Configuring Voice and Video Hardware Conferencing

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This chapter describes voice and video conferencing support for Cisco Unified SIP IP phones in Cisco Unified Communications Manager Express (Cisco Unified CME).

Finding Feature Information in This Module

Your Cisco Unified CME version may not support all of the features documented in this module. For a list of the versions in which each feature is supported, see the "Feature Information for Voice and Video Hardware Conferencing" section on page 1013.

Contents

Information About Voice and Video Hardware Conferencing

To enable voice and video hardware conferencing for Cisco Unified SIP IP phones in Cisco Unified CME, you should understand the following concepts:

Hardware Conferencing for Cisco Unified SIP IP Phones

In Cisco Unified CME 9.0 and later versions, the SIP line side hardware conference support enables a Cisco Unified SIP IP phone to act as the creator of ad-hoc and meet-me conferences with audio and video conferencing streams flowing from participating IP phones through the Cisco Unified CME.

Cisco Unified 7906, 7911, 7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, 7975, 8961, 9951, and 9971 SIP IP Phones can be conference creators that automatically initiate hardware conference calls.

Voice Hardware Conferencing

Voice hardware conferencing requires provisioning a DSPFarm voice profile with the appropriate voice codecs.

The voice hardware conference resource is allocated and released whenever a participant joins and leaves a conference call.

Table 30 lists the limit on the number of participants for voice hardware conferencing.

Table 30 Maximum Number of Voice Hardware Conferencing Participants

Voice Hardware Conference	Maximum Number of Participants
meet-me conferences	32
ad-hoc conferences	16

Video Hardware Conferencing

Video hardware conferencing requires provisioning a heterogeneous DSPFarm video profile with the appropriate video codecs and resolutions.

Once the video profile is available, the SIP line side conference creator can add any type of video-capable IP phone (Cisco Unified 7906, 7911, 7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, 7975, 8961, 9951, and 9971 IP Phones) to create an ad-hoc or meet-me video conference.

For other participants, Cisco Unified Video Advantage (CUVA) and Cisco Unified 7985 SCCP IP Phones are also supported.

The video hardware conference resource is allocated and released whenever a participant joins and leaves a conference call.

Table 31 lists the limit on the number of participants for video hardware conferencing.

Table 31 Maximum Number of Video Conferencing Participants

Video Conference	Maximum Number of Participants
meet-me conferences	16
ad-hoc conferences	16

In Cisco Unified CME 9.0 and later versions, voice-activated (loudest speaker) video switching is supported.

Meet-Me Hardware Conferencing

A meet-me conference is a scheduled conference and takes place when the conference creator goes offhook, presses the MeetMe soft key or feature button, and dials the meet-me conference directory number (DN). Participants can then dial the meet-me DN to join and connect to the conference bridge. The phone’s display shows the meet-me DN as the remote party ID.

With the unlock feature in Cisco Unified CME 9.0 and later versions, there is no need to press the MeetMe soft key to initiate voice or video conferencing and any Cisco Unified SIP IP phone can initiate the call.

Meet-me conferences are straightforward. The conference creator explicitly chooses to make a voice or video call by dialing a voice or video DN.

Ad-Hoc Hardware Conferencing

For Cisco Unified CME 9.0 and later versions, Cisco Unified SIP IP phones act as ad-hoc conference creators while Cisco Unified SIP or Cisco Unified SCCP IP phones act as the participants.

Ad-hoc conference calls are unscheduled conferences and occur when the conference creator adds a third party into the call. However, only consultative conferences, where the creator commits after the consultative party is connected, are supported in these conference calls.

The creator may add participants to the conference until the maximum number of participants is reached.

If the conference is configured to stay, the conference will fall back to a point-to-point call and the conference bridge resource is released when participants leave the conference, leaving only two parties.

Voice (or audio, to avoid confusion in the table) and video calls are connected to an ad-hoc conference bridge based on the following rules. See Table 32 for examples.

- Cisco Unified CME reserves a video conference DN if:
  - Either the primary or consultative call is a video call (See rows 3 and 4.)
  - Both the primary and the consultative calls are video calls (See row 2.)

  Otherwise, an audio conference DN is reserved. (See row 1.)
- When the conference creator's primary and consultative calls have different capabilities, regardless of which one is audio, the creator's call results in a video call to the conference bridge. (See rows 3 and 4.) The audio conference call leg is disconnected.
- Audio conference calls remain audio and cannot be upgraded into video calls. When a conference creator has one video conference call, the conference DN and the resulting conference call are video. However, when the creator has two active conferences using different media, the creator's video conference with one set of participants does not affect the creator's audio conference with the other set of participants.

