



Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

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This chapter describes how to configure and enable features for H.323 Cisco Unified Communications Manager to H.323 service provider connections in a Cisco Unified Border Element topology. 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 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/ios124/124tgc/vcl.htm>.



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Prerequisites for Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity



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>.

- 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.

In order to interconnect with an Cisco UBE, configure Cisco Unified Communications Manager with the following:

- Cisco Unified Communications Manager 3.0 or later release.



Note The Cisco Unified Communications Manager 4.1 has a SIP trunk towards the Cisco UBE in an H.323-to-SIP or a SIP-to-SIP network.

- Media termination point (MTP)—Enables the Cisco Unified Communications Manager to extend supplementary services, such as hold and transfer, to calls that are routed through an H.323 endpoint or an H.323 gateway.
 - Cisco IOS Release 12.4(4)T and earlier releases: MTP is required.
 - Cisco IOS Release 12.4(6)T and later releases: MTP is optional for H.323-to-H.323 calls.
- Intercluster Trunk (ICT)—An H.323 connection that enables multiple Cisco Unified Communications Manager systems to be connected over an IP cloud.
- Configure Cisco Unified Communications Manager in gateway mode while interoperating with the Cisco Unified Border Element.

**Note**

- During ICT configuration on the Cisco Unified Communications Manager, you are asked to enter the IP address of the remote Cisco Unified Communications Manager to which the ICT connects. Do not use this IP address. Instead, enter the IP address of the Cisco UBE.
- For more information on MTP, ICT, and Cisco Unified Communications Manager configuration, see the Cisco Unified Communications Manager documentation at http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm

Restrictions for Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

- This functionality is enabled by default in Cisco IOS images that support the Cisco UBE, so you do not need to enter the commands that follow. However, the commands do appear in software and are explained here for informational purposes.
- Cisco Unified Communications Manager compatibility is as follows:
 - Cisco IOS Release 12.3(7)T and later releases: You must enable Cisco Unified Communications Manager compatibility on the Cisco UBE.
- 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.
- Ringback to the transferee is not supported by the Cisco Unified Border Element Software for calls blind call transferred between Cisco Unified Communications Manager and Cisco Unified Communications Express.

Information About Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

For more information on Cisco Unified Communications Manager configuration, see the Cisco Unified Communications Manager documentation at http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm

How to Configure Cisco Unified Border Elements for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity.

The section contains the following tasks:

- [Configuring Cisco Unified Border Element to Transport Calls With Cisco Unified Cisco Unified Communications Manager, page 278](#)
- [Configuring DTMF Relay Digit-Drop on an Cisco Unified Border Element with Cisco Unified Communications Manager, page 281](#)
- [Configuring H.323 Calling Name Display, page 282](#)

Configuring Cisco Unified Border Element to Transport Calls With Cisco Unified Cisco Unified Communications Manager

To configure the Cisco UBE to transport calls to and from Cisco Unified Communications Manager, perform the steps in this section.

Prerequisites

Configure Cisco Unified Communications Manager with the following:

- Media termination point (MTP)—Enables Cisco Unified Communications Manager to extend supplementary services, such as hold and transfer, to calls that are routed through an H.323 endpoint or an H.323 gateway. Cisco IOS Release 12.4(6)T and later releases: No MTP is optional.
- Intercluster trunk (ICT)—An H.323 connection that enables multiple Cisco Unified Communications Manager systems to be connected over an IP cloud.

**Note**

During ICT configuration, you are asked to enter the IP address of the remote Cisco Unified Communications Manager to which the ICT connects. Do not use this IP address. Instead, enter the IP address of the Cisco UBE.

