Conferencing

Information About Conferencing

Conferencing allows three or more parties to join a telephone conversation. Unified CME offers conferencing functionality for the Unified phones and endpoints that it supports. Unified CME supports conferencing across the SIP and SCCP protocols. Also, the platforms Cisco Integrated Services Router Generation 2 and Cisco 4000 Series Integrated Services Routers support conferencing in Unified CME.

Note Cisco Cloud Services Routers (CSR) do not support DSP resources. As DSP resources are mandatory to support hardware conferencing in Unified CME, you cannot host hardware conferences in a CSR router.

Types of Conference

Based on the conferencing method, conferencing in Unified CME is of two types:

- Hardware Conference—Conferencing based on the Unified CME hardware and DSP resources. The types of hardware conferencing in Unified CME include:
    - Ad Hoc Hardware Conference
    - Meet Me Conference.

    - Connected Conference

- Software Conference—Software Conferencing is a three party conference that is hosted on the phone or on Unified CME. The types of software conferencing in Unified CME include:
    - Ad Hoc Software or Built-in Bridge (BIB) Conference (Supported on Unified IP Phones such as Cisco IP Phone 7800 Series and 8800 Series).
    - Three-Party Software Conference (For Unified CME, the support is only on Cisco Integrated Services Router Generation 2. For Cisco 4000 Series Integrated Service Routers, support is only for Unified SRST.)

The following table provides details on the support for various conferencing types in Unified CME:

Table 1: Types of Conference and Support in Unified CME

Conferencing Feature	Hardware-based		Software-based (Built-in Bridge)		Max Participants
	SIP	SCCP	SIP	SCCP	
Ad Hoc	Yes	Yes	Yes	No (Except 8900 Series Unified IP Phones)	• Ad Hoc (Hardware)—8 • Ad Hoc (Software)—3
Meet Me	Yes	Yes	No	No	32
Connected	Yes (Only for 7800 and 8800 Series Unified IP Phones)	Yes (Supported as Select and Join functionality for SCCP)	No	No	8
Three-party Software Conference	No	No	No	Yes	3

Note Three-party software conference is supported only on Cisco Integrated Services Router Generation 2 for Unified CME. Cisco 4000 Series Integrated Services Routers supports three-party software conference only for Unified SRST.

Hardware Conference

In a hardware-based conference, the conference is established using the hardware resources of a Unified CME system. This includes the routers and the Digital Signal Processors (DSPs.) From Unified CME Release 11.7, Cisco 4000 Series Integrated Services Routers support hardware conferencing.

Hardware-based conferencing uses the DSP resources in a router to perform audio mixing. The DSP resources used for conferencing take care of transcoding, and not just audio mixing. The participants of the conference can be IP phones that are connected to Unified CME or external callers. The external callers are the participants who join the conference call over TDM or SIP trunks. You must configure the DSP resources in a DSP farm for conferencing. Also, the DSP resources that are required for conferencing varies based on the codec complexity. For more information, see Configure the DSP Farm Profile, on page 23.

The following are the hardware-based conferencing models that are supported in Unified CME:

- Ad Hoc Hardware Conference
- Meet Me Conference
- Connected Conference

For information on the basic configurations that are required to enable a hardware conference, see Configure Hardware Conferencing, on page 19.

Ad Hoc Hardware Conference

Ad hoc conferences can be of two types:

- Hardware-based
- Software-based

Note For more information on Ad Hoc software conference, see Ad Hoc Software Conferencing, on page 9.

Ad Hoc conferences allow the conference host or participant to add new participants to the conference. Ad hoc conferences are created when one party calls another, then either party decides to add another party and turn the call into a conference. Hence, Ad Hoc conferencing is not predetermined, but a conference call that is created instantaneously. From Cisco Unified CME Release 11.7, Cisco 4000 Series Integrated Services Routers support Ad Hoc conferencing.

Hardware Ad Hoc Conference is a conference with minimum of three participants and a maximum of eight participants. Hardware-based Ad Hoc conference uses digital signal processors (DSPs) to allow more parties than software-based ad hoc conferences and provides extra features such as Join and Conference Participant List (ConfList). Unified CME manages the conference bridge by using the DSP resources available.

For an Ad Hoc hardware conference hosted on Unified CME:

- You need to configure **ephone-dn** as a placeholder directory number configuration for conference hosting.
- From Unified CME 11.7 onwards, conference participants (line or trunk) with different codecs can be added to the conference bridge without the need for configuring extra DSP resources for LTI-based transcoding. For more information, see Local Transcoding Interface (LTI) Based Transcoding.
- The conference bridge is established when minimum of three participants join the conference and becomes a point-to-point call when there are only two participants.
- Ad Hoc conference supports a mixed deployment of SIP and SCCP phones.
- An Ad Hoc conference supports ITSP or SIP trunk external party.

- Ad Hoc conference supports the ability to play join tone when a participant joins the conference, and leave tone when a participant drops from the conference.
- During a two-party transcoded call on Unified CME (Cisco 4000 Series Integrated Services Router), LTI-based transcoding is invoked. When the two-party call becomes an ad hoc conference, LTI-based transcoding is released, and SCCP-based DSP conference is invoked.
- The DSP inserted for conferencing takes care of both transcoding and mixing of the audio stream.
- For Unified CME 4.1 and earlier, support for ad hoc conferencing was limited to three participants—all participants on G.711 codec.
- You need to configure **max-participant** under **dspfarm** configuration mode to define the number of participants supported by an ad hoc conference.
- Hardware-based multi-party ad hoc conference bridges do not support video phones. In a scenario where the participants joins the conference with video enabled phones, the caller on that phone can connect to the conference as an audio only participant.
- When the participant puts the call on hold in a conference, the other parties in the conference remain connected. The Resume softkey is not displayed to the other active remote-in-use calls on the shared lines. Only, the participant who puts the call on hold can resume the call.
- The maximum number of conference parties you can support on a hardware conference call is limited to eight.
- You can setup an Ad Hoc hardware conference even if different codecs are configured on the conference parties.
- The transcoder is invoked when it is a point-to-point call and its released once the conference is setup. The conference bridge performs codec mixing.
- You need to configure **dspfarm** to support transcoding:

```
enable
configure terminal
dspfarm profile tag transcode universal
codec codec_type
maximum sessions <1-40>
associate application CUBE
no shutdown
end
```

Ad Hoc hardware conferences can be created in several ways. For example, you can configure the Ad Hoc conference in Unified CME, such that:

- Only the conference creator can add parties to the conference.
- Any participant can add new participants to the conference (default behavior for ad hoc conference).
- Conference drops when the creator hangs up.
- Conference drops when the last local party hangs up.
- The default behavior for termination of ad hoc conference is that the conference is not dropped provided three parties remain in the conference. It is regardless of whether the creator hangs up or not.

