Field for Hunt List Support](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_0_1/delta/8_0_3.html#wp82790)



### New CDR Field for Hunt List Support

[Table 1-1](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_0_1/delta/8_0_3.html" \l "wp82705) describes the new CDR field for the hunt list support.

|  |  |  |
| --- | --- | --- |
| Table 1-1 ***New CDR Field for Hunt Lists*** | | |
| **Name** | **Range of Values** | **Description** |
| calledPartyPatternUsage | Positive Integer | This field indicates the pattern of the called party.  Default value is 5 (PATTERN\_ROUTE).  •If the huntPilotDN is populated, use the field value of huntPilotDN as the hunt pilot.  •If the huntPilotDN is not available, check the pattern usage (7 = PATTERN\_HUNT\_PILOT) in the CDR table to identify the call type. If this call is a hunt list call, use the finalCalledPartyNumber as the huntPilotDN. |

### Assisted Directed Call Park

The Assisted Directed Call Park feature enables users to park a call by pressing only one button using the Direct Park feature. This requires administrators to configure a Busy Lamp Field (BLF) Assisted Directed Call Park button. When users press an idle BLF Assisted Directed Call Park button for an active call, the active call is parked at the Direct Park slot associated with the Assisted Directed Call Park button.

This feature is supported on the following Cisco Unified IP Phones (SIP) for Cisco Unified CM 7.1(5) and later:

–Cisco Unified IP Phone 7975G



–Cisco Unified IP Phone 7971G-GE



–Cisco Unified IP Phone 7970G



–Cisco Unified IP Phone 7965G



–Cisco Unified IP Phone 7962G



–Cisco Unified IP Phone 7961G



–Cisco Unified IP Phone 7961G-GE



–Cisco Unified IP Phone 7945G



–Cisco Unified IP Phone 7942G



–Cisco Unified IP Phone 7941G



–Cisco Unified IP Phone 7941G-GE



–Cisco Unified IP Phone 7931G



–Cisco Unified IP Phone 9971G



–Cisco Unified IP Phone 9951G



–Cisco Unified IP Phone 8961G



### Dual Bank Firmware Upgrade

With this release, the firmware upgrade process has been enhanced with a dual-banking capability. Now a new firmware image can be set to download in advance of a maintenance window. Then instead of waiting for all of the phones to download the firmware, the system switches more rapidly between resetting an existing load to Inactive status and installing the new load.

This reduces delays and congestion in downloading during the maintenance windows.

This feature is supported on the following Cisco Unified IP Phones:

•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



**Secure and Nonsecure Indication Tone**

With Firmware Release 9.0(3), the secure indication tone functionality was updated, and the nonsecure indication tone was added to the Secure and Nonsecure Indication Tone feature for the Cisco Unified IP Phones. The 8.0(3) release of Cisco Unified Communications Manager (Unified CM) is a requirement for these changes to function.

If phone is configured as secure (encrypted and trusted) in Unified CM, it can be given a "protected" status (which is separate from the status a call). After that if desired, the protected phone can be configured to play an indication tone at the beginning of a call:

•Protected Device—To change the status of a secure phone to protected, check the "Protected Device" check box in Cisco Unified Communications Manager Administration > Device > Phone > Phone Configuration.



•Play Secure Indication Tone—To enable the protected phone to play a secure or nonsecure indication tone, set the "Play Secure Indication Tone" to True. (The default is False.) You set this option in Cisco Unified Communications Manager Administration > System > Service Parameters. Select the server and then the Unified CM service. In the Service Parameter Configuration window, select the option in the Feature - Secure Tone area. (The default is False.)



Only protected phones hear secure or nonsecure indication tones. (Nonprotected phones never hear tones.) Because the condition for playing the secure indication tone is now based on the overall secure status of the call end to end and not the protected status of the phone, users hear a tone between a protected phone and a nonprotected phone if the Secure Real-Time Transfer Protocol (SRTP) or Real-Time Protocol (RTP) is established.

If the overall call status changes during the call, the indication tone changes accordingly. At that time, the protected phone plays the appropriate tone.

A protected phone plays a tone or not under these circumstances:

•When the option to play a tone, "Play Secure Indication Tone," is enabled (True):



–When end-to-end secure media is established through the Secure Real-Time Transfer Protocol (SRTP) and the call status is secure, the phone plays the secure indication tone (three long beeps with brief pauses).



–When end-to-end nonsecure media is established through the Real-Time Protocol (RTP) and the call status is nonsecure, the phone plays the nonsecure indication tone (six short beeps with brief pauses). (This capability is a change with this release.)



•When the Play Secure Indication Tone option is disabled (False), no tone is played.



These changes were also made with this release:

–Users can invoke supplementary services, such as Transfer or Conference, from protected phones without a software limitation.



–In the past if calls were transferred from a protected phone to another protected phone with RTP established, the call would be dropped. Now users hear a secure or nonsecure indication tone instead of the call being dropped.



The Secure and Nonsecure Indication Tone feature is supported on these IP phones running the SCCP and SIP protocol:

•Cisco Unified IP Phone 7975G



•Cisco Unified IP Phone 7971G-GE



•Cisco Unified IP Phone 7970G



•Cisco Unified IP Phone 7965G



•Cisco Unified IP Phone 7962G



•Cisco Unified IP Phone 7961G-GE



•Cisco Unified IP Phone 7961G



•Cisco Unified IP Phone 7945G



•Cisco Unified IP Phone 7942G



•Cisco Unified IP Phone 7941G-GE



•Cisco Unified IP Phone 7941G



•Cisco Unified IP Phone 7931G



•Cisco Unified IP Phone 7911G



•Cisco Unified IP Phone 7906G



•Cisco Unified IP Phone 9971



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 8961



**Secure Extension Mobility**

The Extension Mobility HTTPS Support feature ensures that when communications are exchanged between a Cisco Unified IP Phone service and other applications, that the communications use the HTTPS protocol to ensure that the communications are secure. Users must log into the Cisco Unified CM applications by providing their authentication information. Their credentials are encrypted after the communication protocol changes to HTTPS.

When a visiting Extension Mobility (EM) application fails to locate a user's identification in the local database, the following occurs:

**1.** Cisco Extension Mobility Cross Cluster (EMCC) sends a request to the local EM service to determine the home cluster of that user (the cluster which owns the user's identification, and which can handle the EM login).



**2.** The visiting EM service sends a user identification message over HTTPS to all the remote clusters added in the local database.



**3.** The visiting EM service then parses the response received from the home cluster to get the list of device profiles associated with that user.



–All further communication between the visiting EM service and home EM service takes place over HTTPS.



–Similarly, visiting logout requests are also sent from the home EM service to the visiting EM service over HTTPS.



