# Configuring Cisco Unified Border Element Videoconferencing

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This chapter describes how to configure the Videoconferencing for the Cisco Unified Border Element (Cisco UBE) feature. The feature provides enhanced quality of service (QoS) through RSVP synchronization with H.323 signaling protocol and differentiated services code point (DSCP) packet marking. A Cisco Unified Border Element, in this guide also called an IP-to-IP gateway (IPIPGW), border element (BE), or session border controller, facilitates connectivity between independent VoIP networks by enabling H.323 VoIP and videoconferencing calls from one IP network to another.

**Activation**

**Cisco Product Authorization Key (PAK)**—A Product Authorization Key (PAK) is required to configure some of the features described in this guide. Before you start the configuration process, please register your products and activate your PAK at the following URL http://www.cisco.com/go/license.

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the "Cisco Unified Border Element Features Roadmap" section on page 1.

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

For more information about Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including feature documents, and troubleshooting information—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios124/124tcg/vcl.htm.

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# Contents



# Prerequisites for Configuring Cisco Unified Border Element Videoconferencing

- Perform the prerequisites listed in the "Prerequisites for Cisco Unified Border Element Configuration" section on page 18 in this guide.
- Perform basic H.323 gateway configuration.
- Perform basic H.323 gatekeeper configuration.

**Note** For configuration instructions, see the "Configuring H.323 Gateways" and "Configuring H.323 Gatekeepers" chapters of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

# Restrictions for Configuring Cisco Unified Border Element Videoconferencing

- H.323-to-SIP video traffic is not supported.
- H.239 for dual video (also known as Picture-in-Picture) is supported in Cisco IOS Release 12.4(20)T and later releases.
- Video is supported with slow-start.
- Dual video is not supported.
- Video with faststart and RSVP is not a supported.
- Video and T.120 data are supported only with H.323 slow-start calls.
- T.120 data is supported only in flow-around mode.
- Video endpoints must have the same H.245 version.
- Cisco Unified Border Elements that are configured for videoconferencing cannot coexist with a Multimedia Conference Manager (MCM) proxy in the same zone. See the "Migrating MCM Proxies" section on page 305 for details.
- Cisco Unified Border Elements are able to process audio and video calls without additional configuration.

- If video calls from Cisco Unified Communications Manager directly to an Cisco UBE fail, go to the Cisco Unified Communications Manager gateway configuration and uncheck the Wait for Far End H.245 Terminal Capability Set check box.
- A Cisco Unified Border Element that is configured for videoconferencing is compatible with MCM proxies. However, the following limitations apply:
  - The videoconferencing gateway cannot coexist with an MCM proxy in the same zone.
  - RSVP status depends on the type of originating and terminating gateway, as shown in the following table.

# Information About Configuring Cisco Unified Border Element Videoconferencing

The Videoconferencing for Cisco Unified Border Element feature improves the quality, reliability, and scalability of IP videoconferencing applications. In addition to the benefits offered by the Cisco UBE feature, the videoconferencing feature provides the following functionality:

- Multiple logical channels per call leg
- Exchange of video and T.120 data between H.323 call legs
- Exchange of H.245 miscellaneous commands and indications and generic capabilities between H.323 call legs
- Far End Camera Control (FECC) support
- Differentiated services code point (DSCP) marking for video streams
- RSVP synchronization of H.323 calls
- New vendor-specific attribute (VSA) for improved accounting of bandwidth usage

Feature benefits include the following:

- FECC enables an endpoint to control the remote camera on a video call connected through the Cisco UBE.
- Cisco gateways can be configured to use the max-bit-rate VSA to report bandwidth usage to accounting servers.

Cisco Unified Border Elements are able to process audio and video calls without additional configuration. However, you will most likely want to set quality-of-service (QoS) levels and control how available bandwidth is divided among the calls passing through the gateway.

This section contains the following information:

- MCM Proxies, page 302
- QoS Levels, page 302
- Bandwidth Usage, page 303

# MCM Proxies

Cisco Multimedia Conference Manager (MCM) is a Cisco IOS software feature set that enables IP networks to support secure, reliable H.323 videoconferencing, with advanced quality of service (QoS) capabilities. MCM functions as a high-performance H.323 gatekeeper and proxy, allowing network managers to control bandwidth and priority setting for H.323 videoconferencing services based on individual network configurations and capacities. These capabilities ensure appropriate allocation of network resources for videoconferencing and other critical applications running simultaneously on the network.

