Main page: Cisco Unified MeetingPlace Express, Release 2.x

Cisco Unified MeetingPlace Express supports integration with various devices in a SIP call-control environment. To deploy Cisco Unified MeetingPlace Express in a SIP environment, your network must have one of the following applications to route calls:

- Cisco Unified Communications Manager or Cisco Unified CallManager
- Cisco Unified Communications Manager Express or Cisco Unified CallManager Express
- Cisco IOS software voice-enabled router
- Cisco SIP Proxy Server

Prerequisites for Integrating in a SIP Environment

- Verify that the versions of your call-control device and Cisco Unified MeetingPlace Express are compatible. See the Release Notes for Cisco Unified MeetingPlace Express Release 2.0 or the Release Notes for Cisco Unified MeetingPlace Express Release 2.1.
- Verify that the Cisco Unified IP Phones are connected and added to the database of your call-control device.
- Verify that you can place and receive internal and external calls on the Cisco Unified IP Phones.

SIP Video Endpoint Requirements

- SIP video endpoints that connect to Cisco Unified MeetingPlace Express Release 2.x must fully support mid-call video escalation and de-escalation. Mid-call video escalation occurs when an endpoint joins a video meeting. Cisco Unified MeetingPlace Express will send a new Reinvite with the video attributes of the meeting. The endpoint must support receipt of this message, and must be capable of negotiating two-way video streams after the initial audio media has been established.
- When an endpoint leaves a meeting, the systems sends a new Reinvite to terminate the video session. The endpoint must support de-escalation of the video stream, returning to an audio-only session.
- SIP video endpoints must support either KPML (RFC 4730) or RFC 2833 mode for DTMF relay. (Endpoints may not send voice-band tones).

Restrictions for Integrating in a SIP Environment

The number of simultaneous calls through the SIP trunk is limited by the number of available Media Termination Point (MTP) resources. This limitation exists because SIP uses in-band RFC2833 for DTMF tones while H.323, MGCP, TAPI/JTAPI, and SCCP all use out-of-band DTMF in Cisco Unified CallManager and Cisco Unified Communications Manager. The following restrictions and conditions apply:

- Do not configure more than 48 MTP resources on the Cisco Unified Communications Manager server. Since each call requires two streams to the MTP device, with one Cisco Unified Communications Manager server and no external hardware MTP resources, the SIP trunk can support only up to 24 calls.

- If you have a cluster of Cisco Unified Communications Manager servers, each server in the cluster can provide up to 48 MTP resources to support calls through the SIP trunk. For example, a cluster with two Cisco Unified Communications Manager servers can support up to 48 calls over the SIP trunk.
- For external hardware MTP resources, you can use a Cisco Catalyst 6500 Series Communication Media Module (CMM) with at least one Ad-Hoc Conferencing and Transcoding (ACT) Port Adapter. This combination provides 512 MTP resources.
- You can avoid the MTP resource issue altogether by using an H.323 connection between Cisco Unified MeetingPlace Express and Cisco Unified Communications Manager.