Restrictions

Cisco UBE does not support call preservation. If an Cisco UBE is located between a gateway and a Cisco Unified Communications Manager that have call preservation configured and the Cisco UBE is configured with media flow-around, calls will be preserved on the gateway when the Cisco UBE's IP interface becomes inaccessible due to a network issue or a reload. In this case, call preservation behavior on the gateway will be the same as the case where there is no Cisco UBE between the gateway and the Cisco Unified Communications Manager. Calls that are “held” will not be preserved as the Cisco UBE passes the nonstandard “do not preserve” indication sent in the Notify message from Cisco Unified Communications Manager to the gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **h225 h245-address**
6. **ccm-compatible**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • 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	<p>h225 h245-address {sync facility progress}</p> <p>Example: Router(config-serv-h323)# h225 h245-address</p>	<p>Enables H.225 and H.245 signaling. Keywords are as follows:</p> <ul style="list-style-type: none"> • sync—Synchronized H.245 address reporting. Timing of H.245 messages is controlled by endpoints, not by the Cisco UBE. • facility—Cisco UBE sends a message to instruct the originating Cisco Unified Communications Manager to begin H.245 procedures on call start. • progress—Cisco UBE sends a Progress message that includes the H.245 address and a Progress Indicator value of 0x03.
Step 6	<p>ccm-compatible</p> <p>Example: Router(config-serv-h323)# ccm-compatible</p>	<p>Enables Cisco Unified Communications Manager-compatible signaling.</p>
Step 7	<p>exit</p> <p>Example: Router(config-serv-h323)# exit</p>	<p>Exits the current mode.</p>

Troubleshooting Tips

In the event that calls setup from Cisco Unified Communications Manager to Cisco Unified CallManager Express have no audio, verify the following:

- Interwork between Fast Start and Slow Start is not supported by the **call start interwork** command.
- When **call start interwork** is configured, both incoming and outgoing dial-peer need to have a specific codec configured.

Configuring DTMF Relay Digit-Drop on an Cisco Unified Border Element with Cisco Unified Communications Manager

To avoid sending both in-band and out-of band tones to the outgoing leg when sending Cisco UBE calls in-band (rtp-nte) to out-of band (h245-alphanumeric), configure the **dtmf-relay rtp-nte digit-drop** command on the incoming SIP dial-peer. On the H.323 side configure either **dtmf-relay h245-alphanumeric** or **dtmf-relay h245-signal** command. This feature may also be used for H.323-to-SIP, and H.323-to-H.323 calls.



Note

For a SIP (rtp-nte) to H.323 (h245-alphanumeric) via Cisco UBE call, if any RTP-NTE packets are sent before the H.323 Endpoint answers the call, the dual-tone multifrequency (DTMF) signal is not audible on a terminating gateway (TGW)

Perform the following task to configure DTMF relay digit drop on an Cisco UBE with Cisco Unified Communications Manager.

Restrictions

- Configuring the **digit-drop** command is required for interworking between OOB and RTP NTE.
- Digit-drop for in-band rtp-nte DTMF conversion requiring a transcoder is not supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **dtmf-relay [cisco-rtp] [h245-alphanumeric] [h245-signal][rtp-nte [digit-drop]]**
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 tag voip Example: Router(config)# dial-peer voice 2 voip	Enters dial-peer voice configuration mode for the specified VoIP dial peer.

	Command or Action	Purpose
Step 4	<pre>dtmf-relay [cisco-rtp] [h245-alphanumeric] [h245-signal] [rtp-nte digit-drop]]</pre> <p>Example: Router (config-dial-peer)# dtmf-relay rtp-nte digit-drop</p>	<p>Forwards DTMF tones. Keywords are as follows:</p> <ul style="list-style-type: none"> • cisco-rtp—Forwards DTMF tones by using RTP with a Cisco-proprietary payload type. • h245-alphanumeric—Forwards DTMF tones by using the H.245 alphanumeric method. • h245-signal—Forwards DTMF tones by using the H.245 signal UII method. • rtp-nte—Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type. • digit-drop—Passes digits out-of-band; and drops in-band digits. <p>Note The digit-drop keyword is available only when the rtp-nte keyword is configured.</p>
Step 5	<pre>exit</pre> <p>Example: Router (config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Examples

The following example shows DTMF-Relay digits configured to avoid sending both in-band and out-of-band tones to the outgoing leg in an Cisco UBE:

```
.
.
.
dial-peer voice 1 voip
 dtmf-relay h245-alphanumeric rtp-nte digit-drop
.
.
.
```

Troubleshooting tips

- The debug output will show that the H245 out of band messages are sent to the TGW. However, entry of the digits are not audible on the phone.

Configuring H.323 Calling Name Display

The H.323 Calling Name Display feature provides a configurable option on the Cisco gateway to send the calling name received in Q931 Facility messages in the Display IE of setup message or Display IE of H.225 Notify messages to the Cisco Unified Communications Manager so that it can interpret the calling name and display it on Cisco IP Phones.