A Cisco Unified Border Element (Cisco UBE) that is configured for videoconferencing is compatible with MCM proxies. However, the following limitations apply:

- The videoconferencing gateway cannot coexist with an MCM proxy in the same zone.
- RSVP status depends on the type of originating and terminating gateway, as shown in the following table.

| Gateway Type | | RSVP Status |
|---|---|---|
| **Originating Gateway** | **Terminating Gateway** | |
| MCM proxy | Cisco UBE | Synchronized |
| Cisco UBE | MCM proxy | Not synchronized |

# QoS Levels

You can configure required and acceptable QoS levels on the gateway by means of the **req-qos** and **acc-qos** commands. The following levels are available:

- Best-effort—Bandwidth reservation is not attempted.
- Controlled-load—Synchronized RSVP is attempted. If it fails, the call is released.
- Guaranteed-delay—Synchronized RSVP is attempted. If it fails, one of the following occurs:
  - If acceptable QoS is best effort, call setup proceeds but without bandwidth reservation.
  - If acceptable QoS on either gateway is anything other than best effort, the call is released.

Table 1 summarizes the results of nine call-setup scenarios based on the QoS levels configured in the dial peers at the originating and terminating gateways. It does not include cases in which the requested QoS is best-effort and the acceptable QoS is something other than best-effort.

***Table 1 Call Results Based on Configured QoS Levels***

| Call Scenario | Originating Gateway | | Terminating Gateway | | |
|---|---|---|---|---|---|
| | **Requested QoS** | **Acceptable QoS** | **Requested QoS** | **Acceptable QoS** | **Results** |
| 1 | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | Call proceeds only if both RSVP reservations succeed. |
| 2 | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | best-effort | Call proceeds only if both RSVP reservations succeed. |

*Table 1 Call Results Based on Configured QoS Levels (continued)*

| Call Scenario | Originating Gateway | | Terminating Gateway | | |
|---|---|---|---|---|---|
| | **Requested QoS** | **Acceptable QoS** | **Requested QoS** | **Acceptable QoS** | **Results** |
| 3 | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | best-effort | best-effort | Call is released. |
| 4 | controlled-load or guaranteed-delay | best-effort | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | Call proceeds only if both RSVP reservations succeed. |
| 5 | controlled-load or guaranteed-delay | best-effort | controlled-load or guaranteed-delay | best-effort | Call proceeds regardless of RSVP results. If RSVP reservation fails, call receives best-effort service. |
| 6 | controlled-load or guaranteed-delay | best-effort | best-effort | best-effort | Call proceeds with best-effort service. |
| 7 | best-effort | best-effort | controlled-load or guaranteed-delay | controlled-load or guaranteed-delay | Call is released. |
| 8 | best-effort | best-effort | controlled-load or guaranteed-delay | best-effort | Call proceeds with best-effort service. |
| 9 | best-effort | best-effort | best-effort | best-effort | Call proceeds with best-effort service. |

# Bandwidth Usage

Cisco Unified Border Elements (Cisco UBE) make bandwidth decisions based on specified or default QoS levels. The **req-qos** command enables you to specify how much bandwidth is used by individual calls passing through the Cisco UBE. You can specify default and maximum amounts of bandwidth to be requested for each call. Bandwidth usage varies depending on the type of gateway, as explained below.

### Originating Cisco Unified Border Element

If you set the required QoS level the default for audio (by means of the **req-qos guaranteed-delay audio bandwidth default** command and keywords), an audio reservation is made for the default value of 64 kbps.

Normally, a video RSVP reservation is made using the value in the SETUP message bearer capability information element (IE). If this value is zero (such as with Microsoft NetMeeting), the value specified with the **video bandwidth default** keyword is used.

When you configure audio streams for either controlled-load or guaranteed-delay and configure maximum values for both audio and video, the setup is rejected if the value from the bearer-capability IE exceeds the sum of the **audio bandwidth max** and **video bandwidth max**. The max values are also checked at the time the audio and video media channels are opened. The Cisco UBE never reserves more bandwidth than the values specified with the **max** keyword.

**Note** If you do not set a maximum for either audio or video, the bearer-capability IE is not checked against max values during SETUP.

#### Terminating Cisco Unified Border Element

The value in the bearer-capability IE is not used. Instead, the audio and video bandwidth values from the SETUP message nonstandard field are used. These values are compared with the maximum values for audio and video max configured on the terminating Cisco UBE. The smaller of the two values is used for RSVP.

Table 2 summarizes the call-setup scenarios based on the configured RSVP behavior in the dial peers at the originating and terminating gateways.

*Table 2 Call Results Based on RSVP Behavior*

| Sync Mode | RSVP Mode | RSVP Result | Behavior |
|---|---|---|---|
| Sync | Requested, not best effort | Audio and video RSVP failed. | Do nothing. |
| | Acceptable, not best effort | | |
| | Requested, not best effort | Audio RSVP failed. | Kill the call. |
| | Acceptable, best effort | Video RSVP failed. | Kill the call. |
| Nonsync | Requested, not best effort | Audio and video RSVP failed. | Do nothing. |
| | Acceptable, best effort | | |
| | Requested, not best effort | Audio RSVP failed. | Kill the call. |
| | Acceptable, not best effort | Video RSVP failed. | Close the video channel. |

# How to Configure Cisco Unified Border Element Videoconferencing

This section contains the following information:

# Migrating MCM Proxies

#### Converting MCM Zones

A network that uses MCM usually consists of multiple zones, each of which includes at least one gatekeeper and one MCM proxy.

