Configuring Conferencing

Last Updated: May 16, 2011

This chapter describes the conferencing support in Cisco Unified Communications Manager Express (Cisco Unified CME).

Finding Feature Information in This Module

Your Cisco Unified CME version may not support all of the features documented in this module. For a list of the versions in which each feature is supported, see the "Feature Information for Conferencing" section on page 1000.

Contents

Restrictions for Conferencing

When you are configuring dial peers or ephone-dns, including park slots and conferencing extensions, on Cisco Integrated Services Router Voice Bundles, the following message may appear to warn you that free memory is not available:

%DIALPEER_DB-3-ADDPEER_MEM_THRESHOLD: Addition of dial-peers limited by available memory

To configure more dial peers or ephone-dns, increase the DRAM in the system. A moderately complex configuration may exceed the default 256 MB DRAM and require 512 MB DRAM. Note that many factors contribute to memory usage, in addition to the number of dial peers and ephone-dns configured.

Information About Conferencing

To enable conferencing, you should understand the following concepts:

Conferencing Overview

Conferencing allows you to join three or more parties in a telephone conversation. Two types of conferencing are available in Cisco Unified CME: ad hoc and meet-me.

Ad hoc conferences can be hardware-based or software-based. Software-based conferences use the router CPU to provide audio mixing (G.711) and are limited to 3 parties. Hardware-based multi-party ad hoc conferencing uses digital signal processors (DSPs) to allow more parties than software-based ad hoc conferencing and also provides additional features such as Join and Conference Participant List (ConfList).

Meet-me conferences are created by parties calling a designated conference number. Meet-me conferencing is hardware-based only. If you configure software-based conferencing, you cannot have meet-me conferences.

Conferencing with Octo-Lines

In Cisco Unified CME 4.3 and later versions, when a conference initiator is an octo-line directory number, Cisco Unified CME selects an idle channel from that directory number and the user must establish a new call to complete the conference. If an idle channel is not available on the same octo-line directory number, the conference aborts and a "No Line Available" message displays.
Cisco Unified CME does not select an idle channel from another directory number and the user cannot select "hold" calls on the other channels of the directory number or other directory numbers, which is the behavior for single-line and dual-line directory numbers.

With octo-line directory numbers, only one directory number is required for an 8-party meet-me or ad hoc conference. Up to eight select and join instances are supported.

Secure Conferencing Limitation

Cisco Unified CME cannot use the secure conference DSP farm capability. If Cisco Unified CME needs a conference DSP farm resource for multiparty ad hoc or meet-me conferencing, it will use a secure or nonsecure DSP farm resource depending on what resources have been registered with Cisco Unified CME. If Cisco Unified CME happens to pick a secure DSP farm resource, the conference itself will not be secure, which is a waste, in terms of sessions capacity, of the more expensive secure DSP farm resource.

To avoid using valuable secure DSP farm resources, we recommend that you do not register a secure conference DSP Farm profile to a Cisco Unified CME because Cisco Unified CME cannot use the DSP farm's secure capabilities.

Ad Hoc Conferencing

Before Cisco Unified CME 4.1, support for conferencing is limited to three-party ad hoc conference calls using a G.711 codec. To have an ad hoc conference with a party that is not using a G.711 codec, transcoding is necessary. For more information, see the "Transcoding When a Remote Phone Uses G.729r8" section on page 453.

The maximum number of simultaneous conferences is platform-specific to the type of Cisco Unified CME router, and each individual Cisco Unified IP phone can host a maximum of one conference at a time. You cannot create a second conference on a phone if you already have an existing conference on hold.

Conference Gain Levels

In Cisco Unified CME 3.3 and later versions, you can adjust the gain level of an external call to provide more adequate volume. This functionality is applied to inbound audio packets so that conference participants can more clearly hear a remote PSTN or VoIP caller joining their call. Note that this functionality cannot discriminate between a remote VoIP/foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP/IP phone, which does not require a volume gain and may therefore incur some sound distortions.

End-of-Conference Options

For Cisco CME 3.2 and later versions, a person who initiates a conference call and hangs up can either keep the remaining parties connected or disconnect them.

Cisco Unified IP phones can be configured to keep the remaining conference parties connected when the conference initiator hangs up (places the handset back in the on-hook position). Conference originators can disconnect from their conference calls by pressing the Confrn (conference) soft key. When an initiator uses the Confrn key to disconnect from the conference call, the oldest call leg will be put on hold, leaving the initiator connected to the most recent call leg. The conference initiator can then navigate between the two parties by pressing either the Hold soft key or the line buttons to select the desired call.

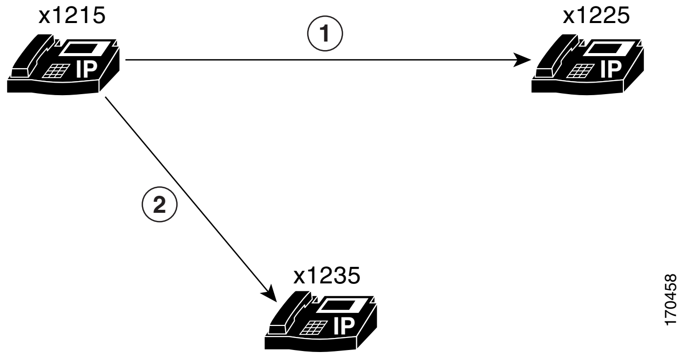
In Cisco Unified CME 4.0 and later versions, behavior for the end of three-way conferences can be configured at a phone level. The options specify whether the last party that joined a conference can be dropped from the conference and whether the remaining two parties should be allowed to continue their connection after the conference initiator has left the conference.

Multi-Party Ad Hoc Conferencing for More Than Three Parties

In Cisco Unified CME 4.1 and later versions, hardware-based multi-party ad hoc conferences allow more than three parties. Ad hoc conferences are created when one party calls another, then either party decides to add another party to the call. Ad hoc conferences can be created in several ways.

The conference shown in Figure 30 is created when extension 1215 dials extension 1225. The two parties decide to add a third party, extension 1235. Extensions 1215, 1225, and 1235 are now parties in an ad hoc conference. Extension 1215 is the creator.

Figure 30 Simple Ad Hoc Conference Using the Conf Soft Key



You can configure ad hoc conferencing so that only the creator can add parties to the conference. The default is that any party can add other parties to the conference.

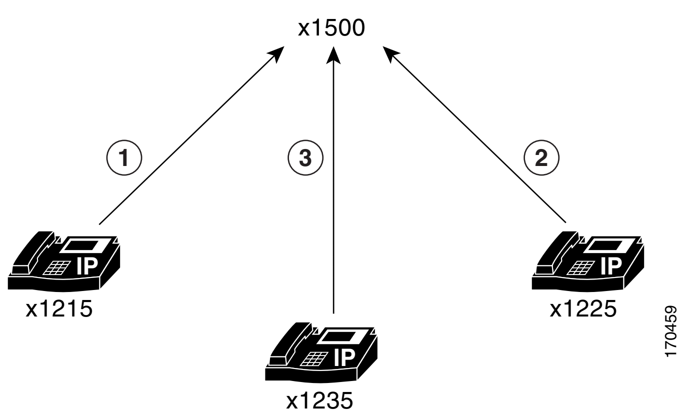
You can configure conferencing so that the conference drops when the creator hangs up, and you can configure it so that the conference drops when the last local party hangs up. The default is that the conference is not dropped, regardless of whether the creator hangs up, provided three parties remain in the conference.

For configuration information, see the "SCCP: Configuring Conferencing Options for a Phone" section on page 972 for more information.

Meet-Me Conferencing in Cisco Unified CME 4.1 and Later versions

In Cisco Unified CME 4.1 and later versions, meet-me conferences consist of at least three parties dialing a meet-me conference number predetermined by a system administrator. For example, the conference shown in Figure 31 is created when the conference creator at extension 1215 presses the MeetMe soft key and hears a confirmation tone, then dials the meet-me conference number 1500. Extension 1225 and extension 1235 join the meet-me conference by dialing 1500. Extensions 1215, 1225, and 1235 are now parties in a meet-me conference on extension 1500.

Figure 31 Simple Meet-Me Conference Scenario



Configuring Maximum Parties

You can configure the maximum number of conference parties to be lower than the actual maximum of 32 for meet-me conferences. See the "SCCP: Configuring the DSP Farm" section on page 965 for more information.

Freeing Conference Resources

If only one party remains in the meet-me conference, for example, if one party has forgotten to hang up, the conference call is disconnected after five minutes to free system resources.

If the creator is waiting for parties to join the conference and is the only party on the conference, the conference is not disconnected because significant resources are not being used.

Soft Keys for Conference Functions

In Cisco Unified CME 4.1 and later versions, the following soft keys provide conferencing functions for hard-ware based multi-party conferencing enhancements on your phone and require the appropriate DSP farm configuration. For configuration information, see the "SCCP: Configuring Multi-Party Ad Hoc and Meet-Me Conferencing in Cisco Unified CME 4.1 and Later Versions" section on page 961.

- ConfList—Conference list. Lists all parties in a conference. For multi-party ad hoc conferences, this soft key is available for all parties in a conference. For meet-me conferences, this soft key is available for the creator only. Press **Update** to update the list of parties in the conference, for instance, to verify that a party has been removed from the conference.
- Join—Joins an established call to an adhoc conference. You must first press **Select** to choose each connected call that you want to join in a conference, then press **Join** to join the selected calls to the conference.
- RmLstC—Remove last caller. Removes the last party added to the conference. This soft key works for the creator only.
- Select—Selects a call or conference to join to a conference and selects a call to remove from a conference. The creator can remove other parties by pressing the **ConfList** soft key, then use the **Select** and **Remove** soft keys to remove the appropriate parties.
- MeetMe—Initiates a meet-me conference. The creator presses this soft key before dialing the conference number. Other meet-me conference parties only dial the conference number to join the conference. This soft key must be configured before you can initiate meet-me conferences.