Prerequisites

- This software operation is transparent to Cisco Unified Communications Manager and works with all releases, although Cisco Unified Communications Manager 4.2 or later is recommended
- The **isdn supp-service name calling** command must be configured under isdn D-channel for the H.323 Calling Name Display feature to work.
- It is recommended that you configure the **signaling forward unconditional** command on the outgoing gateway. This ensures that the outgoing H225 message has both the raw message and the Generic Transparency Descriptor (GTD).
- To configure the H.323 Calling Name Display feature under the voice class configuration, complete your dial-peer configuration first. Additional dial peer configuration information is available in the *Dial Peer Configuration on Voice Gateway Routers Guide* at the following url:
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/int_c/dpeer_c/dp_conf.htm

Restrictions

- When a Cisco gateway is connected to ISDN switch that sends the name-to-follow in Q931 Setup message, and the calling name in subsequent Q931 Facility message, It is recommended that you configure **h225 timeout ntf** command to buffer the setup message for the necessary interval.
- Calling name handling for various messages under ISDN, while interoperating with different switch types is not supported.

Information About H.323 Calling Name Display

Calling name display information can be populated in ISDN messages in the Display Information Element (IE) of a Q.931 Setup or Notify message, or in the Facility IE of a Q.931 Setup or Facility message. The Cisco IOS gateway places this information into the same field of the corresponding H.323 message.

Cisco Unified Communications Manager (Cisco Unified CM) interprets calling name information (for purposes of name display on IP phones registered with Unified CM only in the Display IE of the H.323 Setup and Notify messages. Name display information delivered in an H.323 Facility message is not interpreted by Cisco Unified CM. Some ISDN switch types (for example, NI2) send a “name-to-follow” indication in the Q.931 Setup message and deliver the calling name subsequently in the Facility IE of a Q.931 Facility message. When a Cisco IOS gateway is connected to such an ISDN switch and interoperating with Cisco Unified CM by using the H.323 protocol, Cisco Unified CM is unable to display calling names on IP phones.

Beginning with Cisco IOS Release 12.4(11)XW, two new modes of operation are introduced on Cisco IOS gateway:

H.323 Calling Name Display Without Buffering

When a Q.931 Setup message with a "name-to-follow" indication is received from an ISDN switch, an H.323 Setup message with no name information is sent to Cisco Unified CM. When the subsequent Q.931 Facility message is received with the calling name information, the message is mapped by the gateway to an H.225 Notify message with the Display IE populated with the calling name, so that the Cisco Unified CM can interpret the message correctly and display the calling name on the IP Phone.

H.323 Calling Name Display With Buffering

When a Q.931 Setup message with a “name-to-follow” indication is received from an ISDN switch, the gateway can buffer the setup message until the subsequent Q.931 Facility message with calling name information is received. The name information from the Q.931 Facility message is now placed into the H.323 Setup message Display IE and sent to Cisco Unified CM. If the buffer timer expires before the Q.931 Facility message is received, a H.323 Setup is sent with no name information and, if it subsequently arrives, the information is sent by using an H.225 Notify message.

How to Enable H.323 Calling Name Display

H.323 Calling name display is configured with or without buffering. This section contains the following subsections:

- [Configuring H.323 Calling Name Display Without Buffering, page 284](#)
- [Configuring H.323 Calling Name Display With Buffering, page 287](#)

Configuring H.323 Calling Name Display Without Buffering

To enable the H.323 Calling Name Display feature without buffering for ISDN trunks that use the Facility message to deliver Name Display information, perform the steps in this section. This section contains the following subsections

- [Configuring H.323 Calling Name Display Without Buffering at the Voice Service Level, page 284](#)
- [Configuring H.323 Calling Name Display Without Buffering at the Voice Class Level, page 285](#)

Configuring H.323 Calling Name Display Without Buffering at the Voice Service Level

To configure H.323 Calling Name Display without buffering at the voice service level, perform the steps in this section:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **h225 display-ie ccm-compatible**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: Router(config)# voice service voip	Enters VoIP voice-service configuration mode.
Step 4	<code>h323</code> Example: Router(conf-voi-serv)# h323	Enters H.323 voice-service configuration mode
Step 5	<code>h225 display-ie ccm-compatible</code> Example: Router(conf-voi-serv)# h225 display-ie ccm-compatible	Interprets the H.323 Notify Display IE so that the IP Phone can display the calling name on the IP Phone.