The maximum number of simultaneous conferences is 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 new conference on a phone if you already have an existing conference on hold.

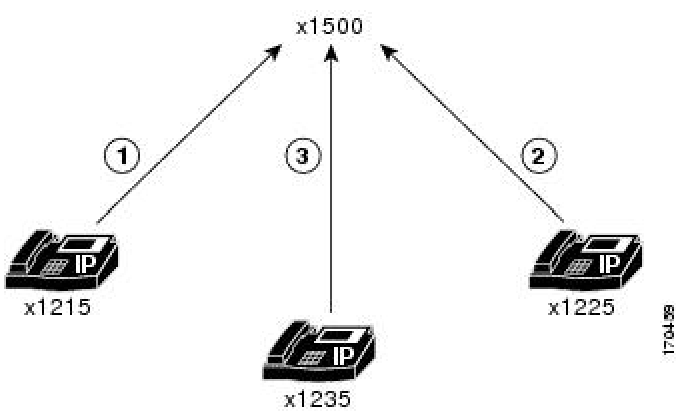
For information on configuration of Ad Hoc or Meet Me conferencing for SIP and SCCP phones, see Configure Ad Hoc or Meet Me Hardware Conference, on page 27

Meet Me Conference

Meet Me conferences consist of at least three parties dialing a Meet Me conference number. . The number is predetermined by the system administrator. Hence, it is not necessary for participants to dial another party to add them into the conference. The conference host uses the **MeetMe** softkey on the phone and dials the designated conference number to initiate the conference. The other participants can join the conference only when the conference host has initiated the conference.

For example, the conference shown in Figure 1: Simple Meet Me Conference Scenario, on page 5 is created when the conference creator at extension 1215 presses the **MeetMe** softkey 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 1: Simple Meet Me Conference Scenario



For a Meet Me Conference in Unified CME:

- Meet Me conference is supported only as a hardware-based conference.
- If you configure software-based conferencing, you cannot host Meet Me conferences.
- For a Meet Me conference configured for multiple ephone-dns with octo line configurations that use the same directory number, a maximum of 32 participants can join. The support for participants is based upon the configuration of DSP resources.
- You can configure the maximum number of conference parties to be lower than the actual maximum of 32 for Meet Me conferences. For more information, see Configure the DSP Farm Profile, on page 23.
- With octo-line ephone directory numbers, only one directory number is required for an eight-party Meet Me conference. Hence, you need four ephone octo-line directory numbers for 32 parties.
- The conference initiator presses **MeetMe** softkey before dialing the conference number. Other Meet Me conference parties (line or trunk) dials the conference number to join the conference.

- If only one party remains in the Meet Me conference, (For example, if one party has forgotten to hang up and other participants have left), the conference call is disconnected after five minutes to free system resources.
- If the creator is waiting for parties to join the conference (that is, only one party has joined the conference), the conference is not disconnected because significant resources are not being used.
- If only one party remains in the Meet Me conference, the conference call is disconnected after five minutes to free system resources.
- Maximum number of participants in a single conference with G.711 codec conference bridge is 32. For a single conference with G.729 codec conference bridge, the maximum numbe rof participants is 16.
- If Music on Hold (MOH) is configured for a conference party that puts the call on hold, the MOH is not played to the other conference. This is because other parties are in an active call.

For information on configuration of Ad Hoc or Meet Me conferencing for SIP and SCCP phones, see Configure Ad Hoc or Meet Me Hardware Conference, on page 27

Meet-Me Conferencing in Cisco Unified CME 11.7 and Later Versions

From Cisco Unified CME Release 11.7, Meet-Me conferencing is supported on Cisco 4000 Series Integrated Services Router.

Configuration of multi party conference on Cisco 4000 Series Integrated Services Routers for Unified CME Release 11.7 and later is same as that of previous releases. Also, the configuraton remains same across both SIP and SCCP phones. For more information, see Configure Hardware Conferencing, on page 19.

Connected Conference

Connected Conference supports Unified CME to host a conference for phones in connected call state. In a connected call scenario for SIP phones, a line on the phone is in an active call. The other lines are in held state. Using the Connected conference feature, you can allow one of the calls on hold to join the active call.

Note For Connected Conference to work on phones, you must enable Ad Hoc hardware conferencing in Unified CME.

Only Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series support Connected Conference.

Only one held call can join the active call at a time for SIP phones. If the other lines on the SIP phone have to join the conference, they can join one at a time.

Note Connected Conference supports a maximum of eight participants.

From Cisco Unified CME Release 11.7 onwards, Connected Conference feature is supported on SIP phones as well. As part of this enhancement, Unified CME introduced a new softkey **Active calls** for SIP phones.

For the Connected Conference feature, the behavior is different across Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series. Cisco IP Phone 7800 Series uses the line key for Connected Conference feature. However, Cisco IP Phone 8800 Series uses **Active calls** softkey.

Following are the steps to invoke connected conferencing on Cisco IP Phone 8800 Series:

1. A call from Phone A (Cisco IP Phone 8800 Series) is answered by Phone B.
2. Phone A puts the call with Phone B on hold.
3. Phone A makes another call to Phone C, and the call is answered by Phone C.
4. Press the **Conference** hard button or softkey on Phone A.
5. Then, press the **Active calls** softkey on Phone A to select the option Phone B.
6. Repeat the above steps to add more parties into conference.

A connected conference between Cisco IP Phone 8800 Series Phone A, Phone B, and Phone C is established.

Following are the steps to invoke connected conferencing on Cisco IP Phone 7800 Series:

1. A call from Phone A (Cisco IP Phone 7800 Series) is answered by Phone B.
2. Phone A puts the call with Phone B on hold.
3. Phone A makes another call to Phone C, and the call is answered by Phone C.
4. Use the line key on Phone A to select the option Phone B.
5. Repeat the preceding steps to add more parties into conference.

A connected conference between Cisco IP Phone 7800 Series IP Phone A, Phone B, and Phone C is established.

Note The phone firmware files that support Connected Conference on Cisco IP Phone 8800 Series is unavailable until the next Unified CME release. Hence, Connected Conference support for SIP phones is limited to Cisco IP Phone 7800 Series for Unified CME Release 11.7.

cBarge Conference

cBarge enables multiple phone users who share a directory number to join an active call on the shared line by pressing a softkey. cBarge facilitates a conference by invoking hardware conference on Unified CME. When the conference initiator barges into a call, hardware conference is created on Unified CME. The conference is established between the barge initiator, the target party, and the other parties connected in the call.