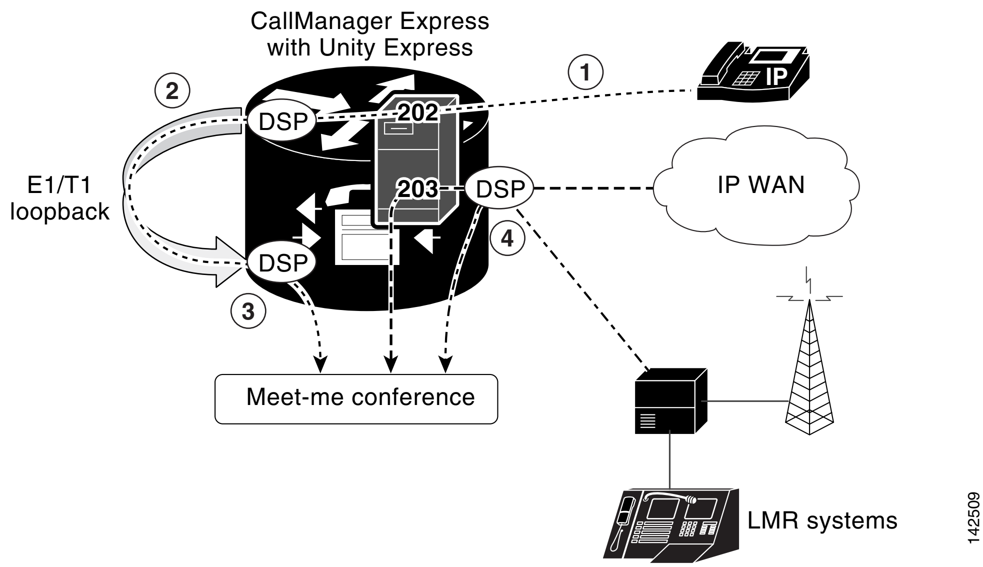
Meet-Me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0

Unlike the built-in Cisco Unified CME conference feature, a meet-me conference does not have a three-party limit. Meet-me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0 requires Cisco Unity Express auto-attendant to transfer callers to the correct Meet-Me bridge and a dual T-1/E-1 VWIC card for providing DSP resources. By default three Meet-Me bridge's with 8 callers each are defined with the maximum number of callers restricted by the number of DSP resources available in the Cisco router. A maximum of 96 callers in conference is supported. Multicast conferences can be accessed from IP phones, public switched telephone network (PSTN) callers, and Cisco Land Mobile Radio (LMR) devices connected to ear and mouth (E&M) voice ports on the Cisco Unified CME router.

The only limiting factor for this solution is the number of T1 or E1 loopback ports and digital-signal-processor (DSP) resources available.

Figure 32 illustrates the callflow for Meet-Me Conferencing on a Cisco router with Cisco CME 3.2 to Cisco Unified CME 4.0 and Cisco Unity Express. IP phones and PSTN callers dial into Cisco Unity Express Auto Attendant using separate access numbers. Cisco Unity Express Auto Attendant routes calls to a multicast conference based on which access number is called. In this example, local IP phones call 202 and PSTN users call 203 to dial into Cisco Unity Express.

Figure 32 Meet-Me Conference in Cisco CME 3.2 to Cisco Unified CME 4.0



1. In order to send or receive audio from a multicast conference, calls must pass through a DSP for audio mixing. By default, IP phone calls are not passed through a DSP. IP phone calls can be routed to T1 or E1 loopback, forcing the call to pass through a DSP. In this example, Cisco Unity Express routes callers who dialed 202, through the E1/T1 loopback.
2. The T1/E1 loopback ports are permanently trunked to the multicast conference. Incoming calls to T1 loopback are routed back to the multicast conference on Cisco CME.
3. All PSTN calls must pass through a DSP, so incoming PSTN calls do not have to be routed to T1 loopback. The Auto Attendant routes PSTN calls directly to the multicast conference. In this example, Cisco Unity Express routes callers who dialed 203 directly into the multicast conference.
4. Cisco LMR ports are permanently trunked into the multicast conference, so radio parties can listen to audio from both the IP phone and the PSTN. Pushing the “talk” button on a radio handset keys the M lead on the Cisco CME E&M port and the radio handset can transmit audio.

Note Cisco LMR devices typically cannot transmit and receive audio at the same time. If a Cisco LMR device receives audio from a multicast conference, it cannot transmit audio. In order for a Cisco LMR device to transmit audio to the conference, all IP phone and PSTN parties must be on mute so the LMR device does not receive any audio. If a single IP phone or PSTN device in the conference is transmitting audio, the individual using the Cisco LMR device cannot talk.

Dial Plan

Before configuring Cisco Unified CME and Cisco Unity Express, you should plan your dial plan for Meet-Me Conferencing. Table 28 lists the dial-plan parameters that must be defined before you can configure Meet-Me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0.

To prevent IP phones from dialing into the multicast bridge directly, the multicast bridge numbers should be set to nondialable numbers starting with an alphabetical character.

IP phones that dial into the multicast bridge cannot send or receive audio, so IP phone calls must be routed to the loopback number. These numbers are required to configure Cisco Unity Express Auto Attendant, which controls all access to the multicast bridge.

Table 28 Dial Plan for Support Meet-Me Conferencing

Parameter	Sample Number	Description
External Number	203	Number used by external callers from PSTN to dial into Cisco Unity Express Auto Attendant conference bridge.
Internal Number	202	Number used by internal callers from local IP phones to dial into Cisco Unity Express Auto Attendant conference bridge.
bridge1	212	Number used by Cisco Unified CME to route calls to E1 or T1 loopback that is trunked to multicast bridge 1.
bridge2	213	Number used by Cisco Unified CME to route calls to E1 or T1 loopback that is trunked to multicast bridge 2
bridge3	214	Number used by Cisco Unified CME to route calls to E1 or T1 loopback that is trunked to multicast bridge 3.
bridge1_pstn	A212	Nondialable number used by Cisco Unified CME to route calls into multicast bridge 1. Number should start with an alphabetical number.
bridge2_pstn	A213	Nondialable number used by Cisco Unified CME to route calls into multicast bridge 2. Number should start with an alphabetical number.
bridge3_pstn	A214	Nondialable number used by Cisco Unified CME to route calls into multicast bridge 3. Number should start with an alphabetical number.
operator	150	Number dialed if user needs assistance.

How to Configure Conferencing

This section contains the following tasks:

(Software-based) Three-Party Ad Hoc Conferencing

- Modifying the Default Configuration for Three-Party Ad Hoc Conferencing, page 956 (optional)
- SCCP: Configuring Conferencing Options on a Phone, page 957 (optional)
- SIP: Configuring Conferencing Options on a Phone, page 959 (optional)

(Hardware-based) Multi-Party Ad Hoc and Meet-Me Conferencing in Cisco Unified CME 4.1 and Later Versions

- SCCP: Configuring Multi-Party Ad Hoc and Meet-Me Conferencing in Cisco Unified CME 4.1 and Later Versions, page 961 (required)
- SCCP: Verifying Multi-Party Ad Hoc and Meet-Me Conferencing, page 975 (optional)

Meet-Me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0

- SCCP: Configuring Meet-Me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0, page 975 (required)

Modifying the Default Configuration for Three-Party Ad Hoc Conferencing

To globally modify the default configuration and change any of the following parameters for three-party ad hoc conferencing, perform the following steps.

- Maximum number of three-party conferences that are supported simultaneously by the Cisco Unified CME router. Maximum number of simultaneous three-party conferences supported by a router is platform-dependent. The default value is half of the maximum number.
- Increase the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call

Restrictions

- When a three-way conference is established, a participant cannot use call transfer to join the remaining conference participants to a different number.
- Three-party ad hoc conferencing does not support meet-me conferences.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **max-conferences** *max-conference-number* [**gain -6** | **0** | **3** | **6**]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**telephony-service** **Example:** `Router(config)#`	Enters telephony-service configuration mode.
Step 4	**max-conferences** *max-conference-number* [**gain -6** | **0** | **3** | **6**] **Example:** `Router(config-telephony)# max-conferences 6`	Sets the maximum number of simultaneous three-party conferences supported by the router. • *max-conference-number*—Maximum value is platform-dependent. Type **?** for maximum value. Default is half of the maximum value. • **gain**—(Optional) Amount to increase the sound volume of VoIP and PSTN calls joining a conference call, in decibels. Valid values are **-6**, **0**, **3**, and **6**. The default is **-6**.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Exits to privileged EXEC mode.

SCCP: Configuring Conferencing Options on a Phone

To configure optional end-of-conference options for three-party ad hoc conferencing on a Cisco Unified IP phone running Skinny Client Control Protocol (SCCP), perform the following steps for each phone to be configured.

Prerequisites

- Conferencing uses call transfer to connect the two remaining parties of a conference when a conference initiator leaves the conference. To use this feature, you must configure the **transfer-system** command. For configuration information, see "Configuring Call Transfer and Forwarding" on page 763.
- Drop-last feature of Keep Conference on analog phones connected to the Cisco Unified CME system through a Cisco VG 224 requires Cisco IOS Release 12.4(9)T or later release.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *phone-tag*
4. **keep-conference** [**drop-last**] [**endcall**] [**local-only**]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone** *phone-tag* **Example:** `Router(config)# ephone 1`	Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks.
Step 4	**keep-conference** [**drop-last**] [**endcall**] [**local-only**] **Example:** `Router(config-ephone)# keep-conference endcall`	Allows conference initiators to exit from conference calls and to either end or maintain the conference for the remaining parties. • **no keep-conference**—(Default; the **no** form of the command) The conference initiator can hang up or press the EndCall soft key to end the conference and disconnect all parties or press the Confrn soft key to drop only the last party that was connected to the conference. • **keep-conference**—(No keywords used) The conference initiator can press the EndCall soft key to end the conference and disconnect all parties or hang up to leave the conference and keep the other two parties connected. The conference initiator can also use the Confrn soft key (IP phone) or hookflash (analog phone) to break up the conference but stay connected to both parties. • **drop-last**—The action of the Confrn soft key is changed; the conference initiator can press the Confrn soft key (IP phone) or hookflash (analog phone) to drop the last party. • **endcall**—The action of the EndCall soft key is changed; the conference initiator can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected. • **local-only**—The conference initiator can hang up to end the conference and leave the other two parties connected only if one of the remaining parties is local to the Cisco Unified CME system (an internal extension).
Step 5	**end** **Example:** `Router(config)# end`	Exits to privileged EXEC mode.