Configuring H.323 Calling Name Display Without Buffering at the Voice Class Level

Behavior and configuration will vary based on the configuration mode the command is configured. When this CLI is configured under voice class:

- When the **h225 display-ie ccm-compatible** command is configured at the global level, and **h225 display-ie ccm-compatible system** command is configured at voice class level under dial-peer, both facility and notify messages are sent.
- When the **h225 display-ie ccm-compatible** command is not configured at global level, and the **h225 display-ie ccm-compatible system** command is configured at voice class level under dial-peer, only the facility message is sent.
- The configured command is visible in the **show running-configuration** output under voice class.
- Configuring the **no h225 display-ie ccm-compatible system** command in voice class configuration mode, the command that is configured under voice class takes precedence. Even when the **no h225 display-ie ccm-compatible system** command is configured under voice class and the **h225 display-ie ccm-compatible** command is configured under voice service voip, the gateway will not send the H225 Notify message received, and the calling name does not display on the IP Phone.

Use the **no** version to disable sending H225 Notify message on a particular VoIP dial-peer. The **no** form of the command is shown under voice class in the **show running-configuration**.

Prerequisites

To configure the H.323 Calling Name Display feature under the voice class configuration, complete your dial-peer configuration first. Additional dial peer configuration information is available in the *Dial Peer Configuration on Voice Gateway Routers Guide* at the following url:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/int_c/dpeer_c/dp_conf.htm

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **voice class h323 tag**
5. **h225 display-ie ccm-compatible**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • 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	voice class h323 tag Example: Router(conf-voi-serv)# voice class h323 1	Enters H.323 voice-service configuration mode
Step 5	h225 display-ie ccm-compatible Example: Router(conf-voi-serv)# h225 display-ie ccm-compatible	Interprets the H.323 Notify Display IE so that the IP Phone can display the calling name on the IP Phone.

Configuring H.323 Calling Name Display With Buffering

To enable the H.323 Calling Name Display feature with buffering for ISDN trunks that use the Facility message to deliver Name Display information, perform the steps in this section. This section contains the following subsections

- [Configuring H.323 Calling Name Display with Buffering at the Voice Service Level, page 287](#)
- [Configuring H.323 Calling Name Display with Buffering at the Voice Class Level, page 288](#)

Configuring H.323 Calling Name Display with Buffering at the Voice Service Level

To configure H.323 calling name display at the voice service level, perform the steps in this section.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `h323`
5. `h225 timeout ntf milliseconds`
6. `h225 display-ie ccm-compatible`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> <code>enable</code>	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# <code>configure terminal</code>	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: Router(config)# <code>voice service voip</code>	Enters VoIP voice-service configuration mode.
Step 4	<code>h323</code> Example: Router(conf-voi-serv)# <code>h323</code>	Enters the H.323 voice-service configuration mode.

	Command or Action	Purpose
Step 5	h225 timeout ntf <i>milliseconds</i> Example: Router#(conf-voi-h323)# h225 timeout ntf 300	Buffers the setup message until the Q.931 Facility message with calling name information is received. Range is 50-5000 milliseconds.
Step 6	h225 display-ie ccm-compatible Example: Router#(conf-if)# h225 display-ie ccm-compatible	Interprets the H.323 Notify Display IE so that the IP Phone can display the calling name on the IP Phone.

Configuring H.323 Calling Name Display with Buffering at the Voice Class Level

To configure H.323 Calling Name Display at the voice class level, perform the steps in this section.

Prerequisites

To configure the H.323 Calling Name Display feature under the voice class configuration, complete your dial-peer configuration first. Additional dial peer configuration information is available in the *Dial Peer Configuration on Voice Gateway Routers Guide* at the following url:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgr/vvfax_c/int_c/dpeer_c/dp_cfg.htm

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class h323 *tag***
4. **h225 display-ie ccm-compatible system**
5. **h225 timeout ntf *milliseconds***

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice class h323 <i>tag</i> Example: Router(config)# voice service voip	Creates an H.323 voice class that is independent of a dial peer and can be used on multiple dial peers. Range is from 1 to 10000. There is no default value.