The Extension Mobility HTTPS Support feature is supported on the following IP phones (SIP):

•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



**Note** Configure Cisco Extension Mobility on Cisco Unified IP Phones before configuring EMCC.



<http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_0_1/delta/8_0_3.html>

CUCM 8.5 new features

### Agent Greeting

Agent Greeting allows an agent or administrator to create and play a prerecorded greeting automatically at the beginning of a call, such as a customer call, before the agent begins the conversation with the caller. An Agent can prerecord a single greeting or multiple ones as needed and create and update them. When a customer calls, both callers hear the prerecorded greeting. The agent can remain on mute until the greeting ends or answer the call over the greeting. All codecs supported for the phone are supported for Agent Greeting calls.

To enable Agent Greeting in the Cisco Unified Communications Manager Administration application, choose **Device > Phone**, locate the Cisco Unified IP Phone that you want to configure. Scroll to the Device Information Layout pane and set Built-in Bridge to **On** or **Default**. If Built-in Bridge is set to Default, in the Cisco Unified Communications Manager Administration application, choose **System > Service Parameter** and select the appropriate Server and Service. Scroll to the Clusterwide Parameters (**Device - Phone**) pane and set Built-in Bridge Enable to **On**.

**For More Information**

•Agent Greeting feature documentation, *CTI Product Description Guide for Cisco Unified Contact Center Enterprise, Release 8.5(1)*



•Features Supported by Cisco Unified JTAPI, *Cisco Unified JTAPI Developers Guide for Cisco Unified Communications Manager 8.5(1)*



•Features Supported by TSP, *Cisco Unified TAPI Developers Guide for Cisco Unified Communications Manager 8.5(1)*



### Assisted Directed Call Park Support Enhancements

Cisco Unified Communications Manager 8.5(1) supports Assisted Directed Call Park on the Cisco Unified IP Phones 7900 Series (SIP).

Assisted directed call park is supported on all Cisco Unified IP Phones 7900, 8900, and 9900 series that support SIP. Assisted directed call park means that the end user needs to press only one button to direct-park a call. This requires you to configure a BLF Directed Call Park button. Then, when the user presses an idle BLF Directed Call Park feature button for an active call, the active call will be immediately parked at the Dpark slot associated with the Directed Call Park feature button.

### Call Back No Answer Support for VG224

Cisco Unified Communications Manager Release 8.0 and earlier supported Call Back only on busy subscriber for Cisco VG224 endpoints. Cisco Unified Communications Manager Release 8.5 and later supports Call Back No Answer for Cisco VG224 endpoints.

For more information on the Call Back feature, see the *Cisco Unified Communications Manager Features and Services Guide*.

### Cisco Mobile 8.0 Support

Cisco Unified Communications Manager supports SIP-based dual-mode mobile phones with Cisco Mobile 8.0. Cisco Unified Communications Manager supports the new Cisco Dual Mode for iPhone device type for iPhone, which specifies a SIP-based dual-mode mobile phone that is capable of leveraging VoIP connectivity over the enterprise WLAN.

Cisco Unified Communications Manager supports dual-mode mobile phones that use the Cisco Unified Mobile Communicator client and SIP protocol within the WLAN. Cisco Unified Communications Manager must handle dual SIP registrations (one from Cisco Unified Mobile Communicator via Cisco Unified Mobility Advantage and one from Wi-Fi as a SIP endpoint).

Cisco Mobile 8.0 provides iPhone users with voice over IP (VoIP) calling, visual voicemail, and access to the corporate directory while users are connected to the corporate network over Wi-Fi, either on premises or over VPN. Cisco Mobile 8.0 specifies an IP telephony endpoint that associates with Cisco Unified Communications Manager.



**Note** Cisco Mobile 8.0 is distinct from the Cisco Mobile application that runs in conjunction with a Cisco Unified Mobility Advantage server.



In order for Cisco Unified Communications Manager to support Cisco Mobile 8.0, Cisco Unified Communications Manager administrators must take at least the following step:

**1.** Configure the new device in Cisco Unified Communications Manager Administration.



**Note** Note that in Release 8.5(1) of Cisco Unified Communications Manager Administration, downloading of a COP file is not required.



The Administration Guide for Cisco Mobile 8.0 for iPhone provides the details of the complete configuration that is required to configure Cisco Mobile 8.0, including the steps that must be performed in Cisco Unified Communications Manager Administration. Refer to the document at the following URL:

<http://www.cisco.com/en/US/products/ps7271/prod_installation_guides_list.html>

### Cisco UCS C200 High-Density Rack-Mount Server Support

Cisco Unified Communications Manager Releases 8.5(1) and later support the Cisco UCS C200 High-Density Rack-Mount Server. This server runs Cisco Unified Communications Manager through VMware ESXi only.

For more information, refer to this document's [Platform](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_5_1/delta/delta.html#wp1889181) section and to the Unified Communications Virtualization wiki at <http://docwiki.cisco.com/wiki/Unified_Communications_Virtualization>.

### Cisco UCS C210 Rack-Mount Server Support

Cisco Unified Communications Manager Releases 8.0(3) and later support the Cisco UCS C210 General-Purpose Rack-Mount Server, Reference Configurations 2 and 3. This server runs Cisco Unified Communications Manager through VMware ESXi only.

For more information, refer to this document's [Platform](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_5_1/delta/delta.html#wp1889181) section and to the Unified Communications Virtualization wiki at <http://docwiki.cisco.com/wiki/Unified_Communications_Virtualization>.

### Cisco Unified Communications Manager Assistant Browser Changes

Cisco Unified Communications Manager Assistant administration (using Cisco Unified Communications Manager Administration) and the Assistant Console are supported on Microsoft Internet Explorer (IE) 5.5 or later, Firefox 3.x or later, and Safari 4.x or later.

### Early Offer

**Description**

To enhance interoperability with third party SIP devices, Cisco Unified Communications Manager now allows you to configure SIP trunks to enable early offer for outgoing voice and video calls without requiring MTP, when media capabilities and media port information of the calling endpoint is available. For outgoing call setup for an early offer trunk, Cisco Unified Communications Manager includes an SDP with the calling device media port, codecs, and IP address of the calling device (when available); inserts an MTP for early offer only when the media information for the caller is unavailable; and advertises multiple codecs when an MTP that supports multiple codecs gets inserted. In previous releases, Cisco Unified Communications Manager only provided early offer SDP when administrators enabled MTP Required or E2E RSVP on the outgoing SIP trunk. The early offer feature enhancement in this release ensures that a higher percentage of outbound early offer SIP trunk calls get made without requiring an MTP, thus reducing the number of MTP resources needed and improving interoperability with third party PBXs.

Cisco Unified Communications Manager supports early offer (without requiring MTP) when the call gets initiated from one of the following devices:

•SIP phones



•SCCP phones with SCCP v20 support (which provides media port information through the getPort capability)



•MGCP gateways



•Incoming H323 fast start calls



•Incoming early offer SIP trunk calls



**Note** For the endpoints where the media port information is not available (e.g. H323 slow start calls or delayed offer SIP calls or legacy SCCP phones), Cisco Unified Communications Manager still allocates an MTP in order to provide SDP in the initial INVITE.