To support cBarge:

- Enable hardware conference
- Disable Privacy

If hardware conference is disabled, cBarge softkey invokes barge. Barge uses the built-in conference bridge on the target phone (the phone that is barged). Hence, a barge conference supports only up to three parties. Configure cBarge if you must support more participants.

Note Even if you have configured cBarge softkey, the softkey display on the phone is **Barge**.

The configurations for cBarge on the conference bridge of Unified CME are same as an Ad Hoc hardware conference, except:

- The configuration to enable cBarge softkey on phone in remote-in-use state.
- Configure **no privacy** under **voice register global**.

To configure softkey template to enable cBarge softkey on phone in remote-in-use:

```
      enable
configure terminal
voice register template <template-tag>
    softkeys remote-in-use {[ Barge ] [ Newcall ] [ cBarge ]}
exit
```

To associate softkey template with the pool:

```
voice register pool <phone-tag>
    template <template-tag>
end
```

To disable **privacy** and enable **conference hardware** under **voice register global** configuration mode:

```
voice register global
    no privacy
    conference hardware
    create profile
    reset
end
```

For more information on Barge and cBarge, see Barge and cBarge.

Drop Mode Conference

A person who initiates a conference call and hangs up can either keep the remaining parties connected or disconnect them. Based on this configuration option, Unified CME supports Drop Mode Conference as an End of Conference option for Hardware Conferencing.

To configure the mode for terminating hardware conferences when parties drop out, use the **conference drop-mode** and **conference add-mode** command in **ephone** or ephone-template configuration mode for SCCP phones. Configure **conference drop-mode** and **conference add-mode** command in **voice register** configuration mode for SIP phones.

The 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.

- For information on configuration of Drop Mode and Add Mode for hardware conferencing, see Configure Softkeys and End of Conference Options for Hardware Conferencing, on page 29
    - For more information on configuration of Add Mode and Drop Mode Conference for SCCP phones, see conference add-mode and conference drop-mode.
    - For more information on configuration of Add Mode and Drop Mode Conference for SIP phones, see conference add-mode (voice register) conference drop-mode (voice register).

Software Conference

Software conference can host a maximum of three participants. There are two types of software-based conferencing available in Unified CME:

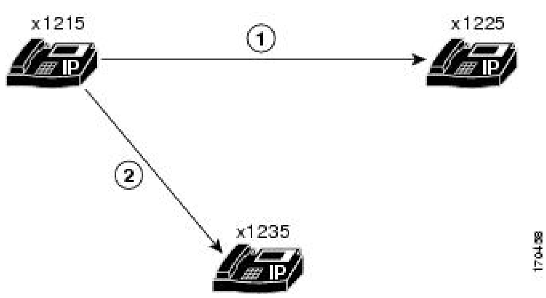
- Ad Hoc Software Conference—Ad Hoc Software Conference or Built-in Bridge Conference is established using the phone or endpoint hardware that provides audio mixing. There is no dependency upon the Unified CME router hardware for Ad Hoc software conferencing.
- Three-Party Software Conference—In a three-party software conference, Unified CME router supports conferencing for phones that do not support BIB-based conferencing (SCCP phones). When BIB conference is enabled, three-party software conference is disabled. It is supported only on Cisco Integrated Services Router Generation 2 and only for SCCP phones. For information on how to configure a three-party software conference, see Configure Three-Party Software Conference, on page 14.

Ad Hoc Software Conferencing

Ad Hoc software conference is also known as Built-in bridge (BIB) conferencing. Ad hoc software conferences do not depend on the Unified CME hardware to support conferencing. Press the conferencing softkey on the phone that hosts the conference bridge to enable the Ad Hoc software conference. In an Ad Hoc software conference, the phone that hosts the conference also performs audio mixing.

The conference that is shown in Figure 2: AdHoc Software Conference Using the Conference Softkey, on page 9 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 conference initiator. Hence, audio mixing happens in 1215.

Figure 2: AdHoc Software Conference Using the Conference Softkey



For a software-based Ad Hoc conference:

- The number of participants is limited to three parties.
- You do not need Unified CME hardware or DSP resources for audio mixing.
- The phone that hosts the conference performs audio mixing.
- Transcoding is not supported in a software-based conference call. Hence, you cannot host a software conference for calls with different audio codecs.

Software conference is enabled using softkeys on the Unified IP phones. The softkey varies depending on the phone model used. **confrn** and **conference** are some of the common softkeys for Software Conferencing in Unified IP Phones.

To configure a software conference, you have to disable hardware conferencing in Unified CME:

- Configure **no conference hardware** under **telephony service** for SCCP phones and no conference hardware under **voice register global** for SIP phones to disable hardware conference.
- Also, you must configure **create profile** under **voice register global** and **create cnf-files** under **telephony-service** configuration mode.

Keep Conference

A person who initiates a conference call and hangs up can either keep the remaining parties connected or disconnect them. Based on this configuration option, Unified CME supports Keep Conference as an End of Conference option for Software Conferencing.

Keep Conference is an end of conference option in Software Conferencing. With Keep Conference option, 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.

The 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.

- For information on configuration of Keep Conference for SCCP phones, see Configure Keep Conference for SCCP Phones, on page 15.

  For an example of Keep Conference for SCCP phones, see Example for Keep Conference Configuration on SCCP Phones, on page 36.

- For information on configuration of Keep Conference for SIP phones, see Configure Keep Conference Option for SIP Phones, on page 17.

  For an example of Keep Conference for SIP phones, see Example for Keep Conference Configuration on SIP Phones, on page 37.

Max Conference

You can set the maximum number of three-party software conferences that are supported simultaneously by the Unified CME router using Max Conference option. Configure the **max-conferences** command in **telephony-service** configuration mode to define maximum number of software conferences.

Note For Max Conference in Unified CME, the configuration is same for both SIP and SCCP phones.

For information on configuration of **max-conferences**, see Configure Three-Party Software Conference, on page 14.

For an example of Max conference, see Example for Configuration of Max Conference and Gain Levels, on page 36.

Conference Gain Levels

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.

Conference gain levels are set using the variable *gain* configured under the CLI command **max-conference** under **telephony-service** configuration mode. The Conference Gain Level configuration is consistent across all the hardware conferencing options supported in Unified CME. For more information, see Configure Three-Party Software Conference, on page 14.

For an example of Max conference, see Example for Configuration of Max Conference and Gain Levels, on page 36.