  In row 4, the primary call between A and B is video and the consultative call between A and C is audio. Although C is in a call with the video-capable creator A, C remains an audio call and does not upgrade into a video call connecting to the video conference bridge.
- There is no mid-call media renegotiation for all parties after the primary and consultative calls are established. The call capabilities do not change after they are redirected to the bridge.

Table 32 Voice (Audio)/Video Ad-Hoc Conferencing

Primary Call (A and B)	Consultative Call (A and C)	Conference DN	Resulting Conference (A)	Resulting Conference (B)	Resulting Conference (C)
audio	audio	audio	audio	audio	audio
video	video	video	video	video	video
audio	video	video	video	audio	video
video	audio	video	video	video	audio

In row 2 of Table 32, when a conference is initiated from a video-enabled IP phone, the conference call is video.

However, in row 4, when a video-enabled IP phone initiates a conference with an audio-only phone, the conference remains audio.

Row 4 of Table 33 shows that when audio-only IP phone A initiates a consultative call with video-enabled IP phone D and D joins the active ad-hoc conference as a fourth participant, the resulting conference remains audio.

Table 33 Adding a Fourth Participant to an Active Ad-Hoc Conference

Active Conference	Consultative Call (A and D)	Result
audio	audio	D joins with audio
video	video	D joins with video
video	audio	D joins with audio
audio	video	D joins with audio

For information on configuring hardware conferences, see the "SIP: Configuring Hardware Conferencing" section on page 1006.

Conference List

During a conference, any of the participants can press the ConfList soft key to display a list of the current participants of the conference call but only the creator or administrator can remove a participant from the list. Once the conference list is displayed, the creator or administrator can navigate through the list and remove any participant by pressing the Remove soft key.

The Remove soft key is useful in removing an unidentified participant or a conference participant who does not display a caller ID.

The conference list is static and you need to press the Update soft key to refresh the list to reflect the addition or removal of a participant. Aside from the soft key, you can also enable the feature through programmable line keys (PLK).

Note Cisco Unified 8961, 9951, and 9971 SIP IP Phones enable the Show Detail soft key to perform the conference list function when an active conference call is detected. However, the conference list cannot be displayed when a Cisco Unified 8961, 9951, or 9971 SIP IP Phone is on an active video conference because the Show Detail soft key is not enabled.

Remove Last Conference Participant

During an ad-hoc conference, the creator can press the RmLstC soft key or use the Remove Last Participant PLK to drop the last added participant.

This feature is not relevant to meet-me conference calls and is only available to conference creators and system administrators.

Supplementary Services Interaction

In Cisco Unified CME 9.0 and later versions, hardware conferencing supports Hold and Resume on SIP-only shared lines but not on mixed SIP-SCCP shared lines.

How to Configure Voice and Video Hardware Conferencing

- SIP: Configuring Hardware Conferencing, page 1006

SIP: Configuring Hardware Conferencing

To configure hardware conferencing on Cisco Unified SIP IP phones, perform the following steps.

Prerequisites

- Cisco Unified CME 9.0 and later versions are configured to enable hardware conferencing using its DSP resource.
- For Cisco Unified 8961, 9951, and 9971 SIP IP Phones, the correct firmware (9.2.1 or a later version) is installed for version negotiation enhancement.
- DSPFarm voice conference profile is configured for voice hardware conferencing.
- DSPFarm video conference profile is configured for video hardware conferencing.
- Correct codecs are defined in the conference profiles.

Restrictions

For Voice and Video Hardware Conferencing:

- Daisy chaining of conference calls is not supported.
- Secure conference calls are not supported.
- Hold and Resume are supported on SIP-only shared lines but not on a mixed SIP-SCCP shared line.
- Early and connected hardware conferences are not supported.
- Hardware conferencing on Cisco Unified SIP IP phones is not supported in Cisco Unified SRST.