**Note** For calls initiated from any of the previous devices, MTP might be needed due to other reasons such as DTMF/codec mismatch, TRP required on the inbound or outbound trunk, or MTP required on calling side.



For this release, Cisco Unified Communications Manager also enhances interoperability with third party devices during mid-call operations including basic hold/resume operations and during supplementary services, such as transfer and conference. In previous releases, Cisco Unified Communications Manager sent an INVITE with an inactive SDP (a=inactive attribute) to indicate a break in media path, sent a Delayed Offer INVITE to insert music on hold or resume the media stream, and expected a send-recv offer SDP in the 200 OK. Because third party devices often provide an inactive offer SDP in the 200 OK instead of providing a send-recv offer SDP, the media path would remain in an inactive state and cause calls to drop. In this release, Cisco Unified Communications Manager allows you to configure a parameter for an early offer SIP trunk so that Cisco Unified Communications Manager suppresses the sending of inactive or sendonly SDP in mid-call INVITEs. When this parameter gets enabled, Cisco Unified Communications Manager connects the SIP Trunk device directly to the MOH or annunciator device without breaking the existing media stream during call hold or other feature invocation. Similarly, Cisco Unified Communications Manager connects the SIP Trunk device to a line side device directly during call resume without breaking the MOH or annunciator stream. By preventing the far end media stream from getting set to inactive, Cisco Unified Communications Manager should always be able to resume the media path.



**Note** You should only configure the suppression of inactive or sendonly SDP if you have interoperability issues with third party SIP devices during hold-resume or media resumption for supplementary services. Certain endpoints such as Cisco Unity Connection may not work if you enable this configuration.



In this release, Cisco Unified Communications Manager also provides a send-receive SDP in response to a delayed offer invite in an initial call or mid-call on a SIP trunk if the device connecting to SIP trunk supports GetPort capability. Cisco Unified Communications Manager provides this functionality whether or not the SIP trunk has been configured for early offer. If the device does not support the GetPort capability, Cisco Unified Communications Manager does not insert another MTP to provide a send-receive offer.

### Agent Greeting

Agent Greeting allows an agent or administrator to create and play a prerecorded greeting automatically at the beginning of a call, such as a customer call, before the agent begins the conversation with the caller. An Agent can prerecord a single greeting or multiple ones as needed and create and update them. When a customer calls, both callers hear the prerecorded greeting. The agent can remain on mute until the greeting ends or answer the call over the greeting. All codecs supported for the phone are supported for Agent Greeting calls.

To enable Agent Greeting in the Cisco Unified Communications Manager Administration application, choose **Device > Phone**, locate the Cisco Unified IP Phone that you want to configure. Scroll to the Device Information Layout pane and set Built-in Bridge to **On** or **Default**. If Built-in Bridge is set to Default, in the Cisco Unified Communications Manager Administration application, choose **System > Service Parameter** and select the appropriate Server and Service. Scroll to the Clusterwide Parameters (**Device - Phone**) pane and set Built-in Bridge Enable to **On**.

**For More Information**

•Agent Greeting feature documentation, *CTI Product Description Guide for Cisco Unified Contact Center Enterprise, Release 8.5(1)*



•Features Supported by Cisco Unified JTAPI, *Cisco Unified JTAPI Developers Guide for Cisco Unified Communications Manager 8.5(1)*



•Features Supported by TSP, *Cisco Unified TAPI Developers Guide for Cisco Unified Communications Manager 8.5(1)*



### Assisted Directed Call Park Support Enhancements

Cisco Unified Communications Manager 8.5(1) supports Assisted Directed Call Park on the Cisco Unified IP Phones 7900 Series (SIP).

Assisted directed call park is supported on all Cisco Unified IP Phones 7900, 8900, and 9900 series that support SIP. Assisted directed call park means that the end user needs to press only one button to direct-park a call. This requires you to configure a BLF Directed Call Park button. Then, when the user presses an idle BLF Directed Call Park feature button for an active call, the active call will be immediately parked at the Dpark slot associated with the Directed Call Park feature button.

### Call Back No Answer Support for VG224

Cisco Unified Communications Manager Release 8.0 and earlier supported Call Back only on busy subscriber for Cisco VG224 endpoints. Cisco Unified Communications Manager Release 8.5 and later supports Call Back No Answer for Cisco VG224 endpoints.

For more information on the Call Back feature, see the *Cisco Unified Communications Manager Features and Services Guide*.

### Cisco Mobile 8.0 Support

Cisco Unified Communications Manager supports SIP-based dual-mode mobile phones with Cisco Mobile 8.0. Cisco Unified Communications Manager supports the new Cisco Dual Mode for iPhone device type for iPhone, which specifies a SIP-based dual-mode mobile phone that is capable of leveraging VoIP connectivity over the enterprise WLAN.

Cisco Unified Communications Manager supports dual-mode mobile phones that use the Cisco Unified Mobile Communicator client and SIP protocol within the WLAN. Cisco Unified Communications Manager must handle dual SIP registrations (one from Cisco Unified Mobile Communicator via Cisco Unified Mobility Advantage and one from Wi-Fi as a SIP endpoint).

Cisco Mobile 8.0 provides iPhone users with voice over IP (VoIP) calling, visual voicemail, and access to the corporate directory while users are connected to the corporate network over Wi-Fi, either on premises or over VPN. Cisco Mobile 8.0 specifies an IP telephony endpoint that associates with Cisco Unified Communications Manager.



**Note** Cisco Mobile 8.0 is distinct from the Cisco Mobile application that runs in conjunction with a Cisco Unified Mobility Advantage server.



In order for Cisco Unified Communications Manager to support Cisco Mobile 8.0, Cisco Unified Communications Manager administrators must take at least the following step:

**1.** Configure the new device in Cisco Unified Communications Manager Administration.



**Note** Note that in Release 8.5(1) of Cisco Unified Communications Manager Administration, downloading of a COP file is not required.



The Administration Guide for Cisco Mobile 8.0 for iPhone provides the details of the complete configuration that is required to configure Cisco Mobile 8.0, including the steps that must be performed in Cisco Unified Communications Manager Administration. Refer to the document at the following URL:

<http://www.cisco.com/en/US/products/ps7271/prod_installation_guides_list.html>

### Cisco UCS C200 High-Density Rack-Mount Server Support

Cisco Unified Communications Manager Releases 8.5(1) and later support the Cisco UCS C200 High-Density Rack-Mount Server. This server runs Cisco Unified Communications Manager through VMware ESXi only.