Design Considerations for Conferencing

The following are some of the characteristics of conferencing in Unified CME:

- The maximum number of conference participants that you can host in a conference is specific to the mode of conference. For more information, see Types of Conference, on page 1.
- Consider a scenario where the ad hoc hardware conference creator transfers the call or parks the call with another call. For Unified CME 11.7 and later releases, the conference bridge remains active, irrespective of whether you have enabled **drop-mode creator** CLI command or not.
- 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 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. Moderately complex configuration may exceed the default 256 MB of DRAM and require 512 MB of DRAM. Many factors contribute to memory usage, in addition to the number of dial peers and ephone-dns configured.
- **Secure Conferencing in Unified CME**—If Unified CME uses a conference DSP farm resource for Ad Hoc or Meet Me hardware conference, it can use a secure or nonsecure DSP farm resource. However, it is recommended that you pick a nonsecure DSP farm resource for Unified CME. This is because the conference itself cannot be secure in Unified CME. Also, you can avoid wastage of the session 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 Unified CME. Unified CME cannot use the DSP farm's secure capabilities.
- **LTI-based Transcoding**—From Unified CME 11.7 onwards, LTI-based transcoding is supported for hardware conferencing in Unified CME. With LTI-based transcoding, conference participants (line or trunk) with different codecs can be added to the conference bridge without configuring extra DSP resources. During a two-party transcoded call on Unified CME (Cisco 4000 Series Integrated Services Router), LTI-based transcoding is invoked. When the two-party call becomes an Ad Hoc conference, LTI-based transcoding is released and SCCP-based DSP conference is invoked. The DSP inserted for conferencing takes care of both transcoding and mixing the audio stream. For information about LTI-based conferencing and configuration, see Local Transcoding Interface (LTI) Based Transcoding and Configure LTI-based Transcoding.

- **Conference Blocking (Conference Pattern Blocked)**—To prevent extensions in an **ephone** or a **voice register pool** from initiating conferences, configure the **conference-pattern blocked** command . For more information, see Conference-Pattern Blocked and Configure Conference Blocking Options for Phones.
- **Conference Max Length**—When **conference max-length** command is configured, Unified CME allows the conferences only if the dialed digits are within the max-length limit. For more information on Conference Max-length and configuration, see Conference Max-Length and Configure the Maximum Number of Digits for a Conference Call.
- **Octo-line Directory Numbers**—With octo-line directory numbers, only one directory number is required for an eight-party Meet Me or Ad Hoc conference. An octo-line directory number supports up to eight active calls, both incoming and outgoing, in a single phone button. It supports up to eight Select and Join instances. When a conference initiator is an octo-line directory number, Unified CME selects an idle channel from that directory number. Establish a new call to complete the conference. If an idle channel is not available in the same octo-line directory number, the conference terminates and a **No Line Available** message displays.

Note If an idle channel is not available in the same octo-line directory number, Unified CME does not pick an idle channel from another directory number. Also, you cannot select **hold** calls in the other channels of the directory number or for other directory numbers. It is supported only for single-line and dual-line directory numbers.

Deploy the DSP Farm Resource with Unified CME

It is mandatory to have DSP farm resources to support hardware conferencing in Unified CME. For more information on configuration of DSP resources with Unified CME, see Configure Transcoding Resources.

You can deploy a DSP farm with Unified CME in two ways:

- Configure DSP Farm and Unified CME in the same router.

  For a sample configuration, see Example of DSP Farm and Cisco Unified CME on the Same Router, on page 37.

- Configure DSP Farm and Unified CME in different routers.

  For a sample configuration, see Example of DSP Farm and Cisco Unified CME on Different Routers, on page 47.

Softkeys for Conference Functions

For the conferencing functions that you configure on Unified CME, you have corresponding softkeys on the phone. The following soft keys provide conferencing functions for conferencing enhancements on your phone:

- 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, press **Update** to verify that a party has been removed from the conference. Press **Remove** softkey to remove the appropriate parties. The suboption **Remove** is available for the conference creator and phones that have **conference admin** configured.

- 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.
- 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 start a Meet Me conference.

In Cisco Unified CME 11.7 and later versions, the following softkeys are also supported.

- Details (Supported only on Cisco IP Phone 7800 Series)—Lists all the participants 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. Press **Remove** softkey to remove the appropriate parties. The suboption **Remove** is available to the conference creator and phones that have **conference admin** configured.
- Show detail (Supported only on Cisco IP Phone 8800 Series)—Lists all the participants 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. Press **Remove** softkey to remove the appropriate parties. The suboption **Remove** is available to the conference creator and phones that have **conference admin** configured.
- Active calls (Supported on Cisco IP Phone 8800 Series)—As part of the Connected Conference support on Unified CME 11.7 and later releases, a new softkey **Active calls** is introduced. The **Active calls** softkey is added to the SIP phones configured on Unified CME. **Active calls** softkey is used in Cisco IP Phone 8800 Series for Unified CME.

For more information on the configuration, see Configure Hardware Conferencing, on page 19.

Restrictions for Conferencing

- Unified CME does not support secure conferencing. All conference calls are nonsecure. This is because Unified CME cannot use the secure conference DSP farm capability.
- For a phone registered to Unified CME, you can support only one conference. If an existing conference is put on hold, you cannot create another conference.
- For calls having different audio codecs, you cannot host a hardware conference call without transcoding (DSPs).
- For calls having different audio codecs, you cannot host a software conference in Unified CME. The calls do not merge into a conference.
- A Software (BIB) conference does not support more than three parties.

- Cisco Jabber is supported only by hardware conferencing in Unified CME.
- At a time, only one held call can be selected to join the Connected conference for SIP phones.
- Each individual Unified IP phone can host a maximum of one conference at a time. You cannot support a new conference in a phone if you have a conference on hold.
- For cBarge, the conference type is listed as **Ad Hoc Barge** instead of Ad Hoc.
- For cBarge, Caller ID on phones in the Barge conference is displayed as **Barge** instead of **Conference**.
- Configurations, limitations and attributes associated with Connected Conference on Unified CME is same as that for Ad Hoc hardware conference.

Configure Software Conferencing

Configure Three-Party Software Conference

You can configure software conferencing on Unified CME as follows. To globally modify the default configuration and change any of the following parameters for three-party software conferencing, perform the following steps.

- The configuration **no conference hardware** is required to enable software conferencing on Unified CME and BIB conferencing on phones.
- Maximum number of simultaneous three-party software conferences that are 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.
- For Max Conference and Gain level in Unified CME, the configuration is consistent across SIP and SCCP phones.

Restriction

- When a three-way software conference is established, a participant cannot use call transfer to join the remaining conference participants to a different number.
- Three-party software 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.
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 that are 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.

Configure Keep Conference for SCCP Phones

- Keep Conference is supported only for BIB Conferencing.
- Keep Conference on SCCP is supported only for Cisco Integrated Services Router Generation 2.
- 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.