Migrate a network from MCM proxies to videoconferencing gateways on a zone-by-zone basis. When a zone is converted, replace all of the MCM proxies in that zone with Cisco Unified Border Element videoconferencing gateways.

#### Converting Individual Devices

Frequently the gatekeeper and the MCM proxy are collocated on the same router. The videoconferencing gateway cannot reside on the same device with the gatekeeper, so you need an additional router to perform videoconferencing gateway functions.

You can reuse the router that hosted the collocated gatekeeper and MCM proxy for the via-zone gatekeeper. Upgrade to a Cisco IOS release that supports via-zones. Reuse the original gatekeeper-configuration data during configuration of the new via-zone gatekeeper as appropriate. Remove the portions related to the MCM proxy and replace them with the equivalent via-zone configuration.

**Note** If a local zone is configured for via-zone, the Cisco UBE is used for all calls.

# Configuring Via-Zone Gatekeepers for Video Calls

To configure video calls to use via-zone gatekeepers, perform the steps in this section.

**Note** Video calls can take advantage of the benefits offered by via-zone gatekeeper processing. For more information, see the "Configuring Via-Zones" section of the Gatekeeper guide.

## Restrictions

Although gatekeepers can support multiple local zones, call routing between a local zone and a via zone on the same gatekeeper is not supported in Cisco IOS Release 12.2(4)T and earlier releases. Via-zone gatekeepers must be dedicated to their own via-zones.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **gatekeeper**
4. **zone local**
5. **exit**

### DETAILED STEPS

| | Command or Action | Purpose |
|---|---|---|
| Step 1 | **enable**<br><br>**Example:**<br>`Router> enable` | Enables privileged EXEC mode. Enter your password if prompted. |
| Step 2 | **configure terminal**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| Step 3 | **gatekeeper**<br><br>**Example:**<br>`Router(config)# gatekeeper` | Enters gatekeeper configuration mode. |
| Step 4 | **zone local** *gatekeeper-name domain-name* [*ras-IP-address*] [**invia** *inbound-gatekeeper* \| **outvia** *outbound gatekeeper* [**enable-intrazone**]]<br><br>**Example:**<br>`Router(config-gk)# zone local termGK example.com 10.16.193.158 invia hurricane outvia hurricane enable-intrazone` | Defines the local gatekeeper zone. Keywords and arguments are as follows:<br>• *gatekeeper-name*—Gatekeeper name or zone name<br>• *domain-name*—Domain name served by this gatekeeper<br>• *ras-IP-address*—IP address of one of the interfaces on the gatekeeper<br>• **invia** *inbound-gatekeeper*—Name of gatekeeper for calls entering this zone<br>• **outvia** *outbound-gatekeeper*—Name of gatekeeper for calls leaving this zone<br>• **enable-intrazone**—All intrazone calls are forced to use the via-zone gatekeeper<br>**Note** You can specify **invia** and **outvia** gatekeepers to be used for intrazone video calls. You can also specify **enable-intrazone** to force all intrazone calls to use the via-zone gatekeeper. |
| Step 5 | **exit**<br><br>**Example:**<br>`Router(config-gk)# exit` | Exits the current mode. |

# Configuring Audio and Video QoS Levels and Bandwidth Usage

To configure QoS and bandwidth usage, perform the steps in this section.

**Note** The following steps include sample settings that may not be appropriate for your network.

## SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **acc-qos guaranteed-delay audio**
5. **acc-qos guaranteed-delay video**
6. **req-qos guaranteed-delay audio bandwidth**
7. **req-qos guaranteed-delay video bandwidth**
8. **ip qos dscp video**
9. **exit**

## DETAILED STEPS

| | Command | Purpose |
|---|---|---|
| **Step 1** | **`enable`**<br><br>**Example:**<br>`Router> enable` | Enters privileged EXEC mode. Enter your password when prompted. |
| **Step 2** | **`configure terminal`**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | **`dial-peer voice`** *`tag`* **`voip`**<br><br>**Example:**<br>`Router(config)# dial-peer voice tag voip` | Enters dial-peer configuration mode for the specified VoIP dial peer. |
| **Step 4** | `acc-qos guaranteed-delay audio`<br><br>**Example:**<br>`Router(config-dial-peer)# acc-qos guaranteed-delay audio` | Sets acceptable QoS for audio traffic. RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded.<br><br>**Note** You cannot use the **acc-qos** command by itself. You must also use **req-qos** to specify a desired QoS for audio traffic. See Step 6. |