For more information, refer to this document's [Platform](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_5_1/delta/delta.html#wp1889181) section and to the Unified Communications Virtualization wiki at <http://docwiki.cisco.com/wiki/Unified_Communications_Virtualization>.

### Cisco UCS C210 Rack-Mount Server Support

Cisco Unified Communications Manager Releases 8.0(3) and later support the Cisco UCS C210 General-Purpose Rack-Mount Server, Reference Configurations 2 and 3. This server runs Cisco Unified Communications Manager through VMware ESXi only.

For more information, refer to this document's [Platform](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_5_1/delta/delta.html#wp1889181) section and to the Unified Communications Virtualization wiki at <http://docwiki.cisco.com/wiki/Unified_Communications_Virtualization>.

### Cisco Unified Communications Manager Assistant Browser Changes

Cisco Unified Communications Manager Assistant administration (using Cisco Unified Communications Manager Administration) and the Assistant Console are supported on Microsoft Internet Explorer (IE) 5.5 or later, Firefox 3.x or later, and Safari 4.x or later.

### Early Offer

**Description**

To enhance interoperability with third party SIP devices, Cisco Unified Communications Manager now allows you to configure SIP trunks to enable early offer for outgoing voice and video calls without requiring MTP, when media capabilities and media port information of the calling endpoint is available. For outgoing call setup for an early offer trunk, Cisco Unified Communications Manager includes an SDP with the calling device media port, codecs, and IP address of the calling device (when available); inserts an MTP for early offer only when the media information for the caller is unavailable; and advertises multiple codecs when an MTP that supports multiple codecs gets inserted. In previous releases, Cisco Unified Communications Manager only provided early offer SDP when administrators enabled MTP Required or E2E RSVP on the outgoing SIP trunk. The early offer feature enhancement in this release ensures that a higher percentage of outbound early offer SIP trunk calls get made without requiring an MTP, thus reducing the number of MTP resources needed and improving interoperability with third party PBXs.

Cisco Unified Communications Manager supports early offer (without requiring MTP) when the call gets initiated from one of the following devices:

•SIP phones



•SCCP phones with SCCP v20 support (which provides media port information through the getPort capability)



•MGCP gateways



•Incoming H323 fast start calls



•Incoming early offer SIP trunk calls



**Note** For the endpoints where the media port information is not available (e.g. H323 slow start calls or delayed offer SIP calls or legacy SCCP phones), Cisco Unified Communications Manager still allocates an MTP in order to provide SDP in the initial INVITE.



**Note** For calls initiated from any of the previous devices, MTP might be needed due to other reasons such as DTMF/codec mismatch, TRP required on the inbound or outbound trunk, or MTP required on calling side.



For this release, Cisco Unified Communications Manager also enhances interoperability with third party devices during mid-call operations including basic hold/resume operations and during supplementary services, such as transfer and conference. In previous releases, Cisco Unified Communications Manager sent an INVITE with an inactive SDP (a=inactive attribute) to indicate a break in media path, sent a Delayed Offer INVITE to insert music on hold or resume the media stream, and expected a send-recv offer SDP in the 200 OK. Because third party devices often provide an inactive offer SDP in the 200 OK instead of providing a send-recv offer SDP, the media path would remain in an inactive state and cause calls to drop. In this release, Cisco Unified Communications Manager allows you to configure a parameter for an early offer SIP trunk so that Cisco Unified Communications Manager suppresses the sending of inactive or sendonly SDP in mid-call INVITEs. When this parameter gets enabled, Cisco Unified Communications Manager connects the SIP Trunk device directly to the MOH or annunciator device without breaking the existing media stream during call hold or other feature invocation. Similarly, Cisco Unified Communications Manager connects the SIP Trunk device to a line side device directly during call resume without breaking the MOH or annunciator stream. By preventing the far end media stream from getting set to inactive, Cisco Unified Communications Manager should always be able to resume the media path.



**Note** You should only configure the suppression of inactive or sendonly SDP if you have interoperability issues with third party SIP devices during hold-resume or media resumption for supplementary services. Certain endpoints such as Cisco Unity Connection may not work if you enable this configuration.



In this release, Cisco Unified Communications Manager also provides a send-receive SDP in response to a delayed offer invite in an initial call or mid-call on a SIP trunk if the device connecting to SIP trunk supports GetPort capability. Cisco Unified Communications Manager provides this functionality whether or not the SIP trunk has been configured for early offer. If the device does not support the GetPort capability, Cisco Unified Communications Manager does not insert another MTP to provide a send-receive offer.

### EMCC Device Maintains Network Setting

When a user uses a phone in a visiting cluster to log into the user Extension Mobility profile, the phone inherits the default provisioning, network, and security settings (specifically, the configuration in the Product Specific Configuration Layout section of the Phone Configuration window) from the home cluster. This behavior may override local security and network settings that are in place in the visiting cluster. Some of the parameters have firmware defaults that the system administrator cannot change until a fix is provided.

### Interoperability with Cisco Unified Customer Voice Portal Release 7.0(2)

Cisco Unified Communications Manager Release 7.1(5) and later is compatible with Cisco Unified Customer Voice Portal Release 7.0(2).

### IP Phone Service Enhancements

There are two new IP Phone service categories available for Android in Cisco Unified Communications Manager 8.5(1):

•Web Link—This is a service category that you can add by specifying its URL in the IP Phone Services Configuration window. When provisioned, the service displays in the application menu of the device. When the user invokes the URL, it opens in the browser.



•Web Widget—This service category is a standalone W3C web widget application that can be installed separately.



### Mobility Feature Enhancements

**Updates to the "Cisco Unified Mobility" Chapter**

The following topics in the "Cisco Unified Mobility" chapter of the *C*isco Unified Communications Manager Features and Services Guide were updated with new information:

•Methods for Enabling and Disabling Mobile Connect



•Mobility Enterprise Feature Configuration



•Handoff Mobility Configuration



•Mobility Profile Configuration



**New "Cisco Mobile VoiP Clients" Chapter**

The new "Cisco Mobile VoiP Clients" chapter of the *C*isco Unified Communications Manager Features and Services Guide contains information and details about these new features:

•**Dial-via-Office (DVO) Optimization Settings for Toll Reduction**—This feature supports a pre-configured policy to determine which mobile origination call (DVO-R or DVO-F) yields the least cost to the enterprise; this determination is typically based on locations. This feature benefits the mobile user by allowing the user to find the least cost when making a mobile call. The DNIS pool provides a list of Direct Inward Dialing (DID) numbers so that the user, if roaming, can choose a non-international number for the mobile call. Least cost routing negotiates with Cisco Unified Communications Manager to determine whether DVO-R or DVO-F generates the least cost, then chooses the less costly method for making the call.