What to Do Next

If you are finished modifying the configuration, you are ready to generate configuration files for the phones to be connected. See "SCCP: Generating Configuration Files for SCCP Phones" on page 361.

SIP: Configuring Conferencing Options on a Phone

To configure optional end-of-conference options for three-party ad hoc conferencing on a Cisco Unified IP phone running SIP, perform the following steps for each phone to be configured.

Prerequisites

- To facilitate call transfer by using the Confrn soft key, conference and transfer attended or transfer blind must be enabled. For configuration information, see "Configuring Call Transfer and Forwarding" on page 763.

Restrictions

Music on hold (MOH) is not supported for call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool pool-tag**
4. **keep-conference**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. • *pool-tag*—Unique sequence number of the SIP phone to be configured. Range is 1 to 100 or the upper limit as defined by **max-pool** command.

	Command or Action	Purpose
Step 4	**keep-conference** **Example:** `Router(config-register-pool)# keep-conference`	Allows a Cisco Unified IP phone conference initiator to exit from conference calls and keeps the remaining parties connected. **Note** This step is included to illustrate how to enable the command if it was previously disabled. • Default is enabled. • Remaining calls are transferred without consultation as enabled by the **transfer-attended** (voice register template) or **transfer-blind** (voice register template) commands.
Step 5	**end** **Example:** `Router(config-register-pool)# end`	Exits to privileged EXEC mode.

What to Do Next

- If you are finished modifying the configuration, you are ready to generate configuration files for the phones to be connected. See "SIP: Generating Configuration Profiles for SIP Phones" on page 363.

Verifying Three-Party Ad Hoc Conferencing

Step 1 Use the **show running-config** command to verify your configuration. Any non-default conferencing parameters are listed in the telephony-service portion of the output, and end-of-conference options are listed in the ephone portion.

```
Router# show running-config
!
ephone-dn  1  dual-line
 ring feature secondary
 number 126 secondary 1261
 description Sales
 name Smith
 call-forward busy 500 secondary
 call-forward noan 500 timeout 10
 huntstop channel
 no huntstop
 no forward local-calls
!
ephone  1
 mac-address 011F.92A0.C10B
 type 7960 addon 1 7914
 no dnd feature-ring
 keep-conference
```

Troubleshooting Three-Party Ad Hoc Conferencing

Step 1 Use the **debug ephone** commands to observe messages and states associated with an ephone. For more information, see the *Cisco Unified CME Command Reference*.

SCCP: Configuring Multi-Party Ad Hoc and Meet-Me Conferencing in Cisco Unified CME 4.1 and Later Versions

To configure multi-party ad hoc conference support for 3-8 parties plus Meet-Me conferencing for up to 32 parties, perform the following tasks:

- SCCP: Enabling DSP Farm Services for a Voice Card, page 962 (required)
- SCCP: Configuring Join and Leave Tones, page 962 (optional)
- SCCP: Configuring SCCP for Cisco Unified CME, page 964 (required)
- SCCP: Configuring the DSP Farm, page 965 (required)
- SCCP: Associating Cisco Unified CME with a DSP Farm Profile, page 967 (required)
- SCCP: Enabling Multi-Party Ad Hoc and Meet-Me Conferencing, page 968 (required)
- SCCP: Configuring Multi-Party Ad Hoc Conferencing and Meet-Me Numbers, page 970 (required)
- SCCP: Configuring Conferencing Options for a Phone, page 972 (required)
- SCCP: Verifying Multi-Party Ad Hoc and Meet-Me Conferencing, page 975 (optional)

Prerequisites

- Cisco Unified CME 4.1 or a later version
- You must have a PVDM2-8, PVDM2-16, PVDM2-32, or PVDM2-64 high-density packet voice digital signal processor module hosted on the motherboard or on a module such as the NM-HDV2 or NM-HD-2VE.
- For Cisco Unified IP Phone 7985, firmware version 4-1-2-0 or a later version

Restrictions

- The maximum number of meet-me conference parties is 32 for one DSP using the G.711 codec and 16 for the G.729 codec.
- A participant cannot join more than one conference at the same time.
- Hardware-based multi-party ad hoc conferencing for more than three parties is not supported on phones that do not support soft keys.
- Hardware-based multi-party ad hoc conferencing for more than three parties is not supported on Cisco Unified IP phones running SIP.
- Hardware-based multi-party ad hoc conferencing does not support the local-consult transfer method (**transfer-system local-consult** command).

SCCP: Enabling DSP Farm Services for a Voice Card

To enable DSP farm services for a voice card to support multi-party ad hoc and meet-me conferences, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-card** *slot*
4. **dsp services dspfarm**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-card`** *`slot`* **Example:** `Router(config)# voice-card 2`	Enters voice-card configuration mode and configure a voice card.
Step 4	**`dsp services dspfarm`** **Example:** `Router(config-voicecard)# dsp services dspfarm`	Enables digital-signal-processor (DSP) farm services for a particular voice network module.
Step 5	**`exit`** **Example:** `Router(config-voicecard)# exit`	Exits voice-card configuration mode.

SCCP: Configuring Join and Leave Tones

To configure tones to be played when parties join and leave multi-party ad hoc conferences and meet-me conferences, perform the following steps for each tone to be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice class custom-cptone** *cptone-name*
4. **dualtone conference**
5. **frequency** *frequency-1* [*frequency-2*]
6. **cadence** {*cycle-1-on-time cycle-1-off-time* [*cycle-2-on-time cycle-2-off-time*] [*cycle-3-on-time cycle-3-off-time*] [*cycle-4-on-time cycle-4-off-time*] | **continuous**}
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class custom-cptone`** *`cptone-name`* **Example:** `Router(config)# voice class custom-cptone jointone`	Creates a voice class for defining custom call-progress tones to be detected.
Step 4	**`dualtone conference`** **Example:** `Router(cfg-cptone)# dualtone conference`	Configures conference join and leave tones.
Step 5	**`frequency`** *`frequency-1`* [*`frequency-2`*] **Example:** `Router(cfg-cp-dualtone)# frequency 600 900`	Defines the frequency components for a call-progress tone.
Step 6	**`cadence`** {*`cycle-1-on-time cycle-1-off-time`* [*`cycle-2-on-time cycle-2-off-time`*] [*`cycle-3-on-time cycle-3-off-time`*] [*`cycle-4-on-time cycle-4-off-time`*] | **`continuous`**} **Example:** `Router(cfg-cp-dualtone)# cadence 300 150 300 100 300 50`	Defines the tone-on and tone-off durations for a call-progress tone.
Step 7	**`end`** **Example:** `Router(cfg-cp-dualtone)# exit`	Exits configuration mode and enters privileged EXEC mode.

SCCP: Configuring SCCP for Cisco Unified CME

To enable SCCP on Cisco Unified CME to support multi-party ad hoc and meet-me conferences, perform the following steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sccp local** *interface-type interface-number* [**port** *port-number*]
4. **sccp ccm** {**ip-address** | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*]
5. **sccp ccm group** *group-number*
6. **bind interface** *interface-type interface-number*
7. **exit**
8. **sccp**
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sccp local** *interface-type interface-number* [**port** *port-number*] **Example:** `Router(config)# sccp local FastEthernet0/0`	Selects the local interface that SCCP applications (transcoding and conferencing) use to register with Cisco Unified CME.
Step 4	**sccp ccm** {**ip-address** | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*] **Example:** `Router(config)# sccp ccm 10.4.158.3 identifier 100 version 4.0`	Enables the Cisco Unified CME router to register SCCP applications. • *version-number*—Must be set to **4.0** or later.
Step 5	**sccp ccm group** *group-number* **Example:** `Router(config)# sccp ccm group 123`	Creates a Cisco Unified CME group.

	Command or Action	Purpose
Step 6	**bind interface** *interface-type interface-number* **Example:** `Router(config-sccp-cm)# bind interface fastethernet 0/0`	Binds an interface to a Cisco Unified CME group.

Step 7	**exit**
	
	`Example:`
	`Router(config-sccp-cm)# exit`
		Exits SCCP Cisco Unified CME configuration mode.

Step 8	**sccp** **Example:** `Router(config)# sccp`	Enables SCCP and its related applications (transcoding and conferencing).

Step 9	**exit**
	
	`Example:`
	`Router(config)# exit`
		Exits global configuration mode.

SCCP: Configuring the DSP Farm

To configure the DSP farm profile for multi-party ad hoc and meet-me conferencing, perform the following steps.

Note The DSP farm can be on the same router as the Cisco Unified CME or on a different router.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dspfarm profile** *profile-identifier* **conference**
4. **codec** {*codec-type* | **pass-through**}
5. **conference-join custom-cptone** *cptone-name*
6. **conference-leave custom-cptone** *cptone-name*
7. **maximum conference-participants** *max-participants*
8. **maximum sessions** *number*
9. **associate application sccp**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dspfarm profile** *profile-identifier* **conference** **Example:** `Router(config)# dspfarm profile 1 conference`	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
Step 4	**codec** {*codec-type* | **pass-through**} **Example:** `Router(config-dspfarm-profile)# codec g711ulaw`	Specifies the codecs supported by a DSP farm profile. **Note** Repeat this step as necessary to specify all the supported codecs.
Step 5	**conference-join custom-cptone** *cptone-name* **Example:** `Router(config-dspfarm-profile)# conference-join custom-cptone jointone`	Associates a custom call-progress tone to indicate joining a conference with a DSP farm profile. **Note** The *cptone-name* argument in this step must be the same as the *cptone-argument* in the **voice class custom-cptone** command configured in the "SCCP: Enabling DSP Farm Services for a Voice Card" section on page 962.
Step 6	**conference-leave custom-cptone** *cptone-name* **Example:** `Router(config-dspfarm-profile)# conference-leave custom-cptone leavetone`	Associates a custom call-progress tone to indicate leaving a conference with a DSP farm profile. **Note** The *cptone-name* argument in this step must be the same as the *cptone-argument* in the **voice class custom-cptone** command configured in the "SCCP: Enabling DSP Farm Services for a Voice Card" section on page 962.
Step 7	**maximum conference-participants** *max-participants* **Example:** `Router(config-dspfarm-profile)# maximum conference-participants 32`	(Optional) Configures the maximum number of conference parties allowed in each meet-me conference. The maximum is codec-dependent.
Step 8	**maximum sessions** *number* **Example:** `Router(config-dspfarm-profile)# maximum sessions 8`	Specifies the maximum number of sessions that are supported by the profile.