	Command or Action	Purpose
Step 4	<pre>h225 display-ie ccm-compatible system</pre> <p>Example: Router#(conf-voi-h323)# h225 display-ie ccm-compatible system</p>	Interprets the H.323 Notify Display IE so that the IP Phone can display the calling name on the IP Phone.
Step 5	<pre>h225 timeout ntf milliseconds</pre> <p>Example: Router#(conf-voi-h323)# h225 timeout ntf 300</p>	<p>Buffers the setup message until the Q.931Facility message with calling name information is received. Range is 50-5000 milliseconds.</p> <p>Note In the event the facility is received before the timer expires, the gateway will stop the buffer timer, extract the relevant information and send it to terminating endpoint.</p>

Troubleshooting Tips

- Enable debug isdn q931 to see the calling name coming into the GW in ISDN messages.
- Enable debug h225 q931 to see the calling name sent out by GW in H225 messages.

Configuration Examples for Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

The following section contains configuration examples for the following:

- [Cisco Unified Border Element and Cisco Unified Communications Manager: Example](#)

Cisco Unified Border Element and Cisco Unified Communications Manager: Example

Cisco UBE interconnects with Cisco Unified Communications Manager, providing a billing and network demarcation point and enabling service providers to transport calls to and from enterprise customers who use Cisco Unified Communications Manager.

H.245 handles end-to-end control messages between H.323 entities. H.245 procedures establish logical channels for transmission of audio and control channel information and are used to negotiate channel usage, flow control, and capabilities exchange messages.

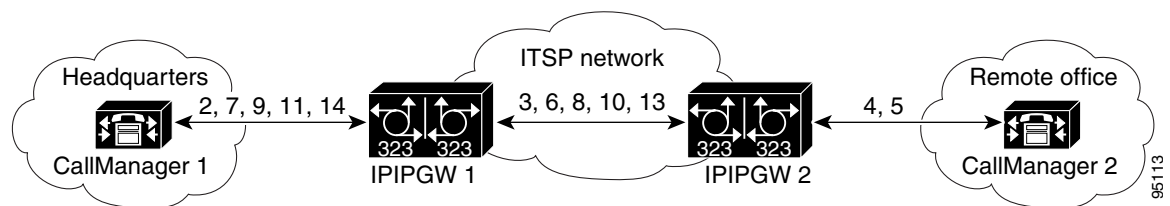
In releases earlier than Cisco IOS Release 12.3(1), the Cisco UBE responded to an incoming slow-start SETUP message with a CALLPROC message containing an H.245 address. This behavior is incompatible with the manner in which Cisco Unified Communications Manager handles H.245 messages. With Cisco IOS Release 12.3(1) and later releases, the Cisco UBE delays the reporting of originating-side H.245 address until a terminating-side H.245 address is received. Thus H.245 procedures are allowed to advance only when the called party is ready to do so.

When a call is set up, the Cisco UBE sends an H.225 message to the originator, instructing it to start H.245 procedures if they have not started by the time the CONNECT message is received from the terminating side. For slow-start calls in this scenario, a FACILITY message is sent containing an H.245 address. Some originating gateways running earlier Cisco IOS releases are unable to process this FACILITY message. To ensure interoperability with these older images, a PROGRESS message is sent containing an H.245 address and a Progress Indicator value of 0x03.

The functionality described above is enabled by default on Cisco UBEs running Cisco IOS Release 12.3(1) and later releases. It can be modified though the use of a new Cisco IOS command (**h225 h245-address facility | on-connect | progress**).

Figure 1 shows a sample topology that connects Cisco Unified Communications Manager systems to an Cisco UBE.

Figure 1 Cisco UBE and Cisco Unified Communications Manager Topology



The following scenario illustrates a basic call placed from a company's headquarters to its remote office using Cisco Unified Communications Manager and two Cisco UBEs.

