H.323-to-H.323 Connections on a Cisco Unified Border Element

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This chapter describes how to configure and enable features for H.323-to-H.323 connections in an Cisco Unified Border Element topology.

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.

Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Contents

Prerequisites for H.323-to-H.323 Connections on a Cisco Unified Border Element

- Perform the prerequisites listed in the "Prerequisites for Cisco Unified Border Element Configuration" section on page 20.
- Perform fundamental gateway configuration listed in the Prerequisites for Fundamental Cisco Unified Border Element Configuration, page 44.
- 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 H.323-to-H.323 Connections on a Cisco Unified Border Element

- Connections are disabled by default in Cisco IOS images that support the Cisco Unified Border Element.
- Slow-start to fast-start interworking is supported only for H.323-to-H.323 calls.
- Transcoding in fast-start to slow-start interworking is not supported.
- Supplementary services with transcoding is not supported.
- DTMF Interworking rtp-nte to out of band is not supported when high density transcoder is enabled. Use normal transcoding for rtp-nte to out of band DTMF interworking.

Information About H.323-to-H.323 Connections on a Cisco Unified Border Element

H.323-to-H.323 Gateway configuration provides a network-to-network demarcation point between independent VoIP and video networks by for billing, security, call-admission control, QoS, and signaling interworking. Performs most of the functions of a PSTN-to-IP gateway but joins two H.323 VoIP call legs.

Note When you configure H.323-to-H.323 connections on a Cisco UBE, the ports on all its interfaces are open by default. This makes the Cisco UBE vulnerable to malicious attackers who can execute toll fraud across the gateway if the Cisco UBE has a public IP address and a PSTN connection. To eliminate the threat, you should bind an interface to private IP address that is not accessible by untrusted hosts. In addition, you should protect any public or untrusted interface by configuring a firewall or an access control list (ACL) to prevent unwanted traffic from traversing the router.

How to Configure H.323-to-H.323 Connections on a Cisco Unified Border Element

This section contains the following tasks:

Configuring H.323-to-H.323 Connections on a Cisco Unified Border Element

To configure H.323-to-H.323 connections on a Cisco UBE, perform the steps in this section.

Restrictions

Connections are disabled by default in Cisco IOS images that support the Cisco Unified Border Element.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type* **to** *to-type*
5. **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	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**allow-connections** *from-type* **to** *to-type* **Example:** `Router(config-voi-serv)# allow-connections h323 to h323`	Allows connections between specific types of endpoints in an Cisco Unified Border Element. Arguments are as follows: • *from-type*—Type of connection. Valid values: **h323**, **sip**. • *to-type*—Type of connection. Valid values: **h323**, **sip**. **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	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits the current mode.

Enabling H.323-to-H.323 Interworking Between Fast Start and Slow Start

Slow-start to fast-start interworking prevents the Cisco Unified Border Element from dropping a call down to slow-start when it detects different call signaling on the incoming and outgoing legs of H.323 to H.323 calls. Configuration may be done at either the dial-peer level or the global level.

To enable H.323-to-H.323 interworking perform the steps in this section. This section contains the following subsections:

- Enabling Slow-Start to Fast-Start Interworking at the Global Level, page 107
- Enabling Slow-Start to Fast-Start Interworking at the Dial Peer Level, page 108

Enabling Slow-Start to Fast-Start Interworking at the Global Level

To configure slow-start to fast-start interworking on an Cisco Unified Border Element at the global level, perform the steps in this section.

Prerequisites

Configure **call start interwork** on both the incoming and outgoing legs.

Restrictions

The **call start interwork** command only supports interwork between fast-start and slow-start. It should not be used in situations where fast-start to fast-start or slow-start to slow-start calls are possible.

SUMMARY STEPS

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

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**h323** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice-service configuration mode.
Step 5	**call start interwork** **Example:** `Router(conf-voi-serv)# call start interwork`	Enables slow-start to fast-start interworking.
Step 6	**exit** **Example:** `Router(conf-voi-serv# exit`	Exits the current mode.