	Command or Action	Purpose
Step 9	**associate application sccp** **Example:** `Router(config-dspfarm-profile)# associate application sccp`	Associates SCCP with the DSP farm profile.
Step 10	**end** **Example:** `Router(config-dspfarm-profile)# end`	Exits to privileged EXEC mode.

SCCP: Associating Cisco Unified CME with a DSP Farm Profile

To associate a DSP farm profile with a group of Cisco Unified CME routers that control DSP services, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sccp ccm group** *group-number*
4. **associate ccm** *identifier-number* **priority** *priority-number*
5. **associate profile** *profile-identifier* **register** *device-name*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sccp ccm group** *group-number* **Example:** `Router(config)# sccp ccm group 1`	Creates a Cisco Unified CME group.
Step 4	**associate ccm** *identifier-number* **priority** *priority-number* **Example:** `Router(config-sccp-ccm)# associate ccm 100 priority 1`	Associates a Cisco Unified CME router with the group and establishes its priority within the group.

	Command or Action	Purpose
Step 5	**associate profile** *profile-identifier* **register** *device-name* **Example:** `Router(config-sccp-ccm)# associate profile 2 register confdsp1`	Associates a DSP farm profile with the Cisco Unified CME group. • *device-name* is a maximum of 16 characters. **Note** Repeat this step for every conferencing DSP farm and transcoding DSP farm.
Step 6	**end** **Example:** `Router(config-sccp-ccm)# end`	Exits to privileged EXEC mode.

SCCP: Enabling Multi-Party Ad Hoc and Meet-Me Conferencing

To allow hardware-based multi-party ad hoc conferences with more than three parties and meet-me conferences, perform the following steps.

Note Configuring multi-party ad hoc conferencing in Cisco Unified CME disables three-party (software-based) ad hoc conferencing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **conference hardware**
5. **transfer-system full-consult**
6. **sdspfarm units** *number*
7. **sdspfarm tag** *number device-name*
8. **sdspfarm conference mute-on** *mute-on-digits* **mute-off** *mute-off-digits*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**conference hardware** **Example:** `Router(config-telephony)# conference hardware`	Configures a Cisco Unified CME system for multi-party conferencing only.
Step 5	**transfer-system full-consult** **Example:** `Router(config-telephony)# transfer-system full-consult`	Transfers calls using H.450.2 with consultation using a second phone line, if available. • The calls fall back to full-blind if a second line is not available. • This is the default transfer method in Cisco Unified CME 4.0 and later versions.
Step 6	**sdspfarm units** *number* **Example:** `Router(config-telephony)# sdspfarm units 3`	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.
Step 7	**sdspfarm tag** *number device-name* **Example:** `Router(config-telephony)# sdspfarm tag 2 confdsp1`	Permits a DSP farm to register to Cisco Unified CME and associates it with a SCCP client interface's MAC address. **Note** The *device-name* in this step must be the same as the *device-name* in the **associate profile** command in Step 5 of the "SCCP: Associating Cisco Unified CME with a DSP Farm Profile" section on page 967.
Step 8	**sdspfarm conference mute-on** *mute-on-digits* **mute-off** *mute-off-digits* **Example:** `Router(config-telephony)# sdspfarm conference mute-on 111 mute-off 222`	Defines mute-on and mute-off digits for conferencing. • Maximum: 3 digits. Valid values are the numbers and symbols that appear on your telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #. • Mute-on and mute-off digits can be the same.
Step 9	**end** **Example:** `Router(config-telephony)# end`	Exits to privileged EXEC mode.

SCCP: Configuring Multi-Party Ad Hoc Conferencing and Meet-Me Numbers

To configure extension numbers for hardware-based multi-party ad hoc and meet-me ad hoc conferencing, based on the maximum number of conference participants you configure, perform the following steps. Ad hoc conferences require four extensions per conference, regardless of how many extensions are actually used by the conference parties.

Note Ensure that you configure enough directory numbers to accommodate the anticipated number of conferences. The maximum number of parties in a multi-party ad hoc conference on an IP phone is eight; the maximum on an analog phone is three.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* [**dual-line**]
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. **conference** {**ad-hoc** | **meetme**}
6. **preference** *preference-order* [**secondary** *secondary-order*]
7. **no huntstop** [**channel**]
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**ephone-dn** *dn-tag* **dual-line** **Example:** `Router(config)# ephone-dn 18 dual-line`	Enters ephone-dn configuration mode to configure an extension (ephone-dn) for a phone line. • Each ephone-dn can carry two parties if it is configured as a dual line. • Configure enough ephone-dns to accommodate the maximum number of conference participants to be supported. • For multi-party ad hoc conferencing, maximum number of directory numbers is 8, but you can configure a lower maximum. • For meet-me conferencing, maximum number of directory numbers is 32, but you can configure a lower maximum. • Minimum number of directory numbers required: 2.
Step 4	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 6789`	Associates a telephone or extension number with an ephone-dn in a Cisco Unified CME system. • Each DN for a conference must have the same primary and secondary number.

Step 5	**conference ad-hoc**
	or
	conference meetme
	
	Example:
	`Router(config-ephone-dn)# conference ad-hoc`
	or
	`Router(config-ephone-dn)# conference meetme`
		Configures a number as a placeholder for ad hoc conferencing to associate the call with the DSP farm.
		or
		(Optional) Associates meet-me conferencing with a directory number.

Step 6	**preference** *preference-order* [**secondary** *secondary-order*] **Example:** `Router(config-ephone-dn)# preference 1`	Sets dial-peer preference order for an extension (ephone-dn) associated with a Cisco Unified IP phone. • Remember to configure "preference x" with low value to last DN. • The lower the value of the *preference-order* argument, the higher the preference of the extension.
Step 7	**no huntstop** [**channel**] **Example:** `Router(config-ephone-dn)# no huntstop`	Continues call hunting behavior for an extension (ephone-dn) or an extension channel. • Remember to configure **no huntstop** for all DNs except the last one.
Step 8	**end** **Example:** `Router(config-ephone-dn)# end`	Exits to privileged EXEC mode.

SCCP: Configuring Conferencing Options for a Phone

To configure a template of conferencing features such as the add party mode, drop party mode, and soft keys for hardware-based multi-party ad hoc and meet-me conferences and apply the template to a phone, perform the following steps.

Note The following commands can also be configured in ephone configuration mode. Commands configured in ephone configuration mode have priority over commands in ephone-template configuration mode.

Prerequisites

- The RmLstC, ConfList, Join, and Select functions and soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration. For configuration information, see these tasks in this module:
  - "SCCP: Enabling DSP Farm Services for a Voice Card" section on page 962
  - "SCCP: Configuring the DSP Farm" section on page 965
  - "SCCP: Associating Cisco Unified CME with a DSP Farm Profile" section on page 967

Restrictions

- The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on a Cisco Unified IP Phone 7902, 7935, and 7936.
- The RmLstC, ConfList, Join, and Select functions and soft keys are not supported for software-based conferencing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-template** *template-tag*
4. **conference add-mode** [**creator**]
5. **conference drop-mode** [**creator** | **local**]
6. **conference admin**
7. **softkeys connected** {[**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**]}
8. **softkeys hold** {[**Join**] [**Newcall**] [**Resume**] [**Select**]}
9. **softkeys idle** {[**Cfwdall**] [**ConfList**] [**Dnd**] [**Gpickup**] [**HLog**] [**Join**] [**Login**] [**Newcall**] [**Pickup**] [**Redial**] [**RmLstC**]}
10. **softkeys seized** {[**CallBack**] [**Cfwdall**] [**Endcall**] [**Gpickup**] [**HLog**] [**MeetMe**] [**Pickup**] [**Redial**]}
11. **exit**
12. **ephone** *phone-tag*
13. **ephone-template** *template-tag*
14. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone-template** *template-tag* **Example:** `Router(config)# ephone-template 1`	Enter ephone-template configuration mode to create an ephone template to configure a set of phone features.
Step 4	**conference add-mode** [**creator**] **Example:** `Router(config-ephone-template)# conference add-mode creator`	(Optional) Configures the mode for adding parties to conferences. • **creator—**Only the creator can add parties to the conference.
Step 5	**conference drop-mode** [**creator** | **local**] **Example:** `Router(config-ephone-template)# conference drop-mode creator`	(Optional) Configures the mode for dropping parties from multi-party ad hoc conferences. • **creator**—The active conference terminates when the creator hangs up. • **local**—The active conference terminates when the last local party in the conference hangs up or drops out of the conference.
Step 6	**conference admin** **Example:** `Router(config-ephone-template)# conference admin`	(Optional) Configures the ephone as the conference administrator. The administrator can: • Dial in to any conference directly through the conference number • Use the ConfList soft key to list conference parties • Remove any party from any conference
Step 7	**softkeys connected** {[**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**]} **Example:** `Router(config-ephone-template)# softkeys connected Hold Trnsfer Park Endcall Confrn ConfList Join Select RmLstC`	Configures an ephone template for soft-key display during the connected call stage. • The soft keys used for multi-party conferencing are **RmLstC**, **ConfList**, **Join**, and **Select**. These soft keys are supported for hard-ware based conferencing only and require the appropriate DSP farm configuration. • The number and order of soft key keywords you enter in this command correspond to the number and order of soft keys on your phone.