1. A caller at headquarters uses an IP phone to call someone at the remote office.
2. Cisco Unified Communications Manager 1 recognizes the called number as an extension at the remote office and sends a SETUP message to Cisco UBE 1.
3. The Cisco UBE, using the ITSP network, sends a SETUP message to Cisco UBE 2. Cisco UBE 1 sends a CALLPROC message to Cisco Unified Communications Manager 1.
4. At the remote office, Cisco UBE 2 sends a SETUP message to Cisco Unified Communications Manager 2 and sends a CALLPROC message to Cisco UBE 1.
5. Cisco Unified Communications Manager 2 rings the extension of the called party and sends an ALERT message with the H.245 address to Cisco UBE 2.
6. Cisco UBE 2 sends an ALERT message with the H.245 address to Cisco UBE 1.
7. Cisco UBE 1 sends an ALERT message with the H.245 address to Cisco Unified Communications Manager 1.
8. Cisco UBE 2 sends a FACILITY message with the H.245 address to Cisco UBE 1.
9. Cisco UBE 1 sends a FACILITY message with the H.245 address to Cisco Unified Communications Manager 1.
10. Cisco UBE 2 sends a PROGRESS message with the H.245 address to Cisco UBE 1.
11. Cisco UBE 1 sends a PROGRESS message with the H.245 address to Cisco Unified Communications Manager 1.
12. The two Cisco Unified Communications Manager systems exchange capabilities and open logical channel messages, and they engage in master/slave determination (not shown in Figure 1.).

13. The called party answers the extension, and Cisco UBE 2 sends a CONNECT message with the H.245 address to Cisco UBE 1.
14. Cisco UBE 1 sends a CONNECT message with the H.245 address to Cisco Unified Communications Manager 1.

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](#)
- [Configuring Cisco Unified Border Element Videoconferencing](#)

Additional References

The following sections provide references related to Cisco Unified Border Element.

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	<ul style="list-style-type: none"> • Cisco IOS Release 12.4 Configuration Guides • Cisco IOS Release 12.4T Configuration Guides
Cisco IOS Release 12.3	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • <i>GKTMP Command Reference:</i> http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_cli.htm • <i>GKTMP Messages:</i> 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	<ul style="list-style-type: none"> • 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_config.html
Cisco Unified Border Element Configuration Examples	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • 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 Platform Documents	<ul style="list-style-type: none"> • 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	<p>Media and Signaling Authentication and Encryption Feature for Cisco IOS H.323 Gateways.</p> <p>http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/htsecure.htm</p>

Related Topic	Document Title
Cisco IOS NAT Configuration Guide, Release 12.4T	<p><i>Configuring Cisco IOS Hosted NAT Traversal for Session Border Controller</i></p> <p>http://www.cisco.com/en/US/docs/ios/12_4t/ip_addr/configuration/guide/htnatsbc.html</p>
Troubleshooting and Debugging guides	<ul style="list-style-type: none"> • Cisco IOS Debug Command Reference, Release 12.4 at http://www.cisco.com/en/US/docs/ios/debug/command/reference/db_book.html • <i>Troubleshooting and Debugging VoIP Call Basics</i> at http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml • <i>VoIP Debug Commands</i> 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	<i>H.323 (ITU-T VOIP protocols)</i>
H.323 - H.245 Version 12, Annex R	<i>H.323 (ITU-T VOIP protocols)</i>

MIBs

MIB	MIBs Link
<ul style="list-style-type: none"> • CISCO-DSP-MGMT-MIB • CISCO-VOICE-DIAL-CONTROL-MIB • IP-TAP-MIB • TAP2-MIB • USER-CONNECTION-TAP-MIB 	<p>To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:</p> <p>http://www.cisco.com/go/mibs</p>

RFCs

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

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

Technical Assistance

Description	Link
<p>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</p> <p>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.</p> <p>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</p>	http://www.cisco.com/cisco/web/support/index.html

Feature Information for Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

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 Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity

Feature Name	Releases	Feature Information
H.323 Calling Name Display	12.4(11)XW 12.4(20)T	Provides a configurable option on the Cisco Gateway so that the display IE received in Q931 Facility message, is sent out to Cisco Unified Communications Manager (Cisco Unified CM), in H.225 Notify message. This way Cisco Unified Cisco Unified CM can interpret it correctly, and display calling name on the IP phones. The following commands were introduced by this feature: h225 timeout ntf and h225 display-ie ccm-compatible
DTMF Relay Digit-Drop on an Cisco Unified Border Element with Cisco Unified Communications Manager	12.4(11)T	This feature passes DTMF tones out-of-band and drops in-band digits to avoid sending both tones to the outgoing leg on an H.323-to-SIP Cisco Unified Border Element.
Cisco Unified Communications Manager Connections	12.3(1) 12.3(8)T	Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity This feature provides interoperability with Cisco Unified Communications Manager for basic calls, caller-ID services, supplementary services, and RSVP synchronization with audio. 12.3(8)T—Restriction for Video Calls made from a Cisco Unified Communications Manager to an Cisco Unified Border Element.

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