Enabling Slow-Start to Fast-Start Interworking at the Dial Peer Level

To configure slow-start to fast-start interworking on an Cisco Unified Border Element at the dial-peer level, perform the steps in this section.

Prerequisites

- Configure **call start interwork** on both the incoming and outgoing legs.
- Specify the codec on both the incoming and outgoing dial-peer.

Restrictions

- The **call start interwork** command only supports interwork between fast-start and slow-start. It should not be used in situations where fast-start to fast-start or slow-start to slow-start calls are possible.
- When **call start interwork** is configured, both incoming and outgoing dial-peer need to have a specific codec configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-class h323**
4. **call start interwork**
5. **exit**
6. Repeat as needed.

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	**`voice class h323`** *tag* **Example:** `Router(config)# voice class h323 4`	Creates an H.323 voice class that is independent of a dial peer and can be used on multiple dial peers.

Step 4	`call start interwork`
	
	Example:
	`Router(config-class)# call start interwork`
		Enables slow-start to fast-start interworking.

Step 5	**`exit`** **Example:** `Router(config-class)# exit`	Exits the current mode.
Step 6	Repeat for both incoming and outgoing legs.	—

Configuring Media Flow-Around

This feature adds media flow-around capability on the Cisco UBE by supporting the processing of call setup and teardown requests (VoIP call signaling) and for media streams (flow-through and flow-around). Media flow-around can be configured the global level or it must be configured on both incoming and outgoing dial peers. If configured only on either the incoming or outgoing dial peer, the call will become a flow-through call.

Media flow-around is a good choice to improve scalability and performance when network-topology hiding and bearer-level interworking features are not required

With the default configuration, the Cisco Unified Border Element receives media packets from the inbound call leg, terminates them, and then reoriginates the media stream on an outbound call leg. Media flow-around enables media packets to be passed directly between the endpoints, without the intervention of the Cisco Unified Border Element. The Cisco Unified Border Element continues to handle routing and billing functions.

Note The Cisco Unified Border Element must be running Cisco IOS Release 12.3(1) or a later release to support media flow-around.

To specify media flow-around for voice class, all VoIP calls, or individual dial peers perform the steps in this section. This section contains the following subsections:

Configuring Media Flow-Around for a Voice Class

To configure media flow-around for a voice class, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class media 1***tag*
4. **media flow-around**
5. **dial-peer voice 2 voip**
6. **voice-class media** *tag*
7. **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	**voice class media** *tag* **Example:** `Router(config)# voice class media 1`	Enters voice-class configuration mode and assign an identification tag for a media voice class.
Step 4	**media flow-around** **Example:** `Router(config-class)# media flow-around`	Enables media flow around.
Step 5	**dial-peer voice 2 voip** **Example:** `Router(config-class)# dial-peer voice 2 voip`	Enters dial-peer configuration mode and assign an identification tag for VoIP.

	Command or Action	Purpose
Step 6	**voice class media** *tag* **Example:** `Router(config-dial-peer)# voice class media 1`	Assign an identification tag for a media voice class.
Step 7	**exit** **Example:** `Router(config-class)# exit`	Exit voice class-configuration mode.

Configuring Media Flow-Around at the Global Level

To configure media flow-around at the global level, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media flow-around**
5. **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	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**media flow-around** **Example:** `Router(config-voi-serv) media flow-around`	Enables media flow-around.
Step 5	**exit** **Example:** `Router(config-voi-serv) exit`	Exits the current mode.

Configuring Media Flow-Around for a Dial Peer

To configure media flow-around for an individual dial-peer, perform the steps in this section.