This feature supports a pre-configured policy to determine which mobile origination call (DVO-R or DVO-F) yields the least cost to the enterprise; this determination is typically based on locations. This feature benefits the mobile user by allowing the user to find the least cost when making a mobile call. The DNIS pool provides a list of Direct Inward Dialing (DID) numbers so that the user, if roaming, can choose a non-international number for the mobile call. Least cost routing negotiates with Cisco Unified Communications Manager to determine whether DVO-R or DVO-F generates the least cost, then chooses the less costly method for making the call.

**Reasons for DVO Optimization Settings for Toll Reduction**

The following reasons make this feature desirable:

–Administrator can decide upon the DVO call type, DVO-F or DVO-R, for least cost call routing. In certain regions and with certain service providers, DVO-F can be more economical for mobile users; in other regions, DVO-R can be more economical. For example, in regions where incoming calls are free for mobile phone users, configuring a DVO-R call for mobile phone users achieves least cost call routing.



–Scalability—Multiple users in a given region can use a single mobility profile, which comprises region, service provider, location, and so forth. Here, "users" refers to the clients under actual end users. The administrator does not need to create a mobility profile for each end user.



–Single DID within a cluster for all DVO-F calls—For such DVO-F calls, the client makes an incoming call to Cisco Unified Communications Manager by using a particular DID.



–Multisite cluster—For a multisite cluster, a client in cluster A (such as the UK) uses the DID of cluster B (such as San Jose) for DVO-F calls, which incurs costs.



–DVO-R—Trunk allows calls that originate from a local DID. At times, when a client makes an outgoing DVO-R call, the client trunk may not allow an outgoing call if the caller ID does not lie in a specific range. For example, if a UK client invokes DVO-R, the callback call from the trunk at the San Jose cluster shows 408. When this call reaches the UK, the service provider trunk may not recognize the 408 and therefore not allow the call. Therefore, the caller IDs need to specify the local identifiable values.



**Characteristics of DVO Optimization Settings for Toll Reduction**

This feature involves the use of mobility profiles, which the administrator configures by using the **Call Routing > Mobility > Mobility Profile** menu path in Cisco Unified Communications Manager Administration.

DVO Optimization Settings for Toll Reduction do not change the alternate callback mechanism that DVO-R calls use: the client continues to control alternate callback.

•**Enable/Disable Mobile Connect From Mobile Phone**—This feature allows the Cisco Unified Mobile Communicator client to change the Mobile Connect status dynamically and keep the Mobile Connect Status between Cisco Unified Communications Manager and the client in sync. This feature provides the flexibility to the end user: the end user can change the user Mobile Connect status from the user mobile phone, not just from the GUI website.



Prior to Release 8.5(1) of Cisco Unified Communications Manager, Cisco Unified Communications Manager sent Mobile Connect status updates to the Cisco Unified Mobile Communicator client via Cisco Unified Mobility Advantage by AXL messages. Direct SIP messages between the Cisco Unified Mobile Communicator client and Cisco Unified Communications Manager now allow the client to change the client Mobile Connect status.

Beginning with Release 8.5(1) of Cisco Unified Communications Manager, the Cisco Unified Mobile Communicator client can update its Mobile Connect status directly. The following processing takes place:

–When the Cisco Unified Mobile Communicator registers with Cisco Unified Communications Manager for the first time, Cisco Unified Communications Manager sends a 200 OK Response message, then sends an out-of-dialog REFER message to Cisco Unified Mobile Communicator with the current Cisco Unified Mobile Communicator Mobile Connect status. This sequence takes place only for the first registration, not for any refresh registrations.



–Whenever the Mobile Connect status changes by means of any Cisco Unified Communications Manager interface, Cisco Unified Communications Manager immediately sends an out-of-dialog REFER message to the Cisco Unified Mobile Communicator client with the updated Mobile Connect status. The client can thus update the mobile phone display with the correct Mobile Connect status.



–If the Cisco Unified Mobile Communicator client changes the Mobile Connect status, the following processing takes place:



The Cisco Unified Mobile Communicator client sends an out-of-dialog REFER message to Cisco Unified Communications Manager with the changed Mobile Connect status.

Cisco Unified Communications Manager stores the changed Mobile Connect status in the database and notifies the other interested Cisco Unified Communications Manager components.

Cisco Unified Communications Manager also sends another out-of-dialog REFER message to the Cisco Unified Mobile Communicator client with the changed Mobile Connect status so that the client knows that the status has been updated.

•**Direct Connection From Cisco Unified Communications Manager to Mobile Client Without Proxy Server**—This feature provides server-side support for Cisco Unified Mobile Communicator clients to connect to Cisco Unified Communications Manager directly and thus eliminate Cisco Unified Mobility Advantage in the deployment. In previous releases, Cisco Unified Communications Manager connections to Cisco Unified Mobile Communicator required the services of a Cisco Unified Mobility Advantage server. This feature thereby reduces the necessary configuration and requires less equipment. Cisco Unified Communications Manager adjusts to support direct connection with the Cisco Unified Mobile Communicator.



Beginning in Release 8.5(1) of Cisco Unified Communications Manager, Cisco Unified Mobile Communicator registers directly with Cisco Unified Communications Manager and no longer needs to register with the Cisco Unified Mobility Advantage server.

Registration between the Cisco Unified Mobile Communicator client and Cisco Unified Communications Manager takes place over a separate TCP port. (The shared or pooled connection that was used by the Cisco Unified Mobility Advantage server is not used.) Keepalive messages between Cisco Unified Mobile Communicator and Cisco Unified Communications Manager remain the same as those passed between Cisco Unified Communications Manager and Cisco Unified Mobility Advantage. That is, keepalive messages occur with a 120 second refresh time. Cisco Unified Mobile Communicator registration with Cisco Unified Communications Manager introduces no new alarms, and registration takes place over the SIP channel without OWLP involvement.

If the client is running on the iPhone and Cisco Unified Mobile Communicator is unable to complete the SIP dialog, the Cisco Unified Communications Manager retains the PSTN call. (The PSTN call does not drop even if the SIP stat times out.) For example, if Cisco Unified Communications Manager does not receive an ACK message after it sends a 200 OK message, the PSTN call gets retained.

### Single Sign On

**Description**

The single sign on feature allows end users to log into Windows, then use the following Cisco Unified Communications Manager applications without signing on again:

•User Options Pages



•Cisco Unified Communication interface for Microsoft Office Communicator



### Unified Communication OS Upgrade

Cisco Unified Communications Manager Release 8.5(1) uses a new version of the operating system. There should be no noticeable change in the operation of Cisco Unified Communications Manager, other than in direct upgrade path support and older server support.

### Whisper Coaching

Silent call monitoring is a feature that allows a supervisor to discreetly listen to a conversation between an agent and a customer without allowing the agent to detect the monitoring session. Whisper coaching is an enhancement to silent call monitoring feature that allows supervisors to talk to agents during a monitoring session. This feature provides applications the ability to change the current monitoring mode of a monitoring call from Silent Monitoring to Whisper Coaching and vice versa.