For Video Hardware Conferencing Only:

- Only Loudest Speaker mode for video switching is supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **conference hardware**
5. **exit**
6. **voice register global**
7. **conference hardware**
8. **exit**
9. **voice register pool** *pool-tag*
10. **conference admin**

11. **conference add-mode** [**creator**]
12. **conference drop-mode** [**creator** | **local**]
13. **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 hardware conferencing only.
Step 5	**`exit`** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 6	**`voice register global`** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set global parameters for all supported Cisco Unified SIP IP phones in Cisco Unified CME.
Step 7	**`conference hardware`** **Example:** `Router(config-register-global)# conference hardware`	Configures Cisco Unified CME DSPFarm hardware-based ad-hoc conferencing.
Step 8	**`exit`** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 9	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 50`	(Optional) Enters voice register pool configuration mode to set phone-specific parameters for a Cisco Unified SIP phone in Cisco Unified CME. • *pool-tag*—Unique number assigned to the pool. Range: 1 to 100.

	Command or Action	Purpose
Step 10	**conference admin** **Example:** `Router(config-register-pool)# conference admin`	(Optional) Assigns a Cisco Unified SIP IP phone as the hardware conference administrator.
Step 11	**conference add-mode** [**creator**] **Example:** `Router(config-register-pool)# conference add-mode creator`	(Optional) Configures the mode for adding participants to ad-hoc hardware conferences on Cisco Unified SIP IP phones. Default is that the conference creator or any of the participants can add a new participant. • **creator**—Specifies that only the conference creator can add participants to an ad-hoc hardware conference.
Step 12	**conference drop-mode** [**creator** | **local**] **Example:** `Router(config-register-pool)# conference drop-mode local`	(Optional) Specifies who can terminate an active hardware conference by hanging up. Default is that an active conference is never dropped. • **creator**—The active conference is terminated when the conference creator hangs up. • **local**—The active conference is terminated when the last local participant hangs up or drops out of the conference.
Step 13	**end** **Example:** `Router(config-register-pool)# end`	Exits to privileged EXEC mode.

Troubleshooting Tips

Use debug commands to display debugging information on hardware conferencing.