Before you begin

- 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 Configure Call Transfer and Forwarding.
- 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

	Command or Action	Purpose
		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 Generate Configuration Profiles for SIP Phones.

Configure Keep Conference Option for SIP Phones

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.

Before you begin

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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag* | OR **voice register template** *template-tag*
4. **keep-conference**
5. **voice register pool** *pool-tag*
6. **template** *template-tag*
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**voice register pool** *pool-tag* | OR **voice register template** *template-tag*
	Example:
	`Router(config)# voice register pool 3`
	OR
	`Router(config)# voice register template 3`
		Enters voice register pool or voice register template 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.
		• *template-tag*—Unique sequence number of the template to be applied to the SIP phone. Range is 1 to 10.

Step 4	**keep-conference**
	Example:
	`Router(config-register-pool)# keep-conference`
	OR
	`Router(config-register-temp)# 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.
		Note **keep-conference** command is configured under voice register template only if you configure voice register template command in the previous step.

Step 5	**voice register pool** *pool-tag* **Example:** `Router(config-register-temp)# voice register pool 1`	(Optional) Enters voice register pool configuration mode to set phone-specific parameters for SIP phones. **Note** This step is required only if you configure voice register template.
Step 6	**template** *template-tag* **Example:** `Router(config-register-pool)# template 1`	(Optional) Attaches the template tag configured to the voice register pool. **Note** This step is required only if you configure voice register template.
Step 7	**end** **Example:**	Exits to privileged EXEC mode.

	Command or Action	Purpose
	`Router(config-register-pool)#` **`end`**	

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 Generate Configuration Profiles for SIP Phones.

Configure Hardware Conferencing

Prerequisites

- The following configuration is applicable to all hardware conferencing types supported in Unified CME, including Meet Me and Ad Hoc conferencing.
- DSP resources are mandatory to support a hardware conference in Unified CME.

Restriction

- 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 Ad Hoc conferencing does not support the local-consult transfer method (**transfer-system local-consult** command).

Enable DSP Farm Services for a Voice Card

To enable DSP farm services for a voice card to support hardware 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:**	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
	`Router> enable`	
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.

Configure Join and Leave Tones

The Join and Leave configuration is applicable for:

- both SIP and SCCP phones in Unified CME.
- all hardware conferencing types supported in Unified CME, including Ad Hoc and Meet Me.

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.

	Command or Action	Purpose
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.

Configure SCCP Infrastructure for Conferencing in Unified CME

The SCCP Infrastructure configuration is applicable to:

- Both SIP and SCCP phones in Unified CME.
- All hardware conferencing types supported in Unified CME, including Ad Hoc and Meet Me.

To enable SCCP Infrastructure in 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-typeinterface-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-typeinterface-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.
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:**	Exits global configuration mode.

	Command or Action	Purpose
	`Router(config)# exit`	

Configure the DSP Farm Profile

The DSP Farm Profile is applicable to:

- Both SIP and SCCP phones in Unified CME.
- All hardware conferencing types supported in Unified CME, including Ad Hoc and Meet Me.

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**}	Specifies the codecs supported by a DSP farm profile.

	Command or Action	Purpose
	Example: `Router(config-dspfarm-profile)# codec g711ulaw`	**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 Enable DSP Farm Services for a Voice Card, on page 19.
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 Enable DSP Farm Services for a Voice Card, on page 19.
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.
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.

Associate Unified CME with a DSP Farm Profile

The steps to associate Unified CME with a DSP farm profile is applicable to:

- Both SIP and SCCP phones in Unified CME.
- All hardware conferencing types supported in Unified CME, including Ad Hoc and Meet Me.

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.
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.

Enable Hardware Conferencing

To allow hardware-based multi-party conferences with more than three parties, perform the following steps.

> **Note**
> - You cannot configure Hardware and Software conference simultaneously in Unified CME. Configuring multi-party hardware conference in Unified CME disables three-party Ad Hoc software conferencing.
> - This configuration is applicable to both SIP and SCCP phones in Unified CME.

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.
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.

	Command or Action	Purpose
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 section Associate Unified CME with a DSP Farm Profile , on page 24.
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.

Configure Ad Hoc or Meet Me Hardware Conference

The configuration steps are applicable to:

- Both SIP and SCCP phones in Unified CME.
- All hardware conferencing types supported in Unified CME.

To configure extension numbers for hardware 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.

Note For Meet Me conference to be enabled, you need to press the **MeetMe** softkey on the phone as well.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* **octo-line**
4. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
5. Enter one of the following commands:
   - **conference ad-hoc**
   - **conference 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.
Step 3	**ephone-dn** *dn-tag* **octo-line** **Example:** `Router(config)# ephone-dn 18 octo-line`	Enters ephone-dn configuration mode to configure an extension (ephone-dn) for a phone line. • Each ephone-dn can carry eight parties if it is configured as an octo 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.

	Command or Action	Purpose

Step 5	Enter one of the following commands:
	• **conference ad-hoc**
	• **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.

Configure Softkeys and End of Conference Options for Hardware Conferencing

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.

Restriction

- 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.

The steps to configure end of conference and softkeys for hardware conference is applicable:

- Only for SCCP phones in Unified CME.

  Note
  - For End of Conference option on SIP phones, you need to configure **conference add-mode** and **conference drop-mode** under **voice register** configuration mode. For more information, see Cisco Unified Communications Manager Express Command Reference.
  - For softkey configuration on SIP phones, you need to configure **softkeys** under **voice register template** configuration mode. For more information, see Cisco Unified Communications Manager Express Command Reference.

- For Ad Hoc and Meet Me hardware conferencing.

Before you begin

- 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:
  - Enable DSP Farm Services for a Voice Card, on page 19
  - Configure the DSP Farm Profile, on page 23
  - Associate Unified CME with a DSP Farm Profile , on page 24

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**]	Configures an ephone template for softkey display during the connected call stage.

	Command or Action	Purpose
	[**LiveRcd**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**] } **Example:** `Router(config-ephone-template)# softkeys connected Hold Trnsfer Park Endcall Confrn ConfList Join Select RmLstC`	• 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.
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 softkey 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 softkey 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 softkey 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 softkey 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:**	Enters ephone configuration mode to create and configure an ephone.

	Command or Action	Purpose
	`Router(config)# ephone 1`	
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 Generate Configuration Files for SCCP Phones.

Verify Conferencing

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.

Example:

```
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
```

Verify Hardware Conferencing

The CLI commands to troubleshoot hardware conferencing is applicable to:

- Both SIP and SCCP conference configurations in Unified CME.