Restrictions

If you plan to configure both incoming and outgoing dial peers, you must specify the transparent codec on the incoming dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **media flow-around**
5. **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	**`dial-peer voice`** *`number`* **`voip`** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	**`media flow-around`** **Example:** `Router(config-dial-peer) media flow-around`	Enables media flow-around.
Step 5	**`exit`** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring H.323-to-H.323 Call Failure Recovery (Rotary) on a Cisco Unified Border Element

- Call failure recovery (Rotary) on the Cisco Unified Border Element eliminates the need for identical codec capabilities for all dial peers in the rotary group, and allows the Cisco Unified Border Element to restart the codec negotiation end-to-end.
- Call failure recovery will continue until "voice hunt stop" is reached.

To configure H.323-to-H.323 call failure recovery (rotary) on an Cisco Unified Border Element, perform the steps in this section.

Restrictions

If extended caps (DTMF or T.38) are configured on the outgoing gateway or the trunking gateway, extended caps must be configured in both places.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **emptycapability**
6. **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	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**h323** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice-service configuration mode.

	Command or Action	Purpose
Step 5	**emptycapability** **Example:** `Router(conf-serv-h323)# emptycapability`	Enables call failure recovery (TCS=0).
Step 6	**exit** **Example:** `Router(conf-serv-h323)# exit`	Exits the current mode.

Managing H.323 IP Group Call Capacities

Managing maximum capacity for the IP group is done with carrier IDs created on an IP trunk group. If you do not configure specific carrier IDs, you can use the **ip circuit default only** command to create a single carrier. However, if you want to use carrier ID-based routing, or if you need extra control for load and resource balancing, you must configure carrier IDs in conjunction with the **voice source-group** command.

The Cisco UBE feature works with the **voice source-group** command to provide matching criteria for incoming calls. The **voice source-group** command assigns a name to a set of source IP group characteristics. The terminating gateway uses these characteristics to identify and translate the incoming VoIP call. If there is no voice source group match, the default carrier ID is used, any source carrier ID on the incoming message is transmitted without change, and no destination carrier is available. Call-capacity information is reported to the gatekeeper, but carrier routing information is not.

If the voice source group matches, the matched source carrier ID is used and the target carrier ID defined in the voice source group is used for the destination carrier ID.

To manage H.323 IP group call capacities, perform the steps in this section.

Restrictions

You can use the commands that follow only when no calls are active. If you try to use these commands with active calls present, the commands are rejected.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **ip circuit max-calls**
6. **ip circuit carrier-id**
7. **ip circuit default only**
8. **ip circuit default name**
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 1	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 2	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 3	**h323** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice-service configuration mode.
Step 4	**ip circuit max-calls** *maximum-calls* **Example:** `Router(config-serv-h323)# ip circuit max-calls 1500`	(Required only if reserved calls are to exceed 1000) Sets the maximum number of aggregate H.323 IP circuit carrier call legs. If you do not configure this value, the default maximum value is 1000 reserved call legs. You may need to configure a lower value to obtain overload behavior. You can also configure a higher value. **Note** After you set a maximum number of call legs for defined circuits, any aggregate capacity left over is available for default circuits. For example, if you specify 1000 as the maximum number of call legs and then reserve 200 call legs for defined circuits, 800 call legs are available for use by default circuits. **Note** The Cisco Unified Border Element prevents you from allocating all of the capacity to specified carriers; at least one available circuit is required, which can be the default.
Step 5	**ip circuit carrier-id** *carrier-name* [**reserved-calls** *reserved*] **Example:** `Router(config-serv-h323)# ip circuit carrier-id AA reserved-calls 500`	(Optional) Defines an IP circuit using the specified name as the circuit ID. **Note** The **reserved** keyword for this command is optional. Using this keyword creates a specified maximum number of calls for that circuit ID. The default value is 200 call legs.

	Command or Action	Purpose
Step 6	**ip circuit default only** **Example:** `Router(config-serv-h323)# ip circuit default only`	(Optional) Creates a single carrier to use all of the call capacity available to the Cisco Unified Border Element. **Note** If you use the **ip circuit default only** command, you cannot use the **ip circuit carrier-id** command to configure more circuits. Using the **ip circuit default only** command creates a single carrier using the default carrier name.
Step 7	**exit** **Example:** `Router(conf-serv-h323)# exit`	Exits the current mode.

