MeetingPlace 7.0: Audio/Video Interoperability White Paper

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Introduction

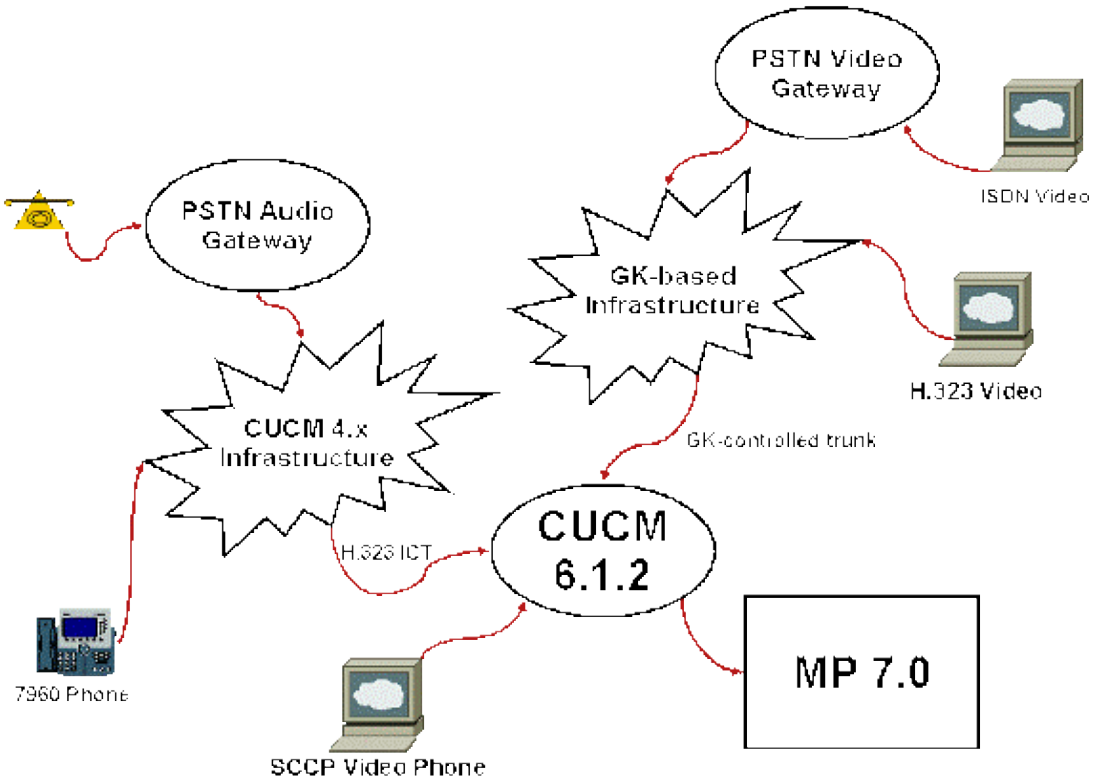
Cisco Unified MeetingPlace Release 7.0 is a Session Initiation Protocol (SIP)−based architecture that provides voice, video, and web conferencing solution to large enterprises. Cisco Unified MeetingPlace 7.0 supports two deployments for combined video and audio support. The deployment you use depends on your endpoints.

The only signaling protocol for Cisco Unified MeetingPlace Release 7.0 is SIP. These are the reasons to chose SIP:

- SIP is needed for Reservationless single−number access (RSNA) and future scalability features. RSNA relies on the SIP REFER method for inbound calls, which is supported by the Cisco Unified MeetingPlace Application Server, Cisco Unified Communications Manager, and other SIP−capable call processing agents.
- SIP is simpler in design, and is highly reliable.

MeetingPlace 7.0 Requires Cisco Unified Communications Manager Front End

All Cisco Unified MeetingPlace 7.0 installations require a Cisco Unified Communications Manager (CUCM) front end. Cisco Unified MeetingPlace 7.0 requires Cisco Unified Communications Manager Release 6.1 or later. However, you can also use Cisco Unified Communications Manager Release 6.0(x), in audio−only mode, in order to remain compatible with existing installations.



These two deployments are supported by Cisco Unified MeetingPlace 7.0:

- Audio only
- Audio and Video

Audio only: Cisco Unified Communications Manager 6.0 or later

Note: Cisco Unified Communications Manager Release 6.1(2) is the preferred deployment choice in all cases for both audio−only and audio−video. It is required if you want directory synchronization. Using Cisco Unified Communications Manager Release 6.1(2) does not limit your endpoint choices in any way. However, you can also use Cisco Unified Communications Manager Release 6.0(x), in audio−only mode in order to remain compatible with existing installations, or if you do not want to upgrade to Cisco Unified Communications Manager Release 6.1(2).