| | Command | Purpose |
|---|---|---|
| **Step 5** | **acc-qos guaranteed-delay video**<br><br>**Example:**<br>`Router(config-dial-peer)# acc-qos guaranteed-delay video` | Sets acceptable QoS for video traffic. RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded.<br><br>**Note** You cannot use the **acc-qos** command by itself. You must also use **req-qo**s to specify a desired QoS for video traffic. See Step 7. |
| **Step 6** | **req-qos guaranteed-delay audio bandwidth default** [*value*] **max** [*value*]<br><br>**Example:**<br>`Router(config-dial-peer)# req-qos guaranteed-delay audio bandwidth default 15 max 45` | Sets required QoS for audio traffic. RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded. Keywords and arguments are as follows:<br>• **default** [*value*]—Default audio bandwidth for RSVP, in kbps. Range: 1 to 64. Default: 64.<br>• **max** [*value*]—Maximum audio bandwidth for RSVP, in kbps. Range: 1 to 64. Default: no maximum. |
| **Step 7** | **req-qos guaranteed-delay video bandwidth default** [*value*] **max** [*value*]<br><br>**Example:**<br>`Router(config-dial-peer)# req-qos guaranteed-delay video bandwidth default 12 max 65` | Sets required QoS for video traffic. RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded. Keywords and arguments are as follows:<br>• **default** [*value*]—Default video bandwidth for RSVP, in kbps. Range: 1 to 5000. Default: 384.<br>• **max** [*value*]—Maximum video bandwidth for RSVP, in kbps. Range: 1 to 5000. Default: no maximum. |
| **Step 8** | **ip qos dscp** [value] **video** [**rsvp-none** \| **rsvp-pass** \| **rsvp-fail**]<br><br>**Example:**<br>`Router(config-dial-peer)# ip qos dscp 65 video rsvp-none` | Sets the DSCP for QoS. In this case, allows DSCP marking of RTP packets for the video stream. Keywords and arguments are as follows:<br>• *value*—DSCP value. Range: 0 to 63.<br>• **video rsvp-none**—Applies DSCP to video stream with no RSVP reservations<br>• **video rsvp-pass**—Applies DSCP to video stream with successful RSVP reservations<br>• **video rsvp-fail**—Applies DSCP to video stream with failed RSVP reservations |
| **Step 9** | **exit**<br><br>**Example:**<br>`Router(config-dial-peer)# exit` | Exits the current mode. |

## Configuring RSVP Synchronization for H.323 Slow Start

To configure RSVP synchronization for H.323 slow start for all H.323 calls, perform the steps in this section.

**Note** This task is optional; RSVP synchronization is enabled by default.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **call start**
6. **exit**

### DETAILED STEPS

| | Command | Purpose |
|---|---|---|
| **Step 1** | **enable**<br><br>**Example:**<br>`Router> enable` | Enters privileged EXEC mode. Enter your password when prompted. |
| **Step 2** | **configure terminal**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | **voice service voip**<br><br>**Example:**<br>`Router(config)# voice service voip` | Enters VoIP voice-service configuration mode. |
| **Step 4** | **h323**<br><br>**Example:**<br>`Router(conf-voi-serv)# h323` | Enters H.323 configuration mode. |

| | Command | Purpose |
|---|---|---|
| **Step 5** | **call** [**start** {**fast** \| **slow** \| **system**}] \| [**sync-rsvp slow-start**]<br><br>**Example:**<br>`Router(config-class)# call slow sync-rsvp slow-start` | Forces an H.323 gateway to use fast-connect or slow-connect procedures for a dial peer. Use the **sync-rsvp slow-start** keyword to enable RSVP synchronization for slow-start calls. Keywords are as follows:<br>• **fast**—Fast-connect procedures<br>• **slow**—Slow-connect procedures<br>• **system**—Voice-service configuration<br>• **sync-rsvp slow-start**—RSVP synchronization for slow-start calls<br><br>Default: **system** |
| **Step 6** | **exit**<br><br>**Example:**<br>`Router(config-class)# exit` | Exits the current mode. |

# Configuring Interworking of Polycom Endpoints

To configure interworking between Polyom endpoints, perform the steps in this section.

## Restrictions

Interworking between Polycom endpoints are determined by the software version running on each endpoint.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **h225 h225 id-passthru**
6. **exit**

### DETAILED STEPS

| | Command or Action | Purpose |
|---|---|---|
| **Step 1** | `enable`<br><br>**Example:**<br>`Router> enable` | Enables privileged EXEC mode. Enter your password if prompted. |
| **Step 2** | `configure terminal`<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | `voice service voip`<br><br>**Example:**<br>`Router(config)# voice service voip` | Enters VoIP voice-service configuration mode. |
| **Step 4** | `h323`<br><br>**Example:**<br>`Router(config-voice-service)# h323` | Enters H.323 voice-service configuration mode. |
| **Step 5** | `h225 h225 id-passthru`<br><br>**Example:**<br>`Router(config-serv-h323)# h225 h225 id-passthru` | Enables signaling between video endpoints with different H.245 versions. |
| **Step 6** | `exit`<br><br>**Example:**<br>`Router(config-serv-h323)# exit` | Exits the current mode. |

# Configuring a Voice Class

To configure a voice class that is independent of a dial peer and can be used on multiple dial peers, perform the steps in this section.

## SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class**
4. **call start**
5. **exit**

## DETAILED STEPS

| | Command | Purpose |
|---|---|---|
| **Step 1** | **enable**<br><br>**Example:**<br>`Router> enable` | Enters privileged EXEC mode. Enter your password when prompted. |
| **Step 2** | **configure terminal**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | **voice class** *tag*<br><br>**Example:**<br>`Router (config)#` **voice class h323 1234** | Creates an H.323 voice class. |
| **Step 4** | **call** [**start** {**fast** \| **slow** \| **system**}] \| [**sync-rsvp slow-start**]<br><br>**Example:**<br>`Router (config-class)# call sync-rsvp slow-start` | Enables RSVP synchronization for slow-start calls. Default: synchronization is enabled. |
| **Step 5** | **exit**<br><br>**Example:**<br>`Router(config-class)# exit` | Exits the current mode. |

# Configuring Delayed-Offer to Early-Offer for SIP Video Calls

This feature the alters the default configuration of the Cisco Unified BE from not distinguishing SIP Delayed-Offer to Early-Offer call flows, to forcing the Cisco Unified BE to generate an Early-Offer with the configured codecs for a incoming Delayed-Offer INVITE. To configure a Cisco Unified Border Element to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL) perform the steps in this section.

To Delayed-Offer to Early-Offer for SIP Audio Calls for all VoIP calls, or individual dial peers, perform the steps in this section. This section contains the following subsections:


- Configuring Delayed-Offer to Early-Offer for SIP Audio Calls at the Global Level, page 314
- Configuring Delayed-Offer to Early-Offer for SIP Audio Calls for a Dial-Peer, page 315


## Prerequisites

- The **allow-connections sip to sip** command must be configured before you configure media flow-around. For more information and configuration steps see the "Configuring SIP-to-SIP Connections in a Cisco Unified Border Element" section on page 168 of the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter.

## Restrictions

- Cisco Unified Communications Manager 5.x supports Early-Offer over SIP trunk for audio calls with MTP
- Support for Cisco Unified Communications Manager Early-Offer for video calls and audio calls without MTP is not supported

Table 3 shows a list of protocol interworking for SIP.

***Table 3 Supported protocol interworking***

| Protocol | In Leg | Out Leg | Support |
|---|---|---|---|
| H.323-to-SIP | Fast Start | Early-Offer | Bi-Directional |
| | Slow Start | Delayed-Offer | Bi-Directional |
| SIP-to-SIP | Early-Offer | Early-Offer | Bi-Directional |
| | Delayed-Offer | Delayed-Offer | Bi-Directional |
| | Delayed-Offer | Early-Offer | Uni-Directional |

## Configuring Delayed-Offer to Early-Offer for SIP Audio Calls at the Global Level

To configure Delayed-Offer to Early-Offer for SIP Audio Calls at the global level, perform the steps in this section.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections sip**
5. **early-offer forced**
6. **exit**

### DETAILED STEPS

| | Command or Action | Purpose |
|---|---|---|
| **Step 1** | **`enable`**<br><br>**Example:**<br>`Router> enable` | Enables privileged EXEC mode.<br>• Enter your password if prompted. |
| **Step 2** | **`configure terminal`**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | **`voice service voip`**<br><br>**Example:**<br>`Router(config)# voice service voip` | Enters VoIP voice-service configuration mode. |
| **Step 4** | **`allow-connections`** *`from-type`* **`to`** *`to-type`*<br><br>**Example:**<br>`Router(config-voi-serv)# allow-connections sip to sip` | Allows connections between specific types of endpoints in an Cisco UBE. Arguments are as follows:<br>• *from-type*—Type of connection. Valid values: **h323**, **sip**.<br>• *to-type*—Type of connection. Valid values: **h323**, **sip**.<br>**Note** H.323-to-H.323: By default, H.323-to-H.323 connections are disabled and POTS-to-any and any-to-POTS connections are enabled. |
| **Step 5** | **`early-offer forced`**<br><br>**Example:**<br>`Router(config-voi-serv)# early-offer forced` | Enables SIP Delayed-Offer to Early-Offer globally. |
| **Step 6** | **`exit`**<br><br>**Example:**<br>`Router(config-voi-serv)# exit` | Exits the current mode. |

## Configuring Delayed-Offer to Early-Offer for SIP Audio Calls for a Dial-Peer

To configure Delayed-Offer to Early-Offer for SIP Audio Calls for an individual dial-peer, perform the steps in this section.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice 1 voip**
4. **voice-class sip early-offer forced**
5. **exit**

### DETAILED STEPS

| | Command or Action | Purpose |
|---|---|---|
| Step 1 | **`enable`**<br><br>**Example:**<br>`Router> enable` | Enables privileged EXEC mode. Enter your password if prompted. |
| Step 2 | **`configure terminal`**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| Step 3 | **`dial-peer voice`** *`number`* **`voip`**<br><br>**Example:**<br>`Router(config)# dial-peer voice 2 voip` | Enters dial-peer configuration mode for the specified VoIP dial peer. |
| Step 4 | **`voice-class sip early-offer forced`**<br><br>**Example:**<br>`Router(config-dial-peer)# voice-class sip early-offer forced` | Forcefully send Early-Offer |
| Step 5 | **`exit`**<br><br>**Example:**<br>`Router(config-dial-peer)# exit` | Exits the current mode. |

# Configuring SIP Video Calls with Flow Around Media

To configure SIP video calls to be placed on the Cisco Unified Border Element (Cisco UBE) where the media flows around the Cisco UBE from endpoint to endpoint.