Examples

The following examples show a default carrier with no voice source group configured:

Default Carrier with No Voice Source Group

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit default only
```

If there is no incoming source carrier ID:

- Capacity only is reported to the gatekeeper using the default circuit (two call legs).
- No source or destination carrier information is reported.

If there is an incoming source carrier ID:

- Two call legs are counted against the default circuit and reported to the GK.
- The source carrier ID is passed through the gateway to the terminating leg.

The following examples show a configuration with more reserved calls than the default value for the **max-calls** argument (1000):

Configuration with Default Calls in Excess of 1000

This example assigns 1100 calls to other carriers, leaving 400 calls available to the default carrier:

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 500
  ip circuit carrier-id bb reserved-calls 500
  ip circuit carrier-id cc reserved-calls 100
```

The following examples show the default carrier configured with an incoming source carrier but no voice source group configured.

Note In this example, 800 call legs are implicitly reserved for the default circuit.

Default Carrier and Incoming Source Carrier with No Voice Source Group

Note A gatekeeper is required with carrier-id routing.

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 200
```

If there is no incoming source carrier ID:

- Capacity only is reported to the GK using the default circuit (two call legs).
- No source or destination carrier information is reported.

If there is an incoming source carrier ID called “AA”:

- One call leg is counted against circuit “AA”.
- One call leg (outbound) is counted against the default circuit.
- The source carrier ID is passed through the gateway to the terminating leg.

If there is an incoming source carrier ID called “BB” (for example) or anything other than “AA”:

- Two call legs are counted against the default circuit.
- The source carrier ID “BB” is passed through the gateway to the terminating leg.

The following examples show the first voice source-group match case:

Voice Source-Group Match Case 1

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 200
!
voice source-group 1
 carrier-id source AA
 carrier-id target AA
```

If there is no incoming source carrier ID, the default circuit is used because there is no match in the voice source group.

If there is an incoming source carrier ID called “AA,” the following are in effect:

- The voice source group matches.
- Both call legs are counted against circuit “AA”.
- The source carrier ID is passed through the gateway to the terminating leg.
- The destination carrier ID is “AA”.

The following examples show the second voice source group match case:

Voice Source-Group Match Case 2

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 200
  ip circuit carrier-id BB reserved-calls 200
!
voice source-group 1
 carrier-id source AA
 carrier-id target BB
```

If there is no incoming source carrier ID, the default circuit is used because there is no match in the voice source group.

If there is an incoming source carrier ID called “AA”:

- The voice source-group matches.
- One leg is counted against circuit “AA”.
- One leg is counted against circuit “BB”.
- The source carrier ID is passed through the gateway to the terminating leg.
- The destination carrier ID is “BB”.

The following examples show the third voice source-group match case:

Voice Source-Group Match Case 3