For deployments that already use Cisco Unified Communications Manager Release 6.0(x), Cisco recommends putting a Cisco Unified Communications Manager Release 6.1(2) cluster in between the older Cisco Unified Communications Manager and Cisco Unified MeetingPlace Release 7.0.

Note: Except as otherwise noted, anything that Cisco Unified Communications Manager supports, Cisco Unified MeetingPlace supports by connection through Cisco Unified Communications Manager.

These are the supported deployments with the audio−only mode:

- **Phone −> Cisco Unified Communications Manager 4.x −> H.323 ICT −> Cisco Unified Communications Manager 6.x −> Cisco Unified MeetingPlace 7.0**: You can connect Cisco Unified Communications Manager 4.x to the Cisco Unified MeetingPlace 7.0 through Cisco Unified Communications Manager 6.x using H.323 ICT and not SIP.
- **Phone −> Cisco Unified Communications Manager 5.1 and later −> SIP ICT −> Cisco Unified Communications Manager 6.x −> Cisco Unified MeetingPlace 7.0**: SIP trunks are preferred for Cisco Unified Communications Manager 5.1 and later (but H.323 is ok)

Audio and Video: CUCM 6.1.2 or later

These are the supported deployments with the audio−video mode:

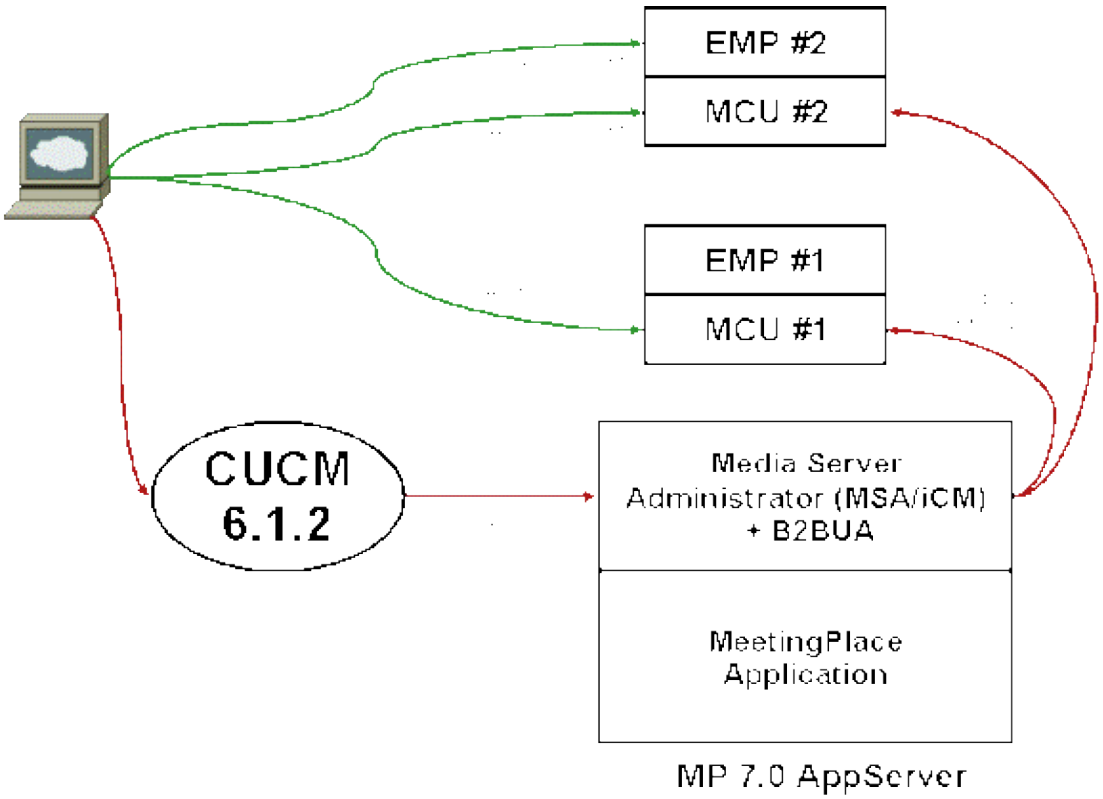
- **H.323 terminal −> Cisco IOS Gatekeeper cloud −> Cisco Unified Communications Manager 6.x −> Cisco Unified MeetingPlace 7.0**: For H.323 video, connect the Cisco IOS Gatekeeper directly to front−end Cisco Unified Communications Manager node.