## Restrictions

- SIP video calls with flow around media is supported in Cisco IOS Release 12.4(20)T and later.

- SIP video calls with flow through media is supported in Cisco IOS Release 12.4(15)XZ and earlier.
- This is normally directly from endpoint to endpoint,

## SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5.
6. **exit**

## DETAILED STEPS

| | Command or Action | Purpose |
|---|---|---|
| **Step 1** | **`enable`**<br><br>**Example:**<br>`Router> enable` | Enables privileged EXEC mode. Enter your password if prompted. |
| **Step 2** | **`configure terminal`**<br><br>**Example:**<br>`Router# configure terminal` | Enters global configuration mode. |
| **Step 3** | **`voice service voip`**<br><br>**Example:**<br>`Router(config)# voice service voip` | Enters VoIP voice-service configuration mode. |
| **Step 4** | **`h323`**<br><br>**Example:**<br>`Router(conf-voi-serv)# h323` | Enters H.323 voice-service configuration mode. |
| **Step 5** | **Example:**<br>`Router(config-serv-h323)#` | . |
| **Step 6** | **`exit`**<br><br>**Example:**<br>`Router(config-serv-h323)# exit` | Exits the current mode. |

# Verifying and Troubleshooting Cisco Unified Border Element Videoconferencing

To troubleshoot or verify Cisco Unified Border Element Videoconferencing, perform the steps in this section. This section contains the following subsections:



## Troubleshooting Tips

For examples of **show** and **debug** command output and details on interpreting the output, see the following resources:

- Cisco IOS Debug Command Reference, Release 12.3T
- *Cisco IOS Voice Troubleshooting and Monitoring Guide*
- *Troubleshooting and Debugging VoIP Call Basics*
- *Voice Gateway Error Decoder for Cisco IOS*
- *VoIP Debug Commands*

## Verifying and Monitoring Cisco Unified Border Element Videoconferencing

To verify, monitor, and maintain audio and video calls, perform the following steps (listed alphabetically).

### SUMMARY STEPS

1. **show call active video**
2. **show call history video**
3. **show dial-peer voice**
4. **show ip rsvp reservation**
5. **show running-config**

### DETAILED STEPS

**Step 1** **show call active video**

Use this command to display call statistics, including video bytes and packets received, video bytes and packets transmitted, bandwidth used, and UDP ports used, for active calls.

**Step 2** **show call history video**

Use this command to display the same call statistics for all calls.

**Step 3** **show dial-peer voice**

Use this command to display dial-peer statistics, including default and maximum bandwidth values for audio and video and DSCP marking for video.

**Step 4** **show ip rsvp reservation**

Use this command to display RSVP-related receiver information currently in the database.

**Step 5** **show running-config**

Use this command to verify audio and video QoS.

```
!
interface FastEthernet0/0
 ip address 10.1.1.5 255.255.255.0
 ip route-cache same-interface
 h323-gateway voip interface
 h323-gateway voip id zone1-gk ipaddr 10.1.1.1 1718
 h323-gateway voip tech-prefix 1#
 h323_gateway voip bind srcaddr 10.1.1.5
 ip rsvp bandwidth 7000 1000
!
!
dial-peer voice 100 voip
 voice-class h323 1
 req-qos guaranteed-delay audio bandwidth default 16 max 32
 req-qos guaranteed-delay video bandwidth default 320 max 768
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 ip qos dscp af11 media
 ip qos dscp af21 signaling
 ip qos dscp af33 video rsvp-none
 ip qos dscp af31 video rsvp-pass
 ip qos dscp af32 video rsvp-fail
 codec transparent
!
```

# Configuration Examples for Cisco Unified Border Element Videoconferencing

This section provides the following configuration examples:

- QoS for Audio and Video on One Gateway: Example, page 318
- QoS for Audio and Video on Two Gateways: Example, page 319