	Command or Action	Purpose
Step 8	**softkeys hold** {[**Join**] [**Newcall**] [**Resume**] [**Select**]} **Example:** `Router(config-ephone-template)# softkeys hold Join Newcall Resume Select`	Configures an ephone template to modify soft-key display during the call-hold call stage. • The soft keys used for multi-party conferencing are **Join** and **Select**. These soft keys are supported for hard-ware based conferencing only and require the appropriate DSP farm configuration. • The number and order of soft key keywords you enter in this command correspond to the number and order of soft keys on your phone.
Step 9	**softkeys idle** {[**Cfwdall**] [**ConfList**] [**Dnd**] [**Gpickup**] [**HLog**] [**Join**] [**Login**] [**Newcall**] [**Pickup**] [**Redial**] [**RmLstC**]} **Example:** `Router(config-ephone-template)# softkeys idle ConfList Gpickup Join Login Newcall Pickup Redial RmLstC`	Configures an ephone template for soft-key display during the idle call stage. • The soft keys used for multi-party conferencing are **RmLstC**, **ConfList**, and **Join**. These soft keys are supported for hard-ware based conferencing only and require the appropriate DSP farm configuration. • The number and order of soft key keywords you enter in this command correspond to the number and order of soft keys on your phone.
Step 10	**softkeys seized** {[**CallBack**] [**Cfwdall**] [**Endcall**] [**Gpickup**] [**HLog**] [**MeetMe**] [**Pickup**] [**Redial**]} **Example:** `Router(config-ephone-template)# softkeys seized Redial Endcall Cfwdall Pickup Gpickup Callback Meetme`	(Optional) Configures an ephone template for soft-key display during the seized call stage. • You must configure the **MeetMe** soft key in the seized state for the ephone to initiate a meet-me conference. • The number and order of soft key keywords you enter in this command correspond to the number and order of soft keys on your phone.
Step 11	**exit** **Example:** `Router(config-ephone-template)# exit`	Exits ephone-template configuration mode.
Step 12	**ephone** *phone-tag* **Example:** `Router(config)# ephone 1`	Enters ephone configuration mode to create and configure an ephone.
Step 13	**ephone-template** *template-tag* **Example:** `Router(config-ephone)# ephone-dn-template 1`	Applies an ephone-dn template to an ephone-dn. **Note** The *template-tag* must be the same as the *template-tag* in Step 3.
Step 14	**end** **Example:** `Router(config-ephone)# exit`	Exits to privileged EXEC mode.

What to Do Next

If you are finished modifying the configuration, you are ready to generate configuration files for the phones to be connected. See "SCCP: Generating Configuration Files for SCCP Phones" on page 361.

SCCP: Verifying Multi-Party Ad Hoc and Meet-Me Conferencing

Use the following **show** commands to verify multi-party ad hoc and meet-me conferencing:

- **show ephone-dn conference**—Displays information about ad hoc and meet-me conferences.
- **show telephony-service conference hardware**—Displays information about hardware-based conferences.

show ephone-dn conference: Example

```
type   active inactive  numbers
=======================================
Meetme     0        8      2345
DN tags: 9, 10, 11, 12

Ad-hoc     0        8      A001
DN tags: 13, 14, 15, 16

Meetme     0        8      1234
DN tags: 20, 21, 22, 23
```

show telephony-service conference hardware detail: Example

```
Conference  Type  Active Max Peak  Master  MasterPhone  Last
                                             cur(initial)
==================================================================
8889        Ad-hoc  3      8    3    8044     29     ( 29)    8012
Conference parties:
    8012
    8006
    8044
```

SCCP: Configuring Meet-Me Conferencing in Cisco CME 3.2 to Cisco Unified CME 4.0

Refer to the "Examples" section on page 977 to configure Meet-Me Conferencing on a Cisco router with Cisco CME 3.2 or a later version and Cisco Unity Express.

Note To configure Meet-Me Conferencing in Cisco Unified CME 4.1 or a later version, see the "SCCP: Configuring Multi-Party Ad Hoc and Meet-Me Conferencing in Cisco Unified CME 4.1 and Later Versions" section on page 961

Prerequisites

- Cisco CME 3.2 to Cisco Unified CME 4.0.

- A dual VWIC-2MFT-T1 or E-1 loopback for internal callers. The number of VWIC-2MFT-T1 cards required depends on the number of local IP phones parties that need to dial into the meet-me conference. Each VWIC-2MFT-T1 card can support 24 local IP phone parties.
- Packet Voice DSP Modules (PVDM DSPs) to handle the number of callers in conference. A maximum of 96 conference parties is supported using an approved platform, such as a Cisco 3800 router, with at least two PVDM2-64DSPs installed.
- Your IP network is operational and you can access Cisco web.
- You have a valid Cisco.com account.
- The recommended Cisco IOS release and Cisco Unified CME phone firmware and GUI files to support Cisco Unity Express are installed on the Cisco Unified CME router.

  To determine whether the Cisco IOS software release and Cisco Unified CME software version are compatible with the Cisco Unity Express version, Cisco router model, and Cisco Unity Express hardware that you are using, see the Cisco Unity Express Compatibility Matrix.

  To verify installed Cisco Unity Express software version, enter the Cisco Unity Express command environment and use the **show software version** user EXEC command. For information about the command environment, see the appropriate *Cisco Unity Express CLI Administrator Guide* at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.
- The proper Cisco Unity Express license for Cisco Unified CME, not Cisco Unified Communications Manager, is installed. To verify installed license, enter the Cisco Unity Express command environment and use the **show software license** user EXEC command. For information about the command environment, see the appropriate *Cisco Unity Express CLI Administrator Guide* at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.

  This is an example of the Cisco Unified CME license:

```
se-10-0-0-0> show software licenses

Core:
- application mode: CCME
- total usable system ports: 8

Voicemail/Auto Attendant:
- max system mailbox capacity time: 6000
- max general delivery mailboxes: 15
- max personal mailboxes: 50

Languages:
- max installed languages: 1
- max enabled languages: 1
```

- Calls can be successfully completed between phones on the same Cisco Unified CME router.
- Dial plan for Meet-Me Conferencing is defined. For information, see "Dial Plan" section on page 955.

Restrictions

- The number of meet-me conferences and parties per conference is limited by the number of DSP resources and number of voice ports available to handle callers.
- There is no set maximum for the number of parties per conference. However, since only the three loudest parties on a multicast conference can be heard, we recommend that the maximum number of parties per conference be limited to eight.

- Only a minimal set of features are provided. Conference bridges can be accessed by any user knowing the correct number to dial (internal or external) with no option to set a password. Callers entering a Meet-Me conference though Cisco Unity Express auto-attendant application are prompted to record their name for playback to all callers on the bridge. No exit tone is played when users leave a conference, nor can a Meet-Me bridge be reserved for use at a future time or date.