  Cisco Unified MeetingPlace Release 7.0 does not provide native support for H.323 endpoints. If you want to use H.323 endpoints, you must use Registration, Admission, and Status (RAS) and a Cisco IOS Gatekeeper. The Cisco IOS Gatekeeper sends all video calls through via Cisco Unified Communications Manager Release 6.1.2 to convert the H.323 signaling to SIP. This deployment accommodates existing deployed video solutions. All H.323 video endpoints must register to a Cisco IOS Gatekeeper. Some video features currently supported by H.323 might not work with SIP. This deployment uses an Cisco IOS Gatekeeper−controlled call−control domain which integrates with Cisco Unified Communications Manager Release 6.1.2.
- **SIP or SCCP voice or video −> Cisco Unified Communications Manager Release 6.x −> Cisco Unified MeetingPlace 7.0**

  You can integrate SCCP or SIP video endpoints by defining a SIP trunk on Cisco Unified Communications Manager and Cisco Unified MeetingPlace.

Call Flows

The Cisco Unified MeetingPlace Application Server acts as the master component and it ties the rest of the components together. The Cisco Unified MeetingPlace Application Server provides SIP B2BUA support and it has SIP connections to other call processing devices such as Cisco Unified Communications Manager. The Cisco Unified MeetingPlace Application Server can integrate with external applications such as LDAP directory services via Cisco Unified Communications Manager, Simple Mail Transfer Protocol (SMTP) email, and Cisco WebEx service.



The next sections explain various call flows in the Cisco Unified MeetingPlace 7.0 deployment.

Call Flow: Initial Dial−In Connection

This is the call flow for the initial dial−in connection from the Cisco Unified MeetingPlace and the various components in the deployment.

1. CUCM −> MP/B2BUA: **INVITE**
2. B2BUA −> CUCM: **100 Trying**
3. Application decides whether to accept the call:
    - B2BUA −> MSA/iCM −> Call Processing−Media Control Protocol (CPMCP) −> VUI: **Incoming call notification**
    - VUI −> CPMCP −> MSA: **Answer call request**
    - MSA selects MCU and tells B2BUA and MCU to answer the call.
4. B2BUA −> MCU: **INVITE**
5. MCU −> B2BUA −> CUCM: **200 OK** (audio−only media offer)
6. CUCM −> B2BUA −> MCU: **ACK** (media accept)
7. Call answer response goes to the application.

Call Flow: Joining a Meeting

This is the call flow for joining a meeting from the Cisco Unified MeetingPlace and the various components in the deployment.

1. Application tells MSA (iCM) to add participant to meeting. It also indicates if video should be offered.
2. Step 1: Drop connection to IVR session (or previous meeting):
    - MCU −> B2BUA −> CUCM: **INVITE (IP=0.0.0.0; a=sendonly)**
    - CUCM −> B2BUA −> MCU: **200 OK (a=inactive or a=recvonly)**
    - MCU −> B2BUA −> CUCM: **ACK**
    - B2BUA −> MCU: **BYE**
    - MCU −> B2BUA: **200 OK (BYE)**
3. For the moment, call is on hold and no MCU is connected and DTMF sent during this short interval will be dropped.
4. Step 2: Establish the connection to meeting.
    - B2BUA −> CUCM: **INVITE** (delayed offer)
    - CUCM −> B2BUA: **200 OK** (audio/video offer, depending on terminal)
    - B2BUA −> MCU: **INVITE** (audio/video offer from CUCM)
    - MCU −> B2BUA: **200 OK** (media answer, audio−video or audio−only)
    - B2BUA −> CUCM: **ACK** (media answer from MCU)
    - B2BUA −> MCU: **ACK**
5. Often the initial offer from CUCM is `a=sendonly` , in which case CUCM will follow up with one or more re−INVITE to complete the negotiation. This can set new addresses and payload types.
6. Negotiation is usually complete when `a=sendrecv` is established.
7. As soon as you have a full−duplex audio channel, this occurs:
    - MCU −> MSA −> Application: **Add complete**
    - At this point, you can start playing the prompts.
8. When the negotiation is complete, this occurs:

MCU −> MSA −> Application: **Media active notification**: This indicates whether video was negotiated and is used for video port accounting.

Note: The same signaling is used for moving between breakout sessions, starting a #31 out−dial conference, and #9 return to IVR.

Note: No signaling (except DTMF) is involved in entering or leaving an in−conference menu.