## QoS for Audio and Video on One Gateway: Example

The following example shows QoS for audio and video configured on a Cisco Unified Border Element. Note that this example uses values and settings that may not be appropriate for your network.

```
!
voice service voip
 no allow-connections any to pots
 no allow-connections pots to any
 allow-connections h323 to h323
 h323
  no call sync-rsvp slow-start
!
!
voice class h323 1
  no call sync-rsvp slow-start
!
!
```

```
interface FastEthernet0/0
 ip address 10.1.1.2 255.255.255.0
 ip route-cache same-interface
 h323-gateway voip interface
 h323-gateway voip id zone1-gk ipaddr 10.1.1.1 1718
 h323-gateway voip tech-prefix 1#
 h323_gateway voip bind srcaddr 10.1.1.2
 ip rsvp bandwidth 7000 1000
!
!
dial-peer voice 100 voip
 voice-class h323 1
 req-qos guaranteed-delay audio bandwidth default 16 max 32
 req-qos guaranteed-delay video bandwidth default 320 max 768
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 ip qos dscp af11 media
 ip qos dscp af21 signaling
 ip qos dscp af33 video rsvp-none
 ip qos dscp af31 video rsvp-pass
 ip qos dscp af32 video rsvp-fail
 codec transparent
```

## QoS for Audio and Video on Two Gateways: Example

The following example shows the dial-peers for two Cisco Unified Border Elements that exchange video calls. Each gateway is connected to an endpoint that does not support RSVP; however, RSVP is used between the Cisco UBEs. One endpoint has an E.164 address of 1231000, and the other endpoint has an E.164 address of 4569000. Because the endpoints do not support RSVP, the gateways must have two dial peers for each call leg, one that prevents RSVP reservations to the endpoints and one that allows RSVP between the gateways.

### Cisco Unified Border Element Connected to 1231000

```
dial-peer voice 123 voip
 description dial-peer incoming from ip-ip gateway
 incoming called-number 123....
 session target ras
 req-qos guaranteed-delay audio
 req-qos guaranteed-delay video
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 codec transparent
!
dial-peer voice 456 voip
 description dial-peer incoming from video endpoint
 incoming called-number 456....
 session target ras
 codec transparent
!
dial-peer voice 4569 voip
 description dial-peer outgoing to ip-ip gateway
 destination-pattern 456....
 session target ras
 req-qos guaranteed-delay audio
 req-qos guaranteed-delay video
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 codec transparent
!
dial-peer voice 1231 voip
```

```
 description dial-peer outgoing to video endpoint
 destination-pattern 123....
 session target ras
 codec transparent
!
```

### Cisco Unified Border Element Connected to 4569000

```
dial-peer voice 123 voip
 description dial-peer incoming from video endpoint
 incoming called-number 123....
 session target ras
 codec transparent
!
dial-peer voice 456 voip
 description dial-peer incoming from ip-ip gateway
 incoming called-number 456....
 session target ras
 req-qos guaranteed-delay audio
 req-qos guaranteed-delay video
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 codec transparent
!
dial-peer voice 1231 voip
 description dial-peer outgoing to ip-ip gateway
 destination-pattern 123....
 session target ras
 req-qos guaranteed-delay audio
 req-qos guaranteed-delay video
 acc-qos guaranteed-delay audio
 acc-qos guaranteed-delay video
 codec transparent
!
dial-peer voice 4569 voip
 description dial-peer outgoing to video endpoint
 destination-pattern 456....
 session target ras
 codec transparent
!
```

# Where to Go Next

- H.323-to-H.323 Connections on a Cisco Unified Border Element
- H.323-to-SIP Connections on a Cisco Unified Border Element
- SIP-to-SIP Connections on a Cisco Unified Border Element
- Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

# Additional References

The following sections provide additional references related to the Cisco UBE Configuration Guide.

**Note**

- In addition to the references listed below, each chapter provides additional references related to Cisco Unified Border Element.
- Some of the products and services mentioned in this guide may have reached end of life, end of sale, or both. Details are available at http://www.cisco.com/en/US/products/prod_end_of_life.html.
- The preface and glossary for the entire voice-configuration library suite of documents is listed below.

## Related Documents

| Related Topic | Document Title |
|---|---|
| Cisco IOS commands | Cisco IOS Master Commands List, All Releases |
| Cisco IOS Voice commands | Cisco IOS Voice Command Reference |
| Cisco IOS Voice Configuration Library | For more information about Cisco IOS voice features, including feature documents, and troubleshooting information—at<br>http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm |
| Cisco IOS Release 15.0 | Cisco IOS Release 15.0 Configuration Guides |
| Cisco IOS Release 12.4 | • Cisco IOS Release 12.4 Configuration Guides<br>• Cisco IOS Release 12.4T Configuration Guides |
| Cisco IOS Release 12.3 | • Cisco IOS Release 12.3 documentation<br>• Cisco IOS Voice Troubleshooting and Monitoring Guide<br>• Tcl IVR Version 2.0 Programming Guide |
| Cisco IOS Release 12.2 | *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2 |
| DSP documentation | High-Density Packet Voice Feature Card for Cisco AS5350XM and AS5400XM Universal Gateways<br>http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/vfc_dsp.html |
| GKTMP (GK API) Documents | • *GKTMP Command Reference*: http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_cli.htm<br>• *GKTMP Messages*: http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_tmp.html |