Examples

The following partial output from the **show running-config** command shows the configuration on a Cisco 2821 router with Cisco Unified CME and Cisco Unity Express, with comments describing the configuration for setting up Meet-Me Conferencing.

```
Router# show running-config
building configuration...
.
.
.
.
.
!
!---Two T1 ports connected back-to-back to bridge VOIP to Multicast
controller T1 0/3/0
 framing esf
 linecode b8zs
    ds0-group 1 timeslots 1 type e&m-immediate-start
    ds0-group 2 timeslots 2 type e&m-immediate-start
    ds0-group 3 timeslots 3 type e&m-immediate-start
    ds0-group 4 timeslots 4 type e&m-immediate-start
    ds0-group 5 timeslots 5 type e&m-immediate-start
    ds0-group 6 timeslots 6 type e&m-immediate-start
    ds0-group 7 timeslots 7 type e&m-immediate-start
    ds0-group 8 timeslots 8 type e&m-immediate-start
    ds0-group 9 timeslots 9 type e&m-immediate-start
    ds0-group 10 timeslots 10 type e&m-immediate-start
    ds0-group 11 timeslots 11 type e&m-immediate-start
    ds0-group 12 timeslots 12 type e&m-immediate-start
    ds0-group 13 timeslots 13 type e&m-immediate-start
    ds0-group 14 timeslots 14 type e&m-immediate-start
    ds0-group 15 timeslots 15 type e&m-immediate-start
    ds0-group 16 timeslots 16 type e&m-immediate-start
    ds0-group 17 timeslots 17 type e&m-immediate-start
    ds0-group 18 timeslots 18 type e&m-immediate-start
    ds0-group 19 timeslots 19 type e&m-immediate-start
    ds0-group 20 timeslots 20 type e&m-immediate-start
    ds0-group 21 timeslots 21 type e&m-immediate-start
    ds0-group 22 timeslots 22 type e&m-immediate-start
    ds0-group 23 timeslots 23 type e&m-immediate-start
    ds0-group 24 timeslots 24 type e&m-immediate-start
!
controller T1 0/3/1
 framing esf
 clock source internal
 linecode b8zs
    ds0-group 1 timeslots 1 type e&m-immediate-start
    ds0-group 2 timeslots 2 type e&m-immediate-start
    ds0-group 3 timeslots 3 type e&m-immediate-start
    ds0-group 4 timeslots 4 type e&m-immediate-start
    ds0-group 5 timeslots 5 type e&m-immediate-start
    ds0-group 6 timeslots 6 type e&m-immediate-start
    ds0-group 7 timeslots 7 type e&m-immediate-start
    ds0-group 8 timeslots 8 type e&m-immediate-start
```

```
    ds0-group 9 timeslots 9 type e&m-immediate-start
    ds0-group 10 timeslots 10 type e&m-immediate-start
    ds0-group 11 timeslots 11 type e&m-immediate-start
    ds0-group 12 timeslots 12 type e&m-immediate-start
    ds0-group 13 timeslots 13 type e&m-immediate-start
    ds0-group 14 timeslots 14 type e&m-immediate-start
    ds0-group 15 timeslots 15 type e&m-immediate-start
    ds0-group 16 timeslots 16 type e&m-immediate-start
    ds0-group 17 timeslots 17 type e&m-immediate-start
    ds0-group 18 timeslots 18 type e&m-immediate-start
    ds0-group 19 timeslots 19 type e&m-immediate-start
    ds0-group 20 timeslots 20 type e&m-immediate-start
    ds0-group 21 timeslots 21 type e&m-immediate-start
    ds0-group 22 timeslots 22 type e&m-immediate-start
    ds0-group 23 timeslots 23 type e&m-immediate-start
    ds0-group 24 timeslots 24 type e&m-immediate-start
!
!
!
!--- Disable keepalive packet to multicast network on voice class and apply to LMR port
!
voice class permanent 1
 signal timing oos restart 50000
 signal timing oos timeout disabled
 signal keepalive disabled
 signal sequence oos no-action

!---Loopback0 used as source for all H323 and SCCP packets generated by CME
interface Loopback0
 ip address 11.1.1.1 255.255.255.255
 h323-gateway voip interface
 h323-gateway voip bind srcaddr 11.1.1.1
!
!---Vif1 (virtual host interface) used as source for all multicast packets generated by
CME
!
interface Vif1
 ip address 192.168.11.1 255.255.255.252
 ip pim dense-mode
!
interface FastEthernet0/0
 no ip address
 shutdown
!
!---Service-engine interface used to access Cisco Unity Express
!
interface Service-Engine0/0
 ip unnumbered Vlan10
service-module ip address 192.168.1.2 255.255.255.0
service-module ip default-gateway 192.168.1.1
!
interface FastEthernet0/1
no ip address
shutdown
!
interface FastEthernet0/0/0
 switchport access vlan 10
 no ip address
!
interface FastEthernet0/0/1
 switchport access vlan 10
 no ip address
!
interface FastEthernet0/0/2
```

```
 switchport access vlan 10
 no ip address
!
interface FastEthernet0/0/3
 switchport access vlan 10
 no ip address
!
interface Vlan1
 no ip address
!
!---All IP phones reside on VLAN 10
interface Vlan10
 ip address 192.168.1.1 255.255.255.0
 ip pim dense-mode
!
 ip classless
!--- Static route to reach other devices on network
ip route 0.0.0.0 0.0.0.0 192.168.1.2
!--- Static route to reach Cisco Unity Express
ip route 192.168.1.2 255.255.255.255 Service-Engine0/0
!
ip http server
ip http path flash:
!
!
tftp-server flash:P00305000301.sbn
!
control-plane
!
!
!
!---VOIP side of the Back-to-Back T1 used for bridging VOIP to
!---Multicast (Hoot n' Holler)
!---Port 0/3/0:x connects to Port 0/3/1:x
voice-port 0/3/0:1
 auto-cut-through
!
voice-port 0/3/0:2
 auto-cut-through
!
.
.
.
!
voice-port 0/3/0:24
 auto-cut-through
!
!---Multicast side of the Back-to-Back T1 used for bridging VOIP to
!---Multicast (Hoot n' Holler)
!--- Port 0/3/1:1 - 8 is permanently trunked to multicast bridge A212
!--- Port 0/3/1:9 - 16 is permanently trunked to multicast bridge A213
!--- Port 0/3/1:17 - 24 is permanently trunked to multicast bridge A214
voice-port 0/3/1:1
 auto-cut-through
 timeouts call-disconnect 3
 connection trunk A212
!
.
.
.
!
voice-port 0/3/1:9
 auto-cut-through
 timeouts call-disconnect 3
```

```
 connection trunk A213
!
.
.
.
!
voice-port 0/3/1:17
 auto-cut-through
 timeouts call-disconnect 3
 connection trunk A214
.
.
.
!
!--- Analog FXO lines on port 0/2/x route incoming calls to CUE AA external extension 203
voice-port 0/2/0
 connection plar opx 203
!
voice-port 0/2/1
 connection plar opx 203
!
voice-port 0/2/2
 connection plar opx 203
!
voice-port 0/2/3
 connection plar opx 203
!
!--- LMR devices are connected to E&M ports 0/1/x. The E&M ports are permanently trunked
to multicast conference bridges. Port 0/1/0 will send and receive audio from conference
A212 and port 0/1/1 will send and receive audio from conference A213.
voice-port 0/1/0
 voice-class permanent 1
 lmr m-lead audio-gate-in
 lmr e-lead voice
 auto-cut-through
 operation 4-wire
 type 3
 signal lmr
 timeouts call-disconnect 3
 connection trunk A212
!
voice-port 0/1/1
 voice-class permanent 1
 lmr m-lead audio-gate-in
 lmr e-lead voice
 auto-cut-through
 operation 4-wire
 type 3
 signal lmr
 timeouts call-disconnect 3
 connection trunk A213
!
!--- Dial-peers to route extension 212 to T1 loopback, which is trunked to bridge A212
dial-peer voice 1 pots
 preference 1
 destination-pattern 212
 port 0/3/0:1
!
.
.
.
!
dial-peer voice 8 pots
 preference 8
```

```
 destination-pattern 212
 port 0/3/0:8
!
!--- Dial-peers to route extension 213 to T1 loopback, which is trunked to bridge A213
dial-peer voice 9 pots
 preference 1
 destination-pattern 213
 port 0/3/0:9
!
.
.
.
!
dial-peer voice 16 pots
 preference 8
 destination-pattern 213
 port 0/3/0:16
!
!--- Dial-peers to route extension 214 to T1 loopback, which is trunked to bridge A214
dial-peer voice 17 pots
 preference 1
 destination-pattern 214
 port 0/3/0:17
!
.
.
.
!
dial-peer voice 24 pots
 preference 8
 destination-pattern 214
 port 0/3/0:24
!--- Dial-peer to route calls to CUE AA for internal ext. 202 and external ext. 203
dial-peer voice 200 voip
 destination-pattern 20.
 session protocol sipv2
 session target ipv4:192.168.1.2
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
!--- Dial-peers for multicast bridges
dial-peer voice 212 voip
 destination-pattern A212
 voice-class permanent 1
 session protocol multicast

 session target ipv4:237.111.0.0:22222
 dtmf-relay cisco-rtp
 codec g711ulaw
 vad aggressive
!
dial-peer voice 213 voip
 destination-pattern A213
 voice-class permanent 1
 session protocol multicast
 session target ipv4:237.111.0.1:22222
 dtmf-relay cisco-rtp
 codec g711ulaw
 vad aggressive
!
dial-peer voice 214 voip
 destination-pattern A214
 voice-class permanent 1
```

```
 session protocol multicast
 session target ipv4:237.111.0.2:22222
 dtmf-relay cisco-rtp
 codec g711ulaw
 vad aggressive
!
telephony-service
 load 7960-7940 P00305000301
 max-ephones 24
 max-dn 144
 ip source-address 11.1.1.1 port 2000
  create cnf-files version-stamp Jan 01 2002 00:00:00
 voicemail 200
 web admin system name cisco password cisco
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
ephone-dn 1 dual-line
 number 150
!
.
.
.
```

What to Do Next

Load and configure the auto-attendant script file for Meet-me Conferencing. For information about logging into and GUI windows and menus, see the appropriate *Cisco Unity Express GUI Administrator Guide* at http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/roadmap/cuedocs.html.

Step 1 Go to the Download Software site. Download the Conference Express TCL and AA voice files (conf-express.zip). Unzip the archive to a folder on your PC.

Step 2 Log into Cisco Unity Express as administrator.

Step 3 Navigate to the Voice mail> Auto Attendant menu and click **Add**. The Add a New Automated Attendant window appears.

Step 4 In the Select Automated Attendant area, configure the parameters listed in the following table. Enter the required information in the corresponding field.

.

Parameter Name	Value
Select Automated Attendant Script	mp-exp.aef
Application Name (lower case)	conference-express
Destination file name	mp-exp.aef

Step 5 Click **Next**. The Upload window appears.

Step 6 Upload the script (mp-exp.aef) from your PC to the auto-attendant application. For information, see online help.

Step 7 On the Add a New Automated Attendant window, configure parameters with numbers as defined in your dial plan and with the values listed in following table. Enter the required information in the corresponding field. For dial plan information, see the "Dial Plan" section on page 955.

Field Name	Value
Script Parameters	
BridgeDir	bridge.wav
record_name	record_name.wav
SystemProblems	SystemProblems.wav
Call Handling	
Call-in Number	InternalNumber as defined in dial plan
Maximum Sessions	4

Step 8 Click **Finish**.

Step 9 Navigate to the Administration>Call-In Numbers menu and click **Add**.

Step 10 On the Add a Call-In Number window, configure the parameters listed in the following table. Enter the required information in the corresponding field.

Field Name	Value
Application	conference-express
Call-in Number	ExternalNumber as defined in dial plan
Maximum Sessions	4

Step 11 Click **Add**.

Step 12 Confirm that two call-in numbers for the conference-express application are enabled on the Administration>Call-In Numbers window.

Configuration Examples for Conferencing

This section provides the following configuration examples:

Basic Conferencing: Example

The following example sets the maximum number of conferences for a Cisco Unified IP phone to 4 and configures a gain of 6 db for inbound audio packets from remote PSTN or VoIP calls joining a conference:

```
telephony-service
 max-conferences 4 gain 6
```

End of Conference Options: Example

In the following example, extension 3555 initiates a three-way conference. After the conference is established, extension 3555 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. If extension 3555 hangs up from the conference, the other two parties remain connected if one of them is local to the Cisco Unified CME system.

```
ephone-dn 35
 number 3555

ephone 24
 button 1:35
 keep-conference drop-last local-only
```

In the following example, extension 3666 initiates a three-way conference. After the conference is established, extension 3666 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3666 can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected.

```
ephone-dn 36
 number 3666

ephone 25
 button 1:36
 keep-conference drop-last endcall
```

In the following example, extension 3777 initiates a three-way conference. After the conference is established, extension 3777 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3777 can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected *only* if one of the two parties is local to the Cisco Unified CME system.

```
ephone-dn 38
 number 3777

ephone 27
 button 1:38
 keep-conference drop-last endcall local-only
```

In the following example, extension 3999 initiates a three-way conference. After the conference is established, extension 3999 can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected *only* if one of the two parties is local to the Cisco Unified CME system. Extension 3999 can also use the Confrn soft key to break up the conference but stay connected to both parties.