To enable Whisper Coaching in the Cisco Unified Communications Manager Administration application, choose **Device > Phone**, locate the Cisco Unified IP Phone that you want to configure. Scroll to the Device Information Layout pane and set Built-in Bridge to **On** or **Default**. If Built-in Bridge is set to Default, in the Cisco Unified Communications Manager Administration application, choose **System > Service Parameter** and select the appropriate Server and Service. Scroll to the Clusterwide Parameters (**Device - Phone**) pane and set Built-in Bridge Enable to **On**.

### Windows 7 Support for RTMT

You can install the Real Time Monitoring Tool (RTMT), which works for resolutions 800\*600 and above, on a computer that is running Windows 98, Windows XP, Windows 2000, Windows Vista, Windows 7 (32-bit), Windows 7 (64-bit) WOW64, or Linux with KDE and/or Gnome client.

<http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/rel_notes/8_5_1/delta/delta.html>

## New and Changed Information for Cisco Unified Communications Manager Release 8.6(1)

\*\*this is a BIG one that many people have asked for over the years

It will allow the cucmadmin session to not time out after 20 minutes

**set webapp session timeout**

This command sets the time, in minutes, that can elapse before a web application, such as Unified CM Administration, times out and logs off the user.

For the new webapp session timeout setting to become effective, you must restart the Cisco Tomcat service. This command prompts you to restart the service.



**Caution** Restarting the Cisco Tomcat service ends all active sessions and can affect system performance. Cisco recommends that you only execute this command during off-peak traffic hours.



**Tip** Until you restart the Cisco Tomcat service, the show webapp session timeout command reflects the new values, but system will continue to use the old values.



**Note** This setting gets preserved through a software upgrade and does not get reset to the default value.



**Command Syntax**

**set webapp session timeout** *minutes*

**Parameters**

*minutes* specifies the time, in minutes, that can elapse before a web application times out and logs off the user.

•Value range: 5—99999 minutes



•Default value: 30 minutes



**Requirements**

Command privilege level: 1

Allowed during upgrade: No

Command privilege level: 1

Allowed during upgrade: No

And for the cli as well

**set cli session timeout**

This command sets the time, in minutes, after which an active CLI session times out and disconnects. Be aware that the new session timeout value becomes effective immediately for a new CLI session; however, existing sessions retain their original timeout value. Also the show cli session timeout command reflects the new value, even if the current session is not using it.



**Note** This setting gets preserved through a software upgrade and does not get reset to the default value.



**Command Syntax**

**set cli session timeout** *minutes*

**Parameters**

*minutes* specifies the time, in minutes, that can elapse before an active CLI session times out and disconnects.

•Value range: 5—99999 minutes



•Default value: 30 minutes



**Requirements**

Command privilege level: 1

Allowed during upgrade: No

**Called Party Trace**

**Description**

Called Party Trace allows you to configure a directory number or list of directory numbers that you want to trace. You can request on-demand tracing of calls by using the Session Trace Tool.

The Called Party Trace feature provides information about the calling party number in addition to the called party number within a node. You can use the information from each node to trace a call back to the originator.

**Unified CM Administration Configuration Tips**

Use the Advanced Features menu in Cisco Unified Communications Manager Administration to add DNs. Once the call log information has been generated, you can you can use the Real Time Monitoring Tool to view logs for specific DNs.

**GUI Changes**

No GUI changes exist for this feature.

**Service Parameter and Enterprise Parameter Changes**

No service or enterprise parameter changes exist for this feature.

**Installation/Upgrade (Migration) Considerations**

No special installation or upgrade considerations exist for this feature. After you install or upgrade to Unified CM 8.6(1), you can use this feature.

**Serviceability Considerations**



**Note** You must be an authorized administrator to access the directory number logs. To grant authorization to a specific role using Role Configuration, the "Called Party Tracing" resource must have read permission enabled for the role.



Procedure

To access the DN Trace report in the Real Time Monitoring Tool (RTMT), follow these steps:

**1.** From the RTMT menu, choose **CallManager > Callprocess > Called Party Trace**. Or, Click the **CallManager** tab; then, click **Called Party Trace**.



**2.** Choose the start time of the report using the drop-down box list.



**Note** The start time cannot be older than five years from the current date.



**3.** The report shows the following information



–Start time



–Calling directory number



–Original called directory number



–Called directory number



–Calling device name



–Called device name



**Note** When 5 megabytes of trace file entries have been written to the log files being accessed by RTMT, the oldest log information is overwritten by new trace entries as they are recorded. The RTMT will only list a maximum of 500 entries for any given search.



### Additional Device Types

Two dual mode device types are have been made available as built-in. The Nokia S60 and Cisco Dual Mode for Android no longer require external COP files to enable these two device types.

**GUI Changes**

When adding a phone, both **Nokia S60** and **Cisco Dual Mode for Android** devices will show up in the **Phone Type** drop-down list.

**Service Parameter and Enterprise Parameter Changes**

No service or enterprise parameter changes exist for this feature.

**Installation/Upgrade (Migration) Considerations**

No special installation or upgrade considerations exist for this feature. After you install or upgrade to Unified CM 8.6(1), you can use this feature.

**Serviceability Considerations**

No serviceability considerations exist for this feature.

**BAT Considerations**

No BAT considerations exist for this feature.

**CAR/CDR Considerations**

No CAR or CDR considerations exist for this feature.

**Security Considerations**

No security considerations exist for this feature.

**AXL and CTI Considerations**

No AXL or CTI considerations exist for this feature.

**User Tips**

No user tips exist for this feature.

### Single Sign On

**Description**

The single sign on feature allows end users to log into a Windows client machine on a Windows domain, then use certain Unified CM applications without signing on again. For this release, the single sign on feature has been extended to include the Cisco Unified Communications Manager Administration application and the Real Time Monitoring Tool (RTMT) application in addition to the User Options pages and the Cisco Unified Communication interface for Microsoft Office Communicator that were previously supported.

**Unified CM Administration Configuration Tips**

Use the following procedure to configure the Single Sign On feature:

**1.** Ensure that your environment meets the requirements described in the "Single Sign On" section of the *Cisco Unified Communications Manager Features and Services Guide 8.6(1)*. Refer to this document online at:  
  
[http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/admin/8\_6\_1/ccmfg/bccm-861-cm.html.](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsgd-861-cm.html)



**2.** Provision the OpenAM server in Active Directory, then generate keytab files.



**Note** If your Windows version does not include the ktpass tool for generating keytab files, then you must obtain it separately.



**3.** Import the OpenAM server certificate into the Cisco Unified Communications Manager tomcat-trust store.



**4.** Configure Windows single sign on with Active Directory and OpenAM.



**5.** (For Unified CM Administration only) Verify that the user is provisioned in the Active Directory.



**6.** (For Unified CM Administration only) Synchronize the user data to the Cisco Unified Communications Manager database using the DirSync service.