Ad Hoc Hardware Conference

You can configure the following show commands to verify Ad Hoc hardwarwe conferencing:

- **show telephony-service conference hardware**
- **show dspfarm profile <profile number>**
- **show sccp**
- **show call active voice compact**
- **show call active voice brief**

The following is a sample output for **show telephony-service conference hardware** command.

```
Router#show telephony-service conference hardware
Conference   Type            Active  Max  Peak    Host         HostPhone   Last
                                                                 cur(initial)
==========================================================================================
A002         Ad-hoc              4      8     5    1111 sip1      1      (  1)   5555 sccp2
```

The following is a sample output for **show dspfarm dsp active** command.

```
Router#show dspfarm dsp active
SLOT   DSP VERSION  STATUS CHNL USE   TYPE    RSC_ID BRIDGE_ID PKTS_TXED PKTS_RXED

0/1    1   44.1.0   UP     1    USED  conf    1      498       3384      3329
0/1    1   44.1.0   UP     1    USED  conf    1      499       3383      1739
0/1    1   44.1.0   UP     1    USED  conf    1      500       3382      3384
0/1    1   44.1.0   UP     1    USED  conf    1      503       2899      671
0/1    1   44.1.0   UP     1    USED  conf    1      506       2525      1269
```

Meet Me Conference

You can configure the following show commands to verify Ad Hoc hardwarwe conferencing:

- **show sccp connection**
- **show ephone-dn conference**
- **show telephony-service conference hardware**
- **show dspfarm dsp active**
- **show call active voice compact**
- **Show voip rtp connections**

The following is a sample output for **show ephone-dn conference** command.

```
Router#show ephone-dn conference
type         active inactive  numbers
=================================
Meetme          4       28     5555
DN tags: 9, 10, 11, 12
```

The following is a sample output for **show telephony-service conference hardware** command.

```
Router#sh telephony-service conference hardware
Conference   Type            Active  Max  Peak    Host         HostPhone   Last
                                                                 cur(initial)
```

```
==========================================================================================
5555         Meetme                4     32    4    phone2 1002      2     (2)    1003 1003
```

The following is a sample output for **show dspfarm dsp active** command.

```
Router#show dspfarm dsp active
SLOT   DSP VERSION  STATUS CHNL USE   TYPE    RSC_ID BRIDGE_ID PKTS_TXED PKTS_RXED

0/1    4   44.2.0   UP     1    USED  conf    1      8         8574      8599
0/1    4   44.2.0   UP     1    USED  conf    1      10        8223      8250
0/1    4   44.2.0   UP     1    USED  conf    1      12        7724      7639
0/1    4   44.2.0   UP     1    USED  conf    1      14        7274      7299

Total number of DSPFARM DSP channel(s) 1
```

The following is a sample output for **show call active voice compact** command.

```
Router#show call active voice compact
 <callID>  A/O FAX T<sec> Codec       type        Peer Address       IP R<ip>:<udp>
 VRF
Total call-legs: 8
     68771 ANS     T301   g711ulaw    VOIP        P1002       10.0.0.1:22018

     68772 ORG     T302   g711ulaw    TELE        P5555
     68775 ANS     T295   g711ulaw    VOIP        P1004       10.0.0.2:22462

     68776 ORG     T296   g711ulaw    TELE        P5555
     68778 ANS     T286   g711ulaw    VOIP        P1001       10.0.0.3:31890

     68779 ORG     T287   g711ulaw    TELE        P5555
     68781 ANS     T278   g711ulaw    VOIP        P1003       10.0.0.4:31202

     68782 ORG     T279   g711ulaw    TELE        P5555
```

Verify Keep Conference

The following is a sample output for **show voice register tftp-bind** command.

```
Router#sh voice register tftp-bind
tftp-server url flash:/its/SEPE0D173E54508.cnf.xml alias SEPE0D173E54508.cnf.xml

    With keep-conference enabled in voice register pool or voice register
template
    Router#more flash:/its/SEPE0D173E54508.cnf.xml | sec cnf
    <cnfJoinEnabled>true</cnfJoinEnabled>

   With keep-conference disabled in both voice register pool and voice register
 template
    Router#more flash:/its/SEPE0D173E54508.cnf.xml | sec cnf
    <cnfJoinEnabled>false</cnfJoinEnabled>
```

Troubleshoot Conferencing

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

Step 2 Use the **debug ephone detail** command for SCCP calls in a software conference.

Step 3 Use the **debug ccsip all** command for SIP calls in a software conference.

Step 4 Use the **debug ephone hw-conference** command for SIP and SCCP calls in a hardware conference.

Configuration Examples for Conferencing

Example for Configuration of Max Conference and Gain Levels

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
```

Example for Keep Conference Configuration on SCCP Phones

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
```

Example for Keep Conference Configuration on SIP Phones

In the following example, extension 3555 initiates a three-way conference on SIP phones using keep-conference configured under **voice register pool** .

```
voice register dn 35
  number 3555

voice register pool 24
  number 1 dn 35
  keep-conference
```

Following is a sample configuration for keep-conference under **voice register template**.

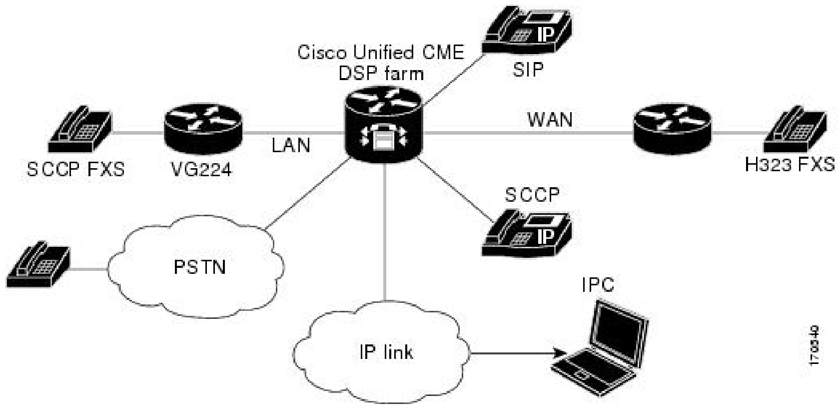
```
voice register template 24
  keep-conference

voice register pool 35
  template 24
```

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

In this example, the DSP farm and Cisco Unified CME are on the same router as shown in Figure 3: CME and the DSP Farm on the Same Router, on page 38.

Figure 3: 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
```

The following is an example of DSP Farm and Unified CME on the same router for SIP Phones.