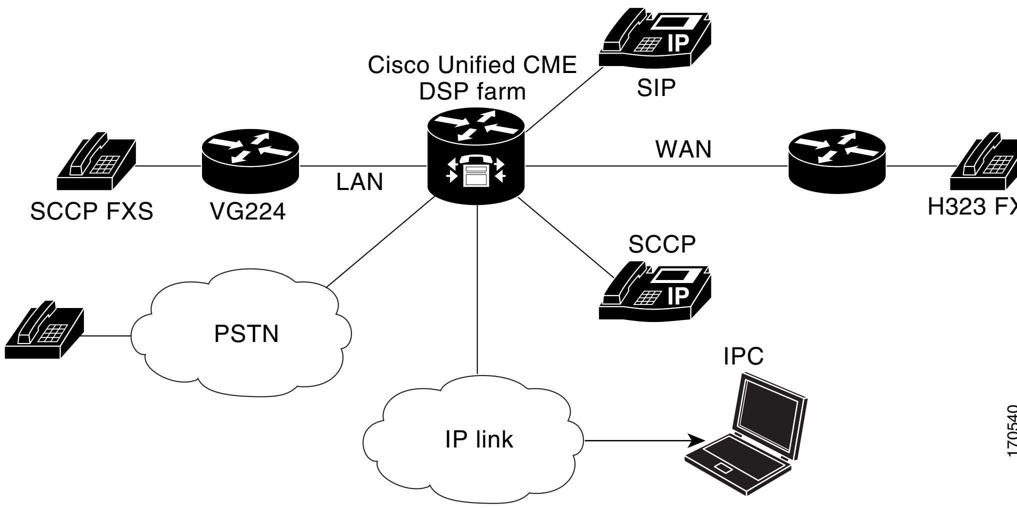
```
ephone-dn 39
 number 3999

ephone 29
 button 1:39
 keep-conference endcall local-only
```

DSP Farm and Cisco Unified CME on the Same Router: Example

In this example, the DSP farm and Cisco Unified CME are on the same router as shown in Figure 33.

Figure 33 CME and the DSP Farm on the Same Router




```
Current configuration : 16345 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
service internal
!
hostname cmedsprtr
!
boot-start-marker
boot-end-marker
!
logging buffered 90000 debugging
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
no network-clock-participate wic 0
ip cef
!
!
ip dhcp pool phone1
 host 10.4.188.66 255.255.0.0
 client-identifier 0100.0ab7.b144.4a
 default-router 10.4.188.65
 option 150 ip 10.4.188.65
!
ip dhcp pool phone2
 host 1.4.188.67 255.255.0.0
 client-identifier 0100.3094.c269.35
 default-router 10.4.188.65
 option 150 ip 10.4.188.65
!
```

```
!
voice-card 1
 dsp services dspfarm
!
!
voice call send-alert
voice call carrier capacity active
!
voice service voip
 allow-connections h323 to h323
 supplementary-service h450.12
 h323
!
!
!
!
controller E1 1/0
 framing NO-CRC4
!
controller E1 1/1
!
!
interface FastEthernet0/0
 ip address 10.4.188.65 255.255.0.0
 duplex auto
 speed auto
 no keepalive
 no cdp enable
 no clns route-cache
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 no clns route-cache
!
ip route 10.4.0.0 255.255.0.0 FastEthernet0/0
ip route 192.168.254.254 255.255.255.255 10.4.0.1
!
ip http server
!
!
control-plane
!
!
sccp local FastEthernet0/0
sccp ccm 10.4.188.65 identifier 1 version 4.0
sccp
!
sccp ccm group 123
 associate ccm 1 priority 1
 associate profile 1 register mtp00097c5e9ce0
 keepalive retries 5
!
!
dspfarm profile 1 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 6
```

```
 associate application SCCP
!
dial-peer cor custom
!
!
!
dial-peer voice 6 voip
 destination-pattern 6...
 session target ipv4:10.4.188.90
!
telephony-service
 conference hardware
 load 7960-7940 P00307020400
 load 7905 CP7905060100SCCP050309A.sbin
 max-ephones 48
 max-dn 180
 ip source-address 10.4.188.65 port 2000
 timeouts ringing 500
 system message MY MELODY (2611)
 sdspfarm units 4
 sdspfarm tag 1 mtp00097c5e9ce0
 max-conferences 4 gain -6
 call-forward pattern ....
 transfer-system full-consult
 transfer-pattern 7...
 transfer-pattern ....
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
ephone-template 1
 softkeys hold Newcall Resume Select Join
 softkeys idle Cfwdall ConfList Dnd Gpickup HLog Join Login Newcall Pickup Redial RmLstC
 softkeys seized Redial Pickup Gpickup HLog Meetme Endcall
 softkeys connected Acct ConfList Confrn Endcall Flash HLog Hold Join Park RmLstC Select
 Trnsfer
!
!
ephone-dn 1 dual-line
 number 8001
 name melody-8001
!
!
ephone-dn 2 dual-line
 number 8002
!
!
ephone-dn 3 dual-line
 number 8003
!
!
ephone-dn 4 dual-line
 number 8004
!
!
ephone-dn 5 dual-line
 number 8005
!
!
ephone-dn 6 dual-line
 number 8006
!
!
ephone-dn 7 dual-line
 number 8007
```

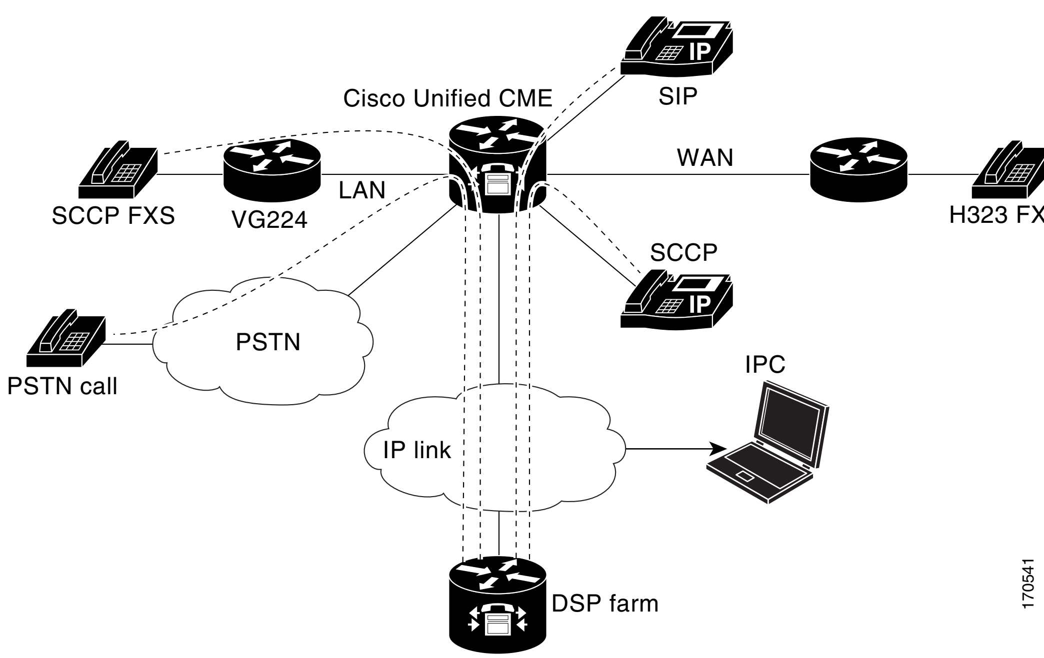
```
!
!
ephone-dn 8 dual-line
 number 8008
!
!
ephone-dn 60 dual-line
 number 8887
 conference meetme
 no huntstop
!
!
ephone-dn 61 dual-line
 number 8887
 conference meetme
 preference 1
 no huntstop
!
!
ephone-dn 62 dual-line
 number 8887
 conference meetme
 preference 2
 no huntstop
!
!
ephone-dn 63 dual-line
 number 8887
 conference meetme
 preference 3
!
!
ephone-dn 64 dual-line
 number 8889
 name Conference
 conference ad-hoc
 no huntstop
!
!
ephone-dn 65 dual-line
 number 8889
 name Conference
 conference ad-hoc
 preference 1
 no huntstop
!
!
ephone-dn 66 dual-line
 number 8889
 name Conference
 conference ad-hoc
 preference 2
 no huntstop
!
!
ephone-dn 67 dual-line
 number 8889
 name Conference
 conference ad-hoc
 preference 3
!
!
ephone 1
 ephone-template 1
```

```
 mac-address 0030.94C2.6935
 type 7960
 button 1:1 2:2
!
!
ephone 2
 ephone-template 1
 mac-address 000A.B7B1.444A
 type 7940
 button 1:4 2:8
!
line con 0
 exec-timeout 0 0
line aux 0
 exec-timeout 0 0
line vty 0 4
 exec-timeout 0 0
 login
line vty 5 15
 login
!
!
end
```

DSP Farm and Cisco Unified CME on Different Routers: Example

In this example, the DSP farm and Cisco Unified CME are on different routers as shown in Figure 34.