The following example is a partial output from the **debug ccsip event**, **debug ephone hw-conference**, and **debug ephone mtp** commands. The **debug ccsip event** command enables the collection of events that are specific to service provider interface (SPI) for debugging purposes. The **debug ephone hw-conference** command enables the collection of debugging information on hardware conferencing for Cisco Unified SCCP IP phones. The **debug ephone mtp** command enables Message Transfer Part (MTP) debugging.

```
Mar 17 22:08:21.867: //21/E2793E678024/SIP/Info/ccsip_indicate_rt_packet_stats: Processing
stats for callid=21, proc_id=9
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/HandleUdpIPv4SocketReads: Msg enqueued for
SPI with IP addr: [1.5.40.97]:49706, local_address:[ - ]
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/ccsip_process_sipspi_queue_event:
ccsip_spi_get_msg_type returned: 2 for event 1
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/ccsip_new_msg_preprocessor: Checking
Invite Dialog
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/sipSPIAddContextToTable: Added
context(0x2995490) with key=[28] to table
Mar 17 22:08:22.027: //-1/000000000000/SIP/Info/ccsip_offer_ans_init:
Mar 17 22:08:22.027: //-1/000000000000/SIP/Info/ccsip_iwf_init:
Mar 17 22:08:22.027: //-1/000000000000/SIP/Info/ccsip_ipip_media_service_init:
Mar 17 22:08:22.027: //-1/000000000000/SIP/Info/sipSPI_ipip_vcc_Initialization:  Entry...
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/resolve_sig_ip_address_to_bind: calling
reg_invoke_ip_first_hop()
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/resolve_sig_ip_address_to_bind: calling
ip_best_local_address()
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/resolve_sig_ip_address_to_bind: return
addr 1.5.40.20
Mar 17 22:08:22.027: //-1/xxxxxxxxxxxx/SIP/Info/sipSPISetDateHeader: Clock Time Zone is
UTC, same as GMT: Using GMT
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Mar 17 22:08:22.031: //-1/000000000000/SIP/Info/sipSPILineControlMsg: nSS_CONF_REQUEST
Mar 17 22:08:22.031: //-1/000000000000/SIP/Info/sipSPILineConfReq: sipSPILineConfReq
oodr_ccb 2995490, CallID -1, scb 14886244, sub_id -1, subscriptionID 309306146
Mar 17 22:08:22.031: //-1/000000000000/SIP/Info/sipSPILineConfReq: ccb CallID 16,
consult_ccb CallID 21
Mar 17 22:08:22.031: sipSPILineConfReq oodr_ccb 2995490, ccCallID -1, scb 14886244, sub_id
-1, subscriptionID 309306146
Mar 17 22:08:22.031: ccb CallID 16, consult_ccb CallID 21Process the conference request,
peer 17

Mar 17 22:08:22.031: Check associated hwconf with callid 17
Mar 17 22:08:22.031: skinny_hwcfb_check_resource: for Audio
Mar 17 22:08:22.031:  skinny_hwcfbi_find_next_mtpcb Got MTPCB conference type req 52:Audio
only as Audio only Pass
Mar 17 22:08:22.031: skinny_hwcfb_get_adhoc_number:
Mar 17 22:08:22.031: skinny_hwcfbi_find_conferenceId_by_number: this is not an active conf
number A000
Mar 17 22:08:22.031: //-1/xxxxxxxxxxxx/SIP/Info/sipSPIAddContextToTable: Added
context(0x29A26F0) with key=[30] to table
Mar 17 22:08:22.031: //0/000000000000/SIP/Info/ccsip_new_scb: Created new scb: 0x29A26F0
with id: -1
Mar 17 22:08:22.031: //-1/xxxxxxxxxxxx/SIP/Event/ccsip_spi_refer_client: Queued event from
SIP SPI : SIPSPI_EV_CC_REFER
Mar 17 22:08:22.031: //16/DC08AF52801D/SIP/Info/sip_conf_holdretrieve:
hold retrieve request  message sent to 2870@1.5.40.97
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Mar 17 22:08:22.031: //-1/xxxxxxxxxxxx/SIP/Event/sipSPIEventInfo: Queued event from SIP
SPI : SIPSPI_EV_CC_REFER_RESP
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Mar 17 22:08:22.035: SkinnyHWConfAPI: reqType 46:MTP for CAP list
Mar 17 22:08:22.035: skinny_hwconf_preselect_mtp_tag_for_video:DN not video  cdn 2 cchan 1
Mar 17 22:08:22.035: SkinnyHWConfAPI: reqType 46:MTP for CAP list
Mar 17 22:08:22.035: skinny_hwconf_preselect_mtp_tag_for_video:DN not video  cdn 3 cchan 1
Mar 17 22:08:22.035: SkinnyHWConfAPI: reqType 46:MTP for CAP list
Mar 17 22:08:22.035: skinny_hwconf_preselect_mtp_tag_for_video:DN not video  cdn 11 cchan
1
Mar 17 22:08:22.035: SkinnyHWConfAPI: reqType 46:MTP for CAP list
Mar 17 22:08:22.035: skinny_hwconf_preselect_mtp_tag_for_video:DN not video  cdn 11 cchan
2
Mar 17 22:08:22.035: SkinnyHWConfAPI: reqType 25:State Update
Mar 17 22:08:22.035: SkinnyUpdateHWConfState: dn 11 chan 1 phone 1 pdn 2 pchan 1 state 7:
Mar 17 22:08:22.035: skinny_hwcfbi_find_conferenceId_by_number: this is not an active conf
number A000
Mar 17 22:08:22.035: skinny_hwconf_admit_conf_call pdn 2 pchan 1 adhoc 1, confID 0x0,
number A000, cdn 11 cchan 1
Mar 17 22:08:22.035: skinny_hwcfbi_confId_to_confBlk: invalid conference id 0x0
Mar 17 22:08:22.039: skinny_hwconf_admit_conf_call:SK = 0, ConfId=0, Master = 0
Mar 17 22:08:22.039: skinny_hwconf_call_open:New conference dn 11 chan 1 lpcor_index 0
Mar 17 22:08:22.039: skinny_hwcfbi_find_conferenceId_by_number: this is not an active conf
number A000
Mar 17 22:08:22.039: skinny_hwconf_call_open:New conference dn 11 chan 1 max_party 8
number A000 conference category 5:Ad-Hoc Audio
Mar 17 22:08:22.039: skinny_hwcfb_get_new_conferenceId: codec 0, parties 8, numStr A000
Mar 17 22:08:22.039: skinny_hwcfbi_find_conferenceId_by_number: this is not an active conf
number A000
Mar 17 22:08:22.039: skinny_hwcfbi_alloc: allocate 154 conf blocks
Mar 17 22:08:22.039: SkinnyHwconfConfSKButtonSet:for dn 2 chan 1 set=0 conf_sk=0
Mar 17 22:08:22.039: skinny_hwconf_call_open: Master=0 party 2 chan 1 lpcor_index 0
Mar 17 22:08:22.039: skinny_hwcfb_get_new_streamId: confId C0010001, codec 0, master FALSE
Mar 17 22:08:22.039: skinny_hwconf_set_preferred_codec: Conf 11:1 Phone 2:1 - 1
Mar 17 22:08:22.039: skinny_hwconf_get_dn_supported_codec: CONF-0: supported codec 980B
Mar 17 22:08:22.039: skinny_hwconf_is_codec_supported: codec 5 is   supported
Mar 17 22:08:22.039: skinny_hwconf_preselect_phone_codec Phone supports the codec
Mar 17 22:08:22.039: skinny_hwconf_set_preferred_codec: Conf 11:1 Pdn 2:1 Phone 1 codec 5
Supported
Mar 17 22:08:22.039: SkinnyUpdateHWConfState for DN 11 chan 1 set ring timer
Mar 17 22:08:22.039: SkinnySetHWConfGcid: dn 11 chan 1 phone 1 pdn 2 pchan 1
SkinnySetHWConfGcid: The first party in the conf[C0010001], set confgcid succeed
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Mar 17 22:08:22.051: //-1/xxxxxxxxxxxx/SIP/Info/ccsip_event_handler: switch(ev.ev_id: 171)
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_event_handler: Current mode is
SIP-TDM
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_iwf_handle_peer_event:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_iwf_map_ccapi_event_to_iwf_event:
Event Category: 1, Event Id: 171
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_iwf_process_event:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_sip_ed:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_sip_md:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_h32x_in_set_mode:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_h323_in_set_mode:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_sccp_in_set_mode:
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Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_get_int_type_frm_set_mode_ev:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/is_mode_sip_sccp_in_set_mode:
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/sip_iwf_def_set_mode_hdlr: Setting SPI
mode to SIP-TDM
Mar 17 22:08:22.051: //16/DC08AF52801D/SIP/Info/ccsip_iwf_handle_peer_event: Return value
: SIP_IWF_SUCCESS
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Mar 17 22:08:22.051: //-1/xxxxxxxxxxxx/SIP/Info/ccsip_event_handler:
CC_R_SUCCESS_WITH_CONFIRMED
Mar 17 22:08:22.051: SkinnyHWConfAPI: reqType 23:Codec List
Mar 17 22:08:22.051: SkinnyHWConfAPI: reqType 23:Codec List
Mar 17 22:08:22.051: Check associated hwconf with callid 28
Mar 17 22:08:22.051: skinny_hwcfbi_find_streamPtrAV: invalid mtp tag in stream id
0xFFFFFFFF
Mar 17 22:08:22.055: //-1/xxxxxxxxxxxx/SIP/Event/sipSPIEventInfo: Queued event from SIP
SPI : SIPSPI_EV_CC_CALL_FACILITY
Mar 17 22:08:22.055: afw_send_conf_response pLeg callID 21, subscriptionID 24, heldCall 0,
rawmsgPtr 280C314, ssInfo 189948EC
Mar 17 22:08:22.055: //-1/xxxxxxxxxxxx/SIP/Event/sipSPIEventInfo: Queued event from SIP
SPI : SIPSPI_EV_CC_CALL_FACILITY
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 50:Check to Open MM
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 24:Codec Switch
Mar 17 22:08:22.055: skinny_hwconf_get_dn_supported_codec: CONF-0: supported codec 980B
Mar 17 22:08:22.055: skinny_hwconf_is_codec_supported: codec 5 is   supported
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 25:State Update
Mar 17 22:08:22.055: SkinnyUpdateHWConfState: dn 11 chan 1 phone 1 pdn 2 pchan 1 state 12:
Mar 17 22:08:22.055: skinny_hwconf_fill_callinfo: initial adhoc party