```
Current configuration : 10821 bytes
!
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
service sequence-numbers
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
 address-family ipv6
exit-address-family
!
! card type command needed for slot/bay 0/1
no logging queue-limit
logging buffered 100000000
no logging rate-limit
no logging console
!
no aaa new-model
!
!
ipv6 unicast-routing
!
!
subscriber templating
!
!
multilink bundle-name authenticated
!
!
voice service voip
no ip address trusted authenticate
media disable-detailed-stats
allow-connections sip to sip
no supplementary-service sip refer
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
  registrar server expires max 240 min 60
!
!
voice register global
```

```
mode cme
source-address 8.39.23.16 port 5060
no privacy
timeouts interdigit 30
max-dn 40
max-pool 40
voicemail 9000
tftp-path flash:
create profile sync 0095202153430137
conference hardware
!
voice register dn  1
number 1001
name SIP Ph 1
!
voice register dn  2
number 1002
name SIP Ph 2
!
voice register dn  3
number 1003
name SIP Ph 3
!
voice register template  1
softkeys idle  HLog Mobility Newcall Pickup Redial
softkeys ringIn  Answer DND
softkeys connected  ConfList Confrn Endcall Hold Mobility Park Trnsfer
softkeys remote-in-use  Barge Newcall cBarge
!
voice register pool  1
busy-trigger-per-button 10
id mac B000.B4BA.F3DA
type 8851
number 1 dn 1
template 1
dtmf-relay rtp-nte
username xxxx password xxxx
codec g711ulaw
no vad
!
voice register pool  2
busy-trigger-per-button 10
id mac 1CE8.5DC9.C054
type 8851
number 1 dn 2
template 1
dtmf-relay rtp-nte
username xxxx password xxxx
codec g711ulaw
no vad
!
voice register pool  3
busy-trigger-per-button 10
id mac 00AF.1F9D.FB9F
type 8841
number 1 dn 3
template 1
dtmf-relay rtp-nte
username xxxx password xxxx
codec g711ulaw
no vad
!
!
voice translation-rule 1
```

```
rule 1 /^1234/ /301/
!
voice translation-rule 4
rule 4 /^1(..)$/ /51237812\1/
!
!
voice translation-profile PSTN_Callforwarding
translate redirect-target 4
!
voice translation-profile cmein
translate called 1
!
!
voice-card 0/1
dsp services dspfarm
!
restconf
!
username xxxx password xxxx
!
redundancy
mode none
!
!
threat-visibility
!
!
interface GigabitEthernet0/0/0
ip address 8.39.23.16 255.255.0.0
negotiation auto
!
interface GigabitEthernet0/0/1
ip address 10.64.86.106 255.255.0.0
shutdown
media-type rj45
negotiation auto
ipv6 address 2001:420:54FF:13::312:55/119
ipv6 enable
!
interface GigabitEthernet0/0/2
no ip address
shutdown
negotiation auto
!
interface GigabitEthernet0/0/3
no ip address
shutdown
negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http secure-port 8443
ip tftp source-interface GigabitEthernet0/0/1
ip tftp blocksize 8192
```

```
ip dns server
ip rtcp report interval 65535
ip route 0.0.0.0 0.0.0.0 8.39.0.1
ip route 8.0.0.0 255.0.0.0 8.39.0.1
ip route 202.153.144.0 255.255.255.0 8.39.0.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
!
!
tftp-server bootflash
tftp-server flash:vc488xx.12-0-1MN-113.sbn
tftp-server flash:sip88xx.12-0-1MN-113.loads
tftp-server flash:sb288xx.BE-01-020.sbn
tftp-server flash:kern88xx.12-0-1MN-113.sbn
tftp-server flash:fbi88xx.BE-01-010.sbn
tftp-server flash:rootfs88xx.12-0-1MN-113.sbn
!
!
ipv6 access-list preauth_v6
permit udp any any eq domain
permit tcp any any eq domain
permit icmp any any nd-ns
permit icmp any any nd-na
permit icmp any any router-solicitation
permit icmp any any router-advertisement
permit icmp any any redirect
permit udp any eq 547 any eq 546
permit udp any eq 546 any eq 547
deny ipv6 any any
!
control-plane
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
sccp local GigabitEthernet0/0/0
sccp ccm 8.39.23.16 identifier 1 version 7.0
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register conf-moto
!
!
!
telephony-service
sdspfarm units 2
sdspfarm tag 1 conf-moto
no privacy
conference hardware
no auto-reg-ephone
max-ephones 40
max-dn 40
ip source-address 8.39.23.16 port 2000
service phone sshAccess 0
service phone webAccess 0
service directed-pickup gpickup
```

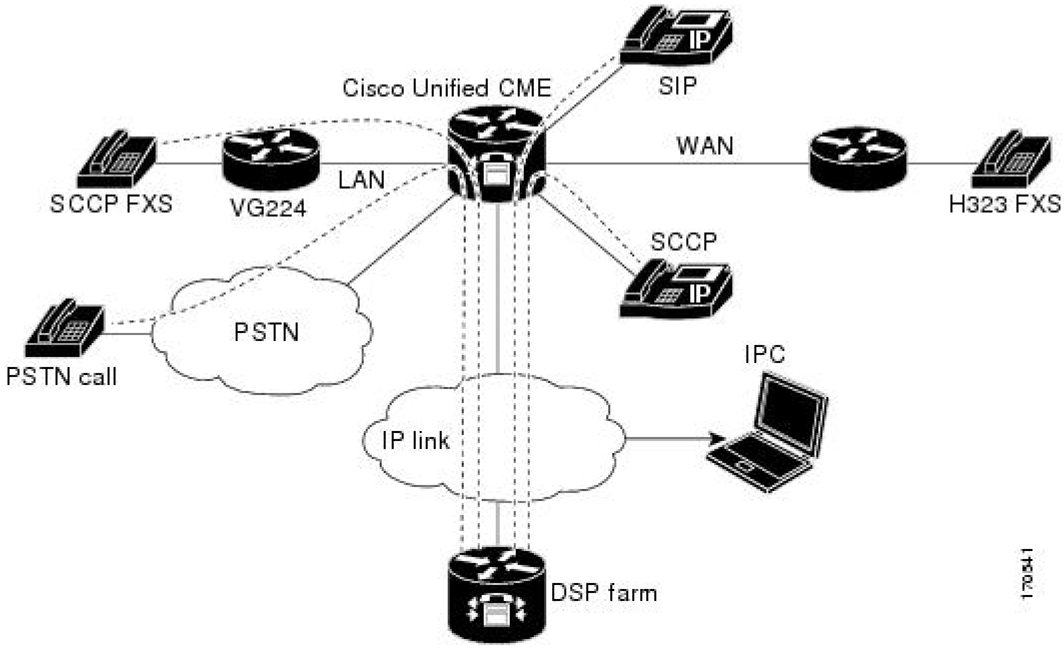
```
max-conferences 8 gain -6
call-park system application
hunt-group logout HLog
moh enable-g711 "flash:/scripts/en_bacd_music_on_hold.au"
transfer-system full-consult
fac standard
create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
dspfarm profile 2 transcode universal
 codec g729abr8
codec g729ar8
codec g711alaw
codec g711ulaw
codec g729br8
maximum sessions 2
associate application CUBE
!
dspfarm profile 1 conference
 codec g729br8
codec g729r8
codec g729abr8
codec g729ar8
codec g711alaw
codec g711ulaw
maximum sessions 2
associate application SCCP
!
dial-peer voice 1 voip
destination-pattern 20..
session protocol sipv2
session target ipv4:8.39.24.41
dtmf-relay rtp-nte
!
!
gateway
 media-inactivity-criteria all
timer receive-rtcp 1000
timer receive-rtp 1200
!
sip-ua
 mwi-server ipv4:8.41.24.7 expires 3600 port 5060 transport udp unsolicited
presence enable
!
!
ephone-dn  1  octo-line
number 1006
!
!
ephone-dn  2  octo-line
number 1007
!
!
ephone-dn  3  octo-line
number 1008
!
!
ephone-dn  4  octo-line
number 1009
!
!
ephone-dn  5  octo-line
number A001
conference ad-hoc
```