Call Flow: Direct to Meeting Out−dial

The direct to meeting outdial requires a special setup for non−conforming terminals. This is the procedure for the setup:

1. In admin web, add a Video Terminal Profile with these settings:
   - Endpoint E.164 Number = ***phone number of terminal***
   - Skip meeting entry voice prompts = **Yes**
   - Method of attending = **Outdial to terminal**
2. When scheduling your meeting, invite the terminal.

   Note: This only works with scheduled meetings.
3. System will out−dial to terminal at the meeting start. Video will be offered immediately, if it is available.
4. Terminal will go directly into the meeting. This does not require video escalation or media transfer. There is no password check required, even if the meeting requires one.

DTMF Support

Cisco Unified MeetingPlace 7.0 supports these DTMF signalling methods:

- **RFC−2833** (special in−band media packets)

  RFC−2833 is preferred by Cisco Unified MeetingPlace and Cisco Unified Communications Manager, but it is not available from Cisco Unified IP Phones registered to Cisco Unified Communications Manager 4.x, and when using H.323, including H.323 ICT. Also, some terminals, for example Cisco 7985, do not support RFC−2833.
- **KPML** (out of band (OOB): translates easily to H.245 and SCCP)

  Cisco Unified Communications Manager can translate OOB to and from RFC−2833 by inserting a MTP. MTP insertion implies that you cannot have video. Cisco Unified Communications Manager 4.x always does this on SIP trunks, and when using SIP trunks with Cisco Unified Communications Manager 4.x, all SIP trunks are required to allocate an MTP for DTMF and Early Offer support. In order to avoid MTP, you can specify `no preference` for DTMF on Cisco Unified Communications Manager SIP trunks.
- In−band tones (traditional PSTN encoding; only works well with G.711)

Basic Compatibility Requirements for the Terminal

These are the requirements for a terminal to be fully compatible with Cisco Unified MeetingPlace 7.0:

- Terminal must be compatible with Cisco Unified Communications Manager. In order to test this, put a Cisco Unified IP Phone 7985 (or Cisco Unified IP Phone 7960+ Cisco Unified Video Advantage) on the front−end Cisco Unified Communications Manager. If you cannot call the Cisco Unified IP

Phone 7985 from the terminal, then you cannot call the Cisco Unified MeetingPlace.
- Terminal must support media transfer. Check if you can transfer the call from one Cisco Unified IP Phone 7985 to another. Refer to the H.323V4 empty capability set (ECS) call transfer specification support.
- Terminal must support video escalation. Check if you can call the terminal from a Cisco Unified IP Phone 7960 (audio only), transfer that call to a Cisco Unified IP Phone 7985, and receive the video.

Common H.323 Terminal Failings

- Terminal mishandles ECS media transfer Sometimes, the terminal fails to wait for master/slave determination as required. As a result, the call gets dropped. This issue is common with older terminals.
- Terminal reports audio−only Terminal Capabilities Set (TCS) after IVR connection Because the Cisco Unified Communications Manager needs to know that the terminal can support video, this results in No Video.
- Terminal limits itself to bearer capabilities from initial connection This prevents audio to video escalation after Cisco Unified MeetingPlace out−dial which results in No Video.

  Note: This refers to the Q.931 bearer capabilities sent to the terminal as part of the initial H.225 setup message (only for calls incoming to the terminal). These capabilities are a hint of what bandwidth is required for the call. When a call starts as audio only, the terminal typically receives bearer capabilities for a 64Kb channel, which is not enough for video. When it comes time to join a meeting, the terminal should not limit itself to the 64Kb channel when deciding whether to offer video capabilities.
- Terminals that have a manual switch which determines whether the terminal is in audio−only mode or audio−video mode. Any terminal with a manual switch of this sort does not work well with Cisco Unified MeetingPlace because:

  - ♦ It will not connect audio−only when in video mode, which results in call setup failure.
  - ♦ It will not escalate to video when in audio mode, which results in no video.

Considerations

- All Cisco Unified IP Phones should work well. However, you need to check the firmware version of the Cisco Unified IP Phone 7985.
- Cisco Unified Personal Communicator, Cisco IP Communicator and Cisco Unified Video Advantage should work well. However, make sure you have the latest software version.
- Telepresence terminals connect as audio only.
- The third party terminals might or might not work as expected, and an upgrade of the firmware can help in some cases. Refer to System Requirements and Compatibility Matrix for Cisco Unified MeetingPlace Release 7.0.
- Most terminals will work with the direct to meeting out−dial, except when not compatible with Cisco Unified Communications Manager at all.

Related Information

- **Planning Guide for Cisco Unified MeetingPlace Release 7.0**
- **Cisco Unified MeetingPlace Support**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**

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