| Related Topic | Document Title |
|---|---|
| internet Low Bitrate Codec (iLBC) Documents | • Codecs section of the Dial Peer Configuration on Voice Gateway Routers Guide<br>http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_ovrvw.html<br>• Dial Peer Features and Configuration section of the Dial Peer Configuration on Voice Gateway Routers Guide<br>http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_confg.html |
| Cisco Unified Border Element Configuration Examples | • Local-to-remote network using the IPIPGW http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00801b0803.shtml<br>• Remote-to-local network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edc.shtml<br>• Remote-to-remote network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edd.shtml<br>• Remote-to-remote network using two IPIPGWs: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edb.shtml |
| Related Application Guides | • Cisco Unified Communications Manager and Cisco IOS Interoperability Guide<br>• Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide<br>• "Configuring T.38 Fax Relay" chapter<br>• Cisco IOS SIP Configuration Guide<br>• Cisco Unified Communications Manager (CallManager) Programming Guides<br>• Quality of Service for Voice over IP |
| Related Platform Documents | • Cisco 2600 Series Multiservice Platforms<br>• Cisco 2800 Series Integrated Services Routers<br>• Cisco 3600 Series Multiservice Platforms<br>• Cisco 3700 Series Multiservice Access Routers<br>• Cisco 3800 Series Integrated Services Routers<br>• Cisco 7200 Series Routers<br>• Cisco 7301 |
| Related gateway configuration documentation | Media and Signaling Authentication and Encryption Feature for Cisco IOS H.323 Gateways.<br>http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/htsecure.htm |

| Related Topic | Document Title |
| --- | --- |
| Cisco IOS NAT Configuration Guide, Release 12.4T | *Configuring Cisco IOS Hosted NAT Traversal for Session Border Controller*<br><br>http://www.cisco.com/en/US/docs/ios/12_4t/ip_addr/configuration/guide/htnatsbc.html |
| Troubleshooting and Debugging guides | • Cisco IOS Debug Command Reference, Release 12.4 at<br>http://www.cisco.com/en/US/docs/ios/debug/command/reference/db_book.html<br>• *Troubleshooting and Debugging VoIP Call Basics* at http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml<br>• *VoIP Debug Commands* at<br>http://www.cisco.com/en/US/docs/routers/access/1700/1750/software/configuration/guide/debug.html |

# Standards

| Standard | Title |
| --- | --- |
| H.323 Version 4 and earlier | *H.323 (ITU-T VOIP protocols)* |
| H.323 - H.245 Version 12, Annex R | *H.323 (ITU-T VOIP protocols)* |

# MIBs

| MIB | MIBs Link |
| --- | --- |
| • CISCO-DSP-MGMT-MIB<br>• CISCO-VOICE-DIAL-CONTROL-MIB<br>• IP-TAP-MIB<br>• TAP2-MIB<br>• USER-CONNECTION-TAP-MIB | To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:<br><br>http://www.cisco.com/go/mibs |

# RFCs

| RFC | Title |
| --- | --- |
| RFC 1889 | *RTP: A Transport Protocol for Real-Time Applications* |
| RFC 2131 | *Dynamic Host Configuration Protocol* |
| RFC 2132 | *DHCP Options and BOOTP Vendor Extensions* |
| RFC 2833 | *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* |
| RFC 3203 | *DHCP reconfigure extension* |
| RFC 3261 | *SIP: Session Initiation Protocol* |

| RFC | Title |
|---|---|
| RFC 3262 | *Reliability of Provisional Responses in Session Initiation Protocol (SIP)* |
| RFC 3323 | *A Privacy Mechanism for the Session Initiation Protocol (SIP)* |
| RFC 3325 | *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks* |
| RFC 3361 | *Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers* |
| RFC 3455 | *Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)* |
| RFC 3608 | *Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration* |
| RFC 3711 | *The Secure Real-time Transport Protocol (SRTP)* |
| RFC 3925 | Vendor-Identifying Vendor Options for Dynamic Host Configuration Protocol version 4 (DHCPv4) |

# 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.<br><br>To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds.<br><br>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password. | http://www.cisco.com/cisco/web/support/index.html |

# Feature Information for Configuring Cisco Unified Border Element Videoconferencing

Table 4 lists the features in this module and provides links to specific configuration information. Only features that were introduced or modified in Cisco IOS Release 12.3(1) or a later release appear in the table.

For information on a feature in this technology that is not documented here, see the "Cisco Unified Border Element Features Roadmap"of this guide.

**Note** Table 4 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

***Table 4 Feature Information for Configuring Cisco Unified Border Element Videoconferencing***

| Feature Name | Releases | Feature Information |
|---|---|---|
| Delayed offer to Early offer for SIP Video Calls | 12.4(20)T1 | This feature was introduced. |
| H.323 Video Calls Support for H.235 Security | 12.4(15)XY | This feature was introduced. |
| H.323 Video Calls Support for H.239 Signaling | 12.4(15)XY | This feature was introduced. |
| Videoconferencing for the Cisco Unified Border Element Feature | 12.3(4)T | This feature was introduced. |