**7.** (For Unified CM Administration only) Add the user to the CCM Super User group to enable access to Cisco Unified Communications Manager Administration.



**8.** Configure client browsers for single sign on.



**9.** Enable single sign on in Cisco Unified Communications Manager.



**Note** For more detailed information about this procedure, see the *Cisco Unified Communications Manager Features and Services Guide 8.6(1)*. Refer to this document online at [http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/admin/8\_6\_1/ccmfg/bccm-861-cm.html.](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsgd-861-cm.html)



**GUI Changes**

No GUI changes exist for this feature.

**Service Parameter and Enterprise Parameter Changes**

No service or enterprise parameter changes exist for this feature.

**Installation/Upgrade (Migration) Considerations**

No special installation or upgrade considerations exist for this feature. After you install or upgrade to Unified CM 8.6(1), you can use this feature.

**Serviceability Considerations**

No serviceability considerations exist for this feature.

**BAT Considerations**

No BAT considerations exist for this feature.

**CAR/CDR Considerations**

No CAR or CDR considerations exist for this feature.

**Security Considerations**

No security considerations exist for this feature.

**AXL and CTI Considerations**

No AXL or CTI considerations exist for this feature.

**User Tips**

No user tips exist for this feature.

### Cisco Unified IP Phone 8941 and 8945

The Cisco Unified IP Phone 8941 and 8945 are new, easy-to-use IP Phones that provide high-quality voice services over IP. The phones offer a variety of features including:

•Integrated camera



•Color graphics display



•Full duplex speakerphone



•Rich media support



•Power over Ethernet (PoE)—IEEE 802.3af Class 1 (Cisco Unified IP Phone 8941) and IEEE 802.3af Class 2 (Cisco Unified IP Phone 8945)



•Built-in Gigabit Ethernet Switch (Cisco Unified IP Phone 8945 only)



•Bluetooth headset (Cisco Unified IP Phone 8945 only)



**Assisted Directed Call Park**

The Assisted Directed Call Park feature enables users to park a call by pressing only one button using the Direct Park feature. This feature requires administrators to configure a Busy Lamp Field (BLF) Assisted Directed Call Park button. When users press an idle BLF Assisted Directed Call Park button for an active call, the active call is parked at the Direct Park slot associated with the Assisted Directed Call Park button.

Support for this feature has been added to the following Cisco Unified IP Phones for Cisco Unified Communications Manager 8.6(1) and later:

•Cisco Unified IP Phone 6921 (SIP)



•Cisco Unified IP Phone 6941 (SIP)



•Cisco Unified IP Phone 6945 (SIP)



•Cisco Unified IP Phone 6961 (SIP)



**Where to Find More Information**

•Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager 8.6(1)



•Cisco Unified IP Phone User Guide for Cisco Unified Communications Manager 8.6(1)



•Cisco Unified Communications Manager Features and Services Guide



**Classic Ringtones**

The Classic Ringtones feature supports 29 ring tones: 2 embedded in the phone firmware and 27 downloaded from the Cisco Unified Communications Manager. The feature makes the available ring tones common with other Cisco Unified IP Phones.

The following phone models support the Classic Ringtones feature:

•Cisco Unified IP Phone 6921 (SCCP)



•Cisco Unified IP Phone 6941 (SCCP)



•Cisco Unified IP Phone 6945 (SCCP)



•Cisco Unified IP Phone 6961 (SCCP)



**Where to Find More Information**

Cisco Unified Communications Manager Features and Services Guide

**CME Version Negotiation**

The Cisco Unified Communications Manager Express (Unified CME) Version Negotiation feature supports a SIS version in the supported tag. The Cisco Unified IP Phones use the supported tag to interact with Cisco Unified Communications Manager Express and its supported SIS version.

The following phone models support the CME Version Negotiation feature:

•Cisco Unified IP Phone 6901 (SIP)



•Cisco Unified IP Phone 6911 (SIP)



•Cisco Unified IP Phone 6921 (SIP)



•Cisco Unified IP Phone 6941 (SIP)



•Cisco Unified IP Phone 6945 (SIP)



•Cisco Unified IP Phone 6961 (SIP)



**Where to Find More Information**

Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager 8.6(1)

**EnergyWise**

Cisco EnergyWise program promotes company-wide sustain ability by monitoring, reporting, and reducing energy consumption across an entire corporate infrastructure. In the Cisco Unified IP Phone firmware, the EnergyWise feature allows phones to participate in an EnergyWise-enabled system. The phones can report power usage to the EnergyWise domain to allow the tracking and control of power within the customer premise.

In the Cisco Unified IP Phones, the EnergyWise feature enables the phone to sleep (power down) and wake (power up). A sleeping phone reduces energy consumption, typically into the 0 to 1 watt range. The administrator sets a working schedule of days, power up times, and power down times for each phone. At the scheduled power down time, the phone automatically powers down, and at the scheduled power up time, the phone automatically powers up.

The following Cisco Unified IP Phones support EnergyWise in this release:

•Cisco Unified IP Phone 6901 (SCCP)



•Cisco Unified IP Phone 6911 (SCCP)



•Cisco Unified IP Phone 6921 (SCCP)



•Cisco Unified IP Phone 6941 (SCCP)



•Cisco Unified IP Phone 6945 (SCCP)



•Cisco Unified IP Phone 6961 (SCCP)



•Cisco Unified IP Phone 7906 (SCCP)



•Cisco Unified IP Phone 7911 (SCCP)



•Cisco Unified IP Phone 7931 (SCCP)



•Cisco Unified IP Phone 7941 (SCCP)



•Cisco Unified IP Phone 7945 (SCCP)



•Cisco Unified IP Phone 7961 (SCCP)



•Cisco Unified IP Phone 7962 (SCCP)



•Cisco Unified IP Phone 7965 (SCCP)



•Cisco Unified IP Phone 7970 (SCCP)



•Cisco Unified IP Phone 7971 (SCCP)



•Cisco Unified IP Phone 7975 (SCCP)



•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



**Where to Find More Information**

•Cisco Unified IP Phone Administration Guide for Cisco Unified Communications Manager 8.6(1)



•Cisco Unified IP Phone User Guide for Cisco Unified Communications Manager 8.6(1)



**EnergyWise in the Cisco Unified IP Phone 7900 Series**

The Cisco Unified IP Phone 7900 series can be configured to automatically sleep and wake at specific times. When these phones are sleeping, users cannot wake them up.