```
voice service voip
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 200
  ip circuit carrier-id BB reserved-calls 200
!
voice source-group 1
 access-list 1
 carrier-id source BB
```

If the access-list matches, the following apply:

- One leg is counted against circuit “BB”.
- One leg is counted against the default circuit (for the destination circuit).
- The source carrier ID is synthesized to “BB” and used to report to the gatekeeper. It is also used on the outgoing setup.

If a source carrier ID is received on the incoming setup, it is overridden with the synthesized carrier ID

Configuring Overlap Signaling for H.323-to-H.323 Connections on a Cisco Unified Border Element

The terminating gateway is responsible for collecting all the called number digits. Overlap signaling is implemented by matching destination patterns on the dial peers. When H.225 signal overlap is configured on the originating gateway, it sends the SETUP to the terminating gateway once a dial-peer match is found. The originating gateway sends all further digits received from the user to the terminating gateway using INFO messages until it receives a sending complete message from the user. The terminating gateway receives the digits in SETUP and subsequent INFO messages and does a dial-peer match. If a match is found, it sends a SETUP with the collected digits to the PSTN. All subsequent digits are sent to the PSTN using INFO messages to complete the call.

To configure overlap signaling in an Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **h225 signal overlap**
6. **h225 timeout t302**
7. **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	**`voice service voip`** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**`h323`** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice-service configuration mode.
Step 5	**`h225 signal overlap`** **Example:** `Router(conf-serv-h323)# h225 signal overlap`	Activates overlap signaling to the destination gateway.

	Command or Action	Purpose
Step 6	**h225 timeout t302** *seconds* **Example:** `Router(conf-serv-h323)# h225 timeout t302 15`	Sets the t302 timer timeout value. The argument is as follows: • *seconds*— Number of seconds for timeouts. Range: 1 to 30.
Step 7	**exit** **Example:** `Router(conf-serv-h323)# exit`	Exits the current mode.

Troubleshooting Tips

Caution Under moderate traffic loads, these **debug** commands produce a high volume of output.

- Use the **debug voip ipipgw** command to debug the Cisco Unified Border Element feature.
- Use any of the following additional **debug** commands on the gateway as appropriate:

H.323 Call-Type Scenarios

  – **debug cch323 all**
  – **debug h225 asn1**
  – **debug h225 events**
  – **debug h225 q931**
  – **debug h245 asn1**
  – **debug h245 events**
  – **debug voip ipipgw**
  – **debug voip ccapi inout**

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

- Cisco IOS Debug Command Reference, Release 12.4T
- *Cisco IOS Voice Troubleshooting and Monitoring Guide*
- *Troubleshooting and Debugging VoIP Call Basics*
- *Voice Gateway Error Decoder for Cisco IOS*
- *VoIP Debug Commands*

Verifying H.323-to-H.323 Cisco Unified Border Element Configuration and Operation

To verify Cisco Unified Border Element feature configuration and operation, perform the following steps (listed alphabetically) as appropriate.

Note The word "calls" refers to call legs in some commands and output.

SUMMARY STEPS

1. **show call active video**
2. **show call active voice**
3. **show call history fax**
4. **show call history video**
5. **show call history voice**
6. **show crm**
7. **show dial-peer voice**
8. **show running-config**
9. **show voip rtp connections**

DETAILED STEPS

Step 1 **show call active video**

Use this command to display the active video H.323 call legs.

Step 2 **show call active voice**

Use this command to display call information for voice calls that are in progress.

Step 3 **show call active fax**

Use this command to display the fax transmissions that are in progress.

Step 4 **show call history video**

Use this command to display the history of video H.323 call legs.

Step 5 **show call history voice**

Use this command to display the history of voice call legs.

Step 6 **show call history fax**

Use this command to display the call history table for fax transmissions that are in progress.

Step 7 **show crm**

Use this command to display the carrier ID list or IP circuit utilization.

Step 8 **show dial-peer voice**

Use this command to display information about voice dial peers.

Step 9 **show running-config**

Use this command to verify which H.323-to-H.323, H.323-to-SIP, or SIP-to-SIP connection types are supported.

Step 10 **show voip rtp connections**

Use this command to display active Real-Time Transport Protocol (RTP) connections.

Where to Go Next

- 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
- Configuring Cisco Unified Border Element Videoconferencing

Additional References

The following sections provide references related to H.323-to-H.323 Cisco UBE Connections:

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 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 • Cisco IOS Release 12.4T Configuration Guides

Related Topic	Document Title
Cisco IOS Release 12.3	• Cisco IOS Release 12.3 documentation • Cisco IOS Voice Troubleshooting and Monitoring Guide • 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 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 • *GKTMP Messages*: http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_tmp.html
internet Low Bitrate Codec (iLBC) Documents	• Codecs section of the Dial Peer Configuration on Voice Gateway Routers Guide http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_ovrvw.html • Dial Peer Features and Configuration section of the Dial Peer Configuration on Voice Gateway Routers Guide 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 • Remote-to-local network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edc.shtml • Remote-to-remote network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edd.shtml • 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 • Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide • "Configuring T.38 Fax Relay" chapter • Cisco IOS SIP Configuration Guide • Cisco Unified Communications Manager (CallManager) Programming Guides • Quality of Service for Voice over IP

Related Topic	Document Title
Related Platform Documents	• Cisco 2600 Series Multiservice Platforms • Cisco 2800 Series Integrated Services Routers • Cisco 3600 Series Multiservice Platforms • Cisco 3700 Series Multiservice Access Routers • Cisco 3800 Series Integrated Services Routers • Cisco 7200 Series Routers • Cisco 7301
Related gateway configuration documentation	Media and Signaling Authentication and Encryption Feature for Cisco IOS H.323 Gateways. http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/htsecure.htm
Cisco IOS NAT Configuration Guide, Release 12.4T	*Configuring Cisco IOS Hosted NAT Traversal for Session Border Controller* 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 http://www.cisco.com/en/US/docs/ios/debug/command/reference/db_book.html • *Troubleshooting and Debugging VoIP Call Basics* at http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml • *VoIP Debug Commands* at 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 • CISCO-VOICE-DIAL-CONTROL-MIB • IP-TAP-MIB • TAP2-MIB • 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: 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 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. 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. 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 H.323-to-H.323 Cisco Unified Border Element Connections

Table 1 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"

Note Table 1 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 1 Feature Information for H.323-to-H.323 Gateway Connections

Feature Name	Releases	Feature Information
Call Admission Control.	12.4(6)T	This feature was introduced.
Cisco Unified Communications Manager Connections	12.4(6)T	No MTP for Cisco Unified Communications Manager Trunks to Cisco Unified Border Element.
Cisco UBE MIB support	15.0(1)XA	This feature was introduced.
Codec Support	12.4(11)T	Configuring iLBC Codec on an Cisco Unified Border Element
DTMF	12.4(11)T	G.711 Inband DTMF to RFC 283.
H.323-to-H.323 Connections on a Cisco Unified Border Element	12.3(1)	H.323-to-H.323 Gateway configuration provides a network-to-network demarcation point between independent VoIP and video networks by for billing, security, call-admission control, QoS, and signaling interworking.
Managing H.323 IP Group Call Capacities	12.2(13)T	Creates a maximum capacity for the IP group providing extra control for load and resource balancing.
Media Modes	12.3(1)	Media Flow-Around.Adds media flow-around capability on the Cisco UBE by supporting the processing of call set-up and teardown request (VoIP call signaling) and for media streams (flow-through and flow-around) Improves scalability and performance when network-topology hiding and bearer-level interworking features are not required.
Overlap Signaling for H.323-to-H.323 Connections on a Cisco Unified Border Element	12.3(11)T	The terminating gateway is responsible for collecting all the called number digits. Overlap signaling is implemented by matching destination patterns on the dial peers.

Table 1 Feature Information for H.323-to-H.323 Gateway Connections (continued)

Feature Name	Releases	Feature Information
Rotary Support	12.3(11)T 12.4(6)T	12.3(11)T—H.323-to-H.323 Call Failure Recovery (Rotary) on a Cisco Unified Border Element. Eliminates codec restrictions and enables the Cisco UBE to restart codec negotiation with the originating endpoint based on the codec capabilities of the next dial peer in the rotary group for H.323-to-H.323 interconnections. 12.4(6)T—Secure RTP with IPSEC for Signaling.
Signal Interworking	12.3(11)T	H.323-to-H.323 Interworking Between Fast Start and Slow Start. This feature enables the Cisco UBE to bridge calls between VoIP endpoints that support only H.323 FastStart procedures and endpoints that support only normal H.245 signaling (SlowStart).