Figure 34 Cisco Unified CME and the DSP Farm on Different Routers



This section contains configuration examples for the following routers:

- Cisco Unified CME Router Configuration: Example, page 990
- DSP Farm Router Configuration: Example, page 996

Cisco Unified CME Router Configuration: Example

```
Current configuration : 5659 bytes
!
version 12.4
no service timestamps debug uptime
no service timestamps log uptime
no service password-encryption
!
boot-start-marker
boot-end-marker
!
!
card type command needed for slot 1
logging buffered 3000000 debugging
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
no network-clock-participate aim 0
!
voice-card 1
 no dspfarm
!
voice-card 3
 dspfarm
!
ip cef
!
!
no ip dhcp use vrf connected
!
ip dhcp pool IPPhones
 network 10.15.15.0 255.255.255.0
 option 150 ip 10.15.15.1
 default-router 10.15.15.1
!
!
interface FastEthernet0/0
 ip address 10.3.111.102 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 duplex auto
 speed auto
!
interface FastEthernet0/1.1
 encapsulation dot1Q 10
 ip address 10.15.14.1 255.255.255.0
!
interface FastEthernet0/1.2
 encapsulation dot1Q 20
 ip address 10.15.15.1 255.255.255.0
!
ip route 0.0.0.0 0.0.0.0 10.5.51.1
ip route 0.0.0.0 0.0.0.0 10.3.0.1
!
ip http server
!
```

```
!
!
!
control-plane!
!
!
!
dial-peer voice 1 voip
 destination-pattern 3...
 session target ipv4:10.3.111.101
!
!
telephony-service
 conference hardware
 load 7910 P00403020214
 load 7960-7940 P003-07-5-00
 max-ephones 50
 max-dn 200
 ip source-address 10.15.15.1 port 2000
 sdspfarm units 4
 sdspfarm transcode sessions 12
 sdspfarm tag 1 confer1
 sdspfarm tag 4 xcode1
 max-conferences 8 gain -6
 moh flash:music-on-hold.au
 multicast moh 239.0.0.0 port 2000
 transfer-system full-consult
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
ephone-template 1
 softkeys hold Resume Newcall Select Join
 softkeys idle Redial Newcall ConfList RmLstC Cfwdall Join Pickup Login HLog Dnd Gpickup
 softkeys seized Endcall Redial Cfwdall Meetme Pickup Callback
 softkeys alerting Endcall Callback
 softkeys connected Hold Endcall Confrn Trnsfer Select Join ConfList RmLstC Park Flash
!
ephone-dn 1 dual-line
 number 6000
!
!
ephone-dn 2 dual-line
 number 6001
!
!
ephone-dn 3 dual-line
 number 6002
!
!
ephone-dn 4 dual-line
 number 6003
!
!
ephone-dn 5 dual-line
 number 6004
!
!
ephone-dn 6 dual-line
 number 6005
!
!
ephone-dn 7 dual-line
 number 6006
!
```

```
!
ephone-dn 8 dual-line
 number 6007
!
!
ephone-dn 9 dual-line
 number 6008
!
!
ephone-dn 10 dual-line
 number 6009
!
!
ephone-dn 11
 number 6011
!
!
ephone-dn 12
 number 6012
!
!
ephone-dn 13
 number 6013
!
!
ephone-dn 14
 number 6014
!
!
ephone-dn 15
 number 6015
!
!
ephone-dn 16
 number 6016
!
!
ephone-dn 17
 number 6017
!
!
ephone-dn 18
 number 6018
!
!
ephone-dn 19
 number 6019
!
!
ephone-dn 20
 number 6020
!
!
ephone-dn 21
 number 6021
!
!
ephone-dn 22
 number 6022
!
!
ephone-dn 23
 number 6023
!
```

```
!
ephone-dn 24
 number 6024
!
!
ephone-dn 25 dual-line
 number 6666
 conference meetme
 preference 1
 no huntstop
!
!
ephone-dn 26 dual-line
 number 6666
 conference meetme
 preference 2
 no huntstop
!
!
ephone-dn 27 dual-line
 number 6666
 conference meetme
 preference 3
 no huntstop
!
!
ephone-dn 28 dual-line
 number 6666
 conference meetme
 preference 4
 no huntstop
!
!
ephone-dn 29 dual-line
 number 8888
 conference meetme
 preference 1
 no huntstop
!
!
ephone-dn 30 dual-line
 number 8888
 conference meetme
 preference 2
 no huntstop
!
!
ephone-dn 31 dual-line
 number 8888
 conference meetme
 preference 3
 no huntstop
!
!
ephone-dn 32 dual-line
 number 8888
 conference meetme
 preference 4
!
!
ephone-dn 33
 number 6033
!
!
```

```
ephone-dn 34
 number 6034
!
!
ephone-dn 35
 number 6035
!
!
ephone-dn 36
 number 6036
!
!
ephone-dn 37
 number 6037
!
!
ephone-dn 38
 number 6038
!
!
ephone-dn 39
 number 6039
!
!
ephone-dn 40
 number 6040
!
!
ephone-dn 41 dual-line
 number 6666
 conference meetme
 preference 5
 no huntstop
!
!
ephone-dn 42 dual-line
 number 6666
 conference meetme
 preference 6
 no huntstop
!
!
ephone-dn 43 dual-line
 number 6666
 conference meetme
 preference 7
 no huntstop
!
!
ephone-dn 44 dual-line
 number 6666
 conference meetme
 preference 8
 no huntstop
!
!
ephone-dn 45 dual-line
 number 6666
 conference meetme
 preference 9
 no huntstop
!
!
ephone-dn 46 dual-line
```

```
 number 6666
 conference meetme
 preference 10
 no huntstop
!
!
ephone-dn 47 dual-line
 number 6666
 conference meetme
 preference 10
 no huntstop
!
!
ephone-dn 48 dual-line
 number 6666
 conference meetme
 preference 10
!
!
ephone-dn 51 dual-line
 number A0001
 name conference
 conference ad-hoc
 preference 1
 no huntstop
!
!
ephone-dn 52 dual-line
 number A0001
 name conference
 conference ad-hoc
 preference 2
 no huntstop
!
!
ephone-dn 53 dual-line
 number A0001
 name conference
 conference ad-hoc
 preference 3
 no huntstop
!
!
ephone-dn 54 dual-line
 number A0001
 name conference
 conference ad-hoc
 preference 4
!
!
ephone 1
 ephone-template 1
 mac-address C863.B965.2401
 type anl
 button 1:1
!
!
!
ephone 2
 ephone-template 1
 mac-address 0016.C8BE.A04A
 type 7920
!
!
```

```
!
ephone 3
 ephone-template 1
 mac-address C863.B965.2400
 type anl
 button 1:2
!
!
!
ephone 4
 no multicast-moh
 ephone-template 1
 mac-address 0017.952B.7F5C
 type 7912
 button 1:4
!
!
!
ephone 5
 ephone-template 1
 ephone 6
 no multicast-moh
 ephone-template 1
 mac-address 0017.594F.1468
 type 7961GE
 button 1:6
!
!
!
ephone 11
 ephone-template 1
 mac-address 0016.C8AA.C48C
 button 1:10 2:15 3:16 4:17
 button 5:18 6:19 7:20 8:21
 button 9:22 10:23 11:24 12:33
 button 13:34 14:35 15:36 16:37
 button 17:38 18:39 19:40
!
!
line con 0
line aux 0
line vty 0 4
 login
!
!
end
```

DSP Farm Router Configuration: Example

```
Current configuration : 2179 bytes
!
! Last configuration change at 05:47:23 UTC Wed Jul 12 2006
!
version 12.4
service timestamps debug datetime msec localtime
no service timestamps log uptime
no service password-encryption
hostname dspfarmrouter
!
boot-start-marker
boot-end-marker
!
```

```
!
card type command needed for slot 1
logging buffered 4096 debugging
enable password lab
!
no aaa new-model
!
resource policy
!
no network-clock-participate slot 1
!
!
ip cef
!
!
no ip domain lookup
!
!
voice-card 0
 no dspfarm
!
voice-card 1
 no dspfarm
 dsp services dspfarm

interface GigabitEthernet0/0
 ip address 10.3.111.100 255.255.0.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1.1
 encapsulation dot1Q 100
 ip address 192.168.1.10 255.255.255.0
!
interface GigabitEthernet0/1.2
 encapsulation dot1Q 200
 ip address 192.168.2.10 255.255.255.0
!
interface GigabitEthernet0/1.3
 encapsulation dot1Q 10
 ip address 10.15.14.10 255.255.255.0
!
interface GigabitEthernet0/1.4
 encapsulation dot1Q 20
 ip address 10.15.15.10 255.255.255.0
!
ip route 10.0.0.0 255.0.0.0 10.3.0.1
ip route 192.168.0.0 255.0.0.0 10.3.0.1
!
!
ip http server
!
!
!
!
control-plane
!
sccp local GigabitEthernet0/0
sccp ccm 10.15.15.1 identifier 1 version 4.1
!
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 101 register confer1
```

```
 associate profile 103 register xcode1
!
!
dspfarm profile 103 transcode
 codec g711ulaw
 codec g711alaw
 codec g729r8
 maximum sessions 6
 associate application SCCP
!
dspfarm profile 101 conference
 codec g711ulaw
 codec g711alaw
 codec g729r8
 maximum sessions 5
 associate application SCCP
!
!
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 session-timeout 300
 exec-timeout 0 0
 password
 no login
!
scheduler allocate 20000 1000
!
end
```

Where to Go Next

Controlling Use of the Conference Soft Key

To block the functioning of the conference (Confrn) soft key without removing the key display, create and apply an ephone template that contains the **features blocked** command. For more information, see “Creating Templates” on page 1525.

To remove the conference (Confrn) soft key from one or more phones, create and apply an ephone template that contains the appropriate **softkeys** command. For more information, see “Customizing Soft Keys” on page 1335.

Additional References

The following sections provide references related to conferencing.

Related Documents

Related Topic	Document Title
Cisco Unified CME configuration	• *Cisco Unified CME Command Reference* • Cisco Unified CME Documentation Roadmap
Cisco IOS commands	• *Cisco IOS Voice Command Reference* • Cisco IOS Software Releases 12.4T Command References
Cisco IOS configuration	• *Cisco IOS Voice Configuration Library* • Cisco IOS Software Releases 12.4T Configuration Guides
Phone documentation for Cisco Unified CME	• User Documentation for Cisco Unified IP Phones

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Feature Information for Conferencing

Table 29 lists the features in this module and enhancements to the features by version.

To determine the correct Cisco IOS release to support a specific Cisco Unified CME version, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/requirements/guide/33matrix.htm.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 29 lists the Cisco Unified CME version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature.

Table 29 Feature Information for Conferencing

Feature Name	Cisco Unified CME Version	Feature Information
Meet-me Conferences	4.1	Added support for hardware-based meet-me conferences created by parties calling a designated conference number.
Multi-party Ad Hoc Conferencing	4.1	Added support for hardware-based Multi-party Conferencing Enhancements which uses DSPs to enhance ad hoc conferencing by allowing more parties than software-based ad hoc conferencing. Configuring multi-party ad hoc conferencing disables three-party ad hoc conferencing.
Three-Party Ad Hoc Conferencing	4.0	• End-of-conference options were introduced. • Phones connected in a three-way conference display "Conference."
	3.2.2	Conference gain control for external calls was introduced.
	3.2	Conference initiator drop-off control was introduced.
	2.0	Support for software-based conferencing was introduced.