**EnergyWise in the Cisco Unified IP Phone 6900, 8900, and 9900 Series**

The Cisco Unified IP Phone 6900, 8900, and 9900 series support EnergyWise by using configured sleep and wake times. In addition, users can wake a sleeping phone using the Select button.This feature allows the phone to participate in an EnergyWise enabled system. The phone reports its power usage to a EnergyWise compliant switch to allow the tracking and control of power within the customer premise. The phone provides alternate reduced power modes including an extremely low, off mode. The Unified CM administrator configures and exclusively manages the phones power state through vendor specific configuration on the Unified CM Admin pages.

When the phone turns off power after negotiation with an EnergyWise switch, it unregisters from Unified CM and enters Deep Sleep/PowerSavePlus mode.

For Cisco Unified IP Phone 9900 and 6900 Series, press the Select button on the phone to wake up the phone from the Deep Sleep/PowerSavePlus mode, but there is no way to register Cisco Unified IP Phone 7900 Series back to the Unified CM during Deep Sleep. This is the limitation for the Cisco Unified IP Phone 7900 Series. However, both types of phones automatically re-register with the Unified CM once the Deep Sleep mode configured PowerON time occurs. You configure Deep Sleep mode on the Device page of the Unified CM. Configure Deep Sleep mode for the phones at least 10 minutes before the actual power off time to allow the information to synchronize between the switch and the phone.

The configured power off idle timer enables only in the case when there is physical interaction on the phone. If you do not physically interact with the phone such as call disconnection using an application, then the power idle timer defaults to 10 minutes.

**Enhanced Call Forward Notification**

The Enhanced Call Forward Notification feature provides additional call information to display in the notification window when a call forwards. This additional information includes the name or number of phone that forwarded the call. The type of information displayed is set by the system administrator.

The Enhanced Call Forward Notification feature is supported on the following phones:

•Cisco Unified IP Phone 6921 (SCCP)



•Cisco Unified IP Phone 6941 (SCCP)



•Cisco Unified IP Phone 6945 (SCCP)



•Cisco Unified IP Phone 6961 (SCCP)



•Cisco Unified IP Phone 8961 (SIP)



•Cisco Unified IP Phone 9951 (SIP)



•Cisco Unified IP Phone 9971 (SIP)



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**Forced Authentication Code and Client Matter Code Support**

The Forced Authentication Code (FAC) and Client Matter Code (CMC) Support features extends the FAC and CMC features to additional Cisco Unified IP Phones.

•FAC controls the types of calls that certain users can place. When placing a call, a user receives a prompt to enter a valid authorization code before the call is made.



•CMC enables a user to specify that a call relates to a specific client matter. When placing a call, a user can enter a code to indicate the type of call being placed (for example, to a specific customer).



The Forced Authentication Code and Client Matter Code Support feature is now supported on the following phones:

•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



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•Cisco Unified Communications Manager Features and Services Guide



**HTTP Download**

The HTTP Download feature enhances the file download process to the phone. By default, the phone uses HTTP. If the HTTP download fails, the phone reverts to using the TFTP download.

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



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**Missed Call Logs**

The Missed Call Logs feature allows a user to specify whether missed calls are logged in the missed calls directory for a given line appearance.

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



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**Multiple Calls per Line Appearance**

The Multiple Calls Per Line feature supports multiple calls for each line. By default, your phone supports two active calls per line, and a maximum of six active calls per line. You can adjust this number of active calls (not exceeding six calls) according to your need using the Cisco Unified Communications Manager Assistant. Only one call can be connected at any time; other calls are automatically placed on hold.

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



**Limitations**

**1.** The system supports up to a maximum of 6 calls per line.



**2.** For phones with SCCP, the Cisco Unified Communications Manager must be running Release 8.6 or later to support the feature.



**Where to Find More Information**

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•Cisco Unified IP Phone User Guide for Cisco Unified Communications Manager 8.6(1)



•Cisco Unified Communications Manager Features and Services Guide



**Next Generation Power over Ethernet**

The Next Generation Power over Ethernet (NGPoE+) feature enhances the ability of the phones to exceed the industrial standard IEEE 802.at. NGP0E+ provides up to 60 Watts (PSE) or 51 Watts (PD) to phones. On the Cisco Unified IP Phone side, the models support a maximum of 50.333 Watts (PSE) or 44 Watts (PD).

The phones self-adapt to the increased power. No user configuration is required.

This feature is supported on the following Cisco Unified IP Phones:

•Cisco Unified IP Phone 8951



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



**PLKs as Softkeys**

The PLKs as Softkeys feature enables you to provide certain features to users as either softkeys or programmable line buttons on the phone.

The following features are now available as softkeys:

•Call PickUp



•Other Call PickUp



•Group Call PickUp



•Mobility



•Malicious Call Trace



•Meet Me



•Quality Reporting



This feature is supported on the following Cisco Unified IP Phones:

•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



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**SSH Access**

The SSH Access settings option allows you to enable or disable the SSH port on the phone using Unified CM Administration. When enabled, the option allows the phone to accept the SSH connections. Disabling the SSH server functionality of the phone blocks the SSH access to the phone. This setting is disabled by default.

This feature is supported on the following Cisco Unified IP Phones:

•Cisco Unified IP Phone 6901 (SCCP and SIP)



•Cisco Unified IP Phone 6911 (SCCP and SIP)



•Cisco Unified IP Phone 6921 (SCCP and SIP)



•Cisco Unified IP Phone 6941 (SCCP and SIP)



•Cisco Unified IP Phone 6945 (SCCP and SIP)



•Cisco Unified IP Phone 6961 (SCCP and SIP)



•Cisco Unified IP Phone 7906G (SCCP and SIP)



•Cisco Unified IP Phone 7911G (SCCP and SIP)



•Cisco Unified IP Phone 7931G (SCCP and SIP)



•Cisco Unified IP Phone 7941G (SCCP and SIP)



•Cisco Unified IP Phone 7941G-GE (SCCP and SIP)



•Cisco Unified IP Phone 7942G (SCCP and SIP)



•Cisco Unified IP Phone 7945G (SCCP and SIP)



•Cisco Unified IP Phone 7961G (SCCP and SIP)



•Cisco Unified IP Phone 7961G-GE (SCCP and SIP)



•Cisco Unified IP Phone 7962G (SCCP and SIP)



•Cisco Unified IP Phone 7965G (SCCP and SIP)



•Cisco Unified IP Phone 7970G (SCCP and SIP)



•Cisco Unified IP Phone 7975G (SCCP and SIP)



•Cisco Unified IP Phone 8961



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



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**Toast Timer**

The Toast Timer feature controls the time that the Call Notification Pop-up Window (toast) remains visible for an incoming call. You select a time for all phones. The user cannot alter the timer.

This feature is supported on the following Cisco Unified IP Phones:

•Cisco Unified IP Phone 8951



•Cisco Unified IP Phone 9951



•Cisco Unified IP Phone 9971



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http://www.cisco.com/en/US/docs/voice\_ip\_comm/cucm/rel\_notes/8\_6\_1/delta/delta.html#wp2139571