Mar 17 22:08:22.055: skinny_update_far_end_conf_info:Update farend info cdn 11 chan 1 Cid
17 pdn 2, pchan 1
Mar 17 22:08:22.055: skinny_update_far_end_conf_info: cdn=11 cchan=1 far end sdn=2 schan=1
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 33:Meetme Opened
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 33:Meetme Opened
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 17:Is XFR to Conf
Mar 17 22:08:22.055: SkinnyHideAdhocConfNumber for dn=2 chan=1 streamID=65537
Mar 17 22:08:22.055: SkinnyHWConfAPI: reqType 28:Is Adhoc Barge
Mar 17 22:08:22.055: skinny_hwconf_fill_callinfo: cdn = 11, cchan = 1 stream 10001,
confID=C0010001
 sCallID=17, cCallID=27 Ccodec=4, Cbytes=160, Scontext = 0, Ccontext = 0, Scodec=4,
Sbytes=160, vad=0, dtmf_method = 8, C:Spt 0:0
Mar 17 22:08:22.055: skinny_hwconf_open_receive_channel: stream=10001, reuse 0
confID=C0010001conf_cb_ix = 0, PVer = 18
Mar 17 22:08:22.055: skinny_hwconf_orc_V17: stream=10001, confID=C0010001 conf_cb_ix = 0,
socket = 10
Mar 17 22:08:22.055: skinny_hwconf_compression_type: CONF-0: switch codec from 4 to 4
Mar 17 22:08:22.055: SkinnyUpdateHWConfState: conf connected phone=1 cdn=11 cchan=1 pdn=2
pchan=1, state=SKINNY_CALL_START
Mar 17 22:08:22.059: SkinnyHWConfAPI: reqType 24:Codec Switch
Mar 17 22:08:22.059: SkinnyHWConfAPI: reqType 46:MTP for CAP list
Mar 17 22:08:22.059: skinny_hwconf_preselect_mtp_tag_for_video:DN not video  cdn 11 cchan
3
Mar 17 22:08:22.059: SkinnyHWConfAPI: reqType 14:Is Conf
Mar 17 22:08:22.059: SkinnyHWConfAPI: reqType 25:State Update
Mar 17 22:08:22.059: SkinnyUpdateHWConfState: dn 11 chan 3 phone -1 pdn -1 pchan 1 state
7:
Mar 17 22:08:22.059: skinny_hwconf_admit_conf_call pdn -1 pchan 1 adhoc 1, confID
0xC0010001, number A000, cdn 11 cchan 3
Mar 17 22:08:22.059: skinny_hwconf_conf_sk_button_status:Invalid dn index for dn -1 chan 1
Mar 17 22:08:22.059: skinny_hwconf_check_adhoc_register A000 No match
Mar 17 22:08:22.059: skinny_hwconf_conf_sk_button_status:Invalid dn index for dn -1 chan 1
Mar 17 22:08:22.059: skinny_hwconf_admit_conf_call:SK = 1, ConfId=C0010001, Master = 0
Mar 17 22:08:22.059: skinny_hwconf_conf_sk_button_status:Invalid dn index for dn -1 chan 1
Mar 17 22:08:22.059: SkinnyHwconfConfSKButtonSet:Invalid dn index for dn -1 chan 1
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Mar 17 22:08:22.059: skinny_hwconf_call_open: Master=1 party -1 chan 1 lpcor_index 0
Mar 17 22:08:22.059: skinny_hwcfb_get_new_streamId: confId C0010001, codec 0, master TRUE
Mar 17 22:08:22.059: skinny_hwconf_set_preferred_codec: Conf 11:3 Phone -1:1 - -1
Mar 17 22:08:22.059: skinny_hwconf_get_dn_supported_codec: CONF-0: supported codec 980B
Mar 17 22:08:22.059: skinny_hwconf_is_codec_supported: codec 5 is   supported
Mar 17 22:08:22.059: skinny_hwconf_preselect_voip_codec selected g711u
Mar 17 22:08:22.059: skinny_hwconf_set_preferred_codec: Conf 11:3 Phone -1:1 codec = 5
Mar 17 22:08:22.063: SkinnyUpdateHWConfState for DN 11 chan 3 set ring timer
Mar 17 22:08:22.063: SkinnySetHWConfGcid: dn 11 chan 3 phone -1 pdn -1 pchan 1
skinny_hwcfb_set_cmm_confgcid: warning: conf_gcid is not zero. conf_gcid[0]=220,
[1]=9,[2]=75,[3]=122
SkinnySetHWConfGcid: Not first party in the Conf[C0010001].
SkinnySetHWConfGcid: party set conference gcid succeed
```

Additional References

The following sections provide references related to Cisco Unified CME features.

Related Documents

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

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password. If you have a valid service contract but do not have a user ID or password, you can register on Cisco.com.	http://www.cisco.com/techsupport

Feature Information for Voice and Video Hardware Conferencing

Table 34 lists the features in this module and enhancements to the features by version.

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

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

Note Table 34 lists the Cisco Unified CME version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature.

Table 34 Feature Information for Voice and Video Hardware Conferencing

Feature Name	Cisco Unified CME Version	Feature Information
Voice and Video Hardware Conferencing	9.0	Enables a Cisco Unified SIP IP phone to act as the creator of ad-hoc and meet-me conferences with audio and video conferencing streams flowing from participating IP phones through the Cisco Unified CME.