```
!
!
ephone-dn  6  octo-line
number A002
conference ad-hoc
!
!
ephone  1
device-security-mode none
mac-address 9876.0000.0006
type 7975
button  1:1
!
!
!
ephone  2
device-security-mode none
mac-address 9876.0000.0007
type 7975
button  1:2
!
!
!
ephone  3
device-security-mode none
mac-address 9876.0000.0008
type 7975
button  1:3
!
!
!
ephone  4
device-security-mode none
mac-address 9876.0000.0009
type 7975
button  1:4
!
!
alias exec poolall show voice register pool all brief
!
line con 0
transport input none
stopbits 1
speed 115200
line aux 0
stopbits 1
line vty 0 4
password xxxx
login local
transport input telnet
!
no network-clock synchronization automatic
!
end
```

Example of DSP Farm and Cisco Unified CME on Different Routers

In this example, the DSP farm and Cisco Unified CME are on different routers as shown in Figure 4: Cisco Unified CME and the DSP Farm on Different Routers, on page 48.

Figure 4: Cisco Unified CME and the DSP Farm on Different Routers



This section contains configuration examples for the following routers:

- Example of Cisco Unified CME Router Configuration, on page 48
- Example of DSP Farm Router Configuration, on page 55

Example of Cisco Unified CME Router Configuration

```
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
```

Example of DSP Farm Router Configuration

```
    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
```

Example for Verification of Meet Me Conference

The following partial output from the **show running-config** command shows the configuration on a Cisco 2821 router with 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&-immediate-start
ds0-group 2 timeslots 2 type e&-immediate-start
ds0-group 3 timeslots 3 type e&-immediate-start
ds0-group 4 timeslots 4 type e&-immediate-start
ds0-group 5 timeslots 5 type e&-immediate-start
ds0-group 6 timeslots 6 type e&-immediate-start
ds0-group 7 timeslots 7 type e&-immediate-start
ds0-group 8 timeslots 8 type e&-immediate-start
ds0-group 9 timeslots 9 type e&-immediate-start
ds0-group 10 timeslots 10 type e&-immediate-start
ds0-group 11 timeslots 11 type e&-immediate-start
ds0-group 12 timeslots 12 type e&-immediate-start
ds0-group 13 timeslots 13 type e&-immediate-start
ds0-group 14 timeslots 14 type e&-immediate-start
ds0-group 15 timeslots 15 type e&-immediate-start
ds0-group 16 timeslots 16 type e&-immediate-start
ds0-group 17 timeslots 17 type e&-immediate-start
ds0-group 18 timeslots 18 type e&-immediate-start
ds0-group 19 timeslots 19 type e&-immediate-start
ds0-group 20 timeslots 20 type e&-immediate-start
ds0-group 21 timeslots 21 type e&-immediate-start
ds0-group 22 timeslots 22 type e&-immediate-start
ds0-group 23 timeslots 23 type e&-immediate-start
ds0-group 24 timeslots 24 type e&-immediate-start
!
controller T1 0/3/1
 framing esf
 clock source internal
 linecode b8zs
ds0-group 1 timeslots 1 type e&-immediate-start
ds0-group 2 timeslots 2 type e&-immediate-start
ds0-group 3 timeslots 3 type e&-immediate-start
ds0-group 4 timeslots 4 type e&-immediate-start
ds0-group 5 timeslots 5 type e&-immediate-start
ds0-group 6 timeslots 6 type e&-immediate-start
ds0-group 7 timeslots 7 type e&-immediate-start
ds0-group 8 timeslots 8 type e&-immediate-start
ds0-group 9 timeslots 9 type e&-immediate-start
ds0-group 10 timeslots 10 type e&-immediate-start
ds0-group 11 timeslots 11 type e&-immediate-start
ds0-group 12 timeslots 12 type e&-immediate-start
ds0-group 13 timeslots 13 type e&-immediate-start
ds0-group 14 timeslots 14 type e&-immediate-start
ds0-group 15 timeslots 15 type e&-immediate-start
```

```
ds0-group 16 timeslots 16 type e&-immediate-start
ds0-group 17 timeslots 17 type e&-immediate-start
ds0-group 18 timeslots 18 type e&-immediate-start
ds0-group 19 timeslots 19 type e&-immediate-start
ds0-group 20 timeslots 20 type e&-immediate-start
ds0-group 21 timeslots 21 type e&-immediate-start
ds0-group 22 timeslots 22 type e&-immediate-start
ds0-group 23 timeslots 23 type e&-immediate-start
ds0-group 24 timeslots 24 type e&-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& ports 0/1/x. The E& 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
!
.
.
.
```

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 Templates.

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 Customize Softkeys.

Feature Information for Conferencing

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 2: Feature Information for Conferencing

Feature Name	Cisco Unified CME Version	Feature Information
Meet-me Conference	11.7	Added support for hardware-based Meet-me Conference on Cisco 4000 Series Integrated Services Router.
	4.1	Added support for hardware-based meet-me conferences created by parties calling a designated conference number.
Multi-party Ad Hoc Conference	11.7	Added support for hardware-based Multi-party Conference on Cisco 4000 Series Integrated Services Router.
	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 Conference	11.7	Added support for three-party Ad Hoc conference on Cisco 4000 Series Integrated Services Router.
	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.