SIP-to-SIP Connections on a Cisco Unified Border Element

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This chapter describes how to configure and enable features for SIP-to-SIP connections in an Cisco Unified Border Element topology. A Cisco Unified Border Element (Cisco UBE), 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/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm.

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Contents

This chapter describes how to configure SIP-to-SIP connections in a Cisco Unified Border Element (Cisco UBE). It covers the following features:

Prerequisites for Configuring SIP-to-SIP Connections on a Cisco Unified Border Element

- Perform the prerequisites listed in the "Prerequisites for Cisco Unified Border Element Configuration" procedure on page -22 in this guide.
- Perform fundamental gateway configuration listed in the "Prerequisites for Fundamental Cisco Unified Border Element Configuration" procedure on page -48 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 SIP-to-SIP Connections on a Cisco Unified Border Element

Cisco IOS Release 12.4(15)XY and later releases:

- Registration is not supported.

Cisco IOS Release 12.4(15)T and before:

- Delayed-Offer to Delayed-Offer is not supported.
- Codec T is not supported.
- Registration is not supported.
- Supplementary services are not supported.

- Transcoding is not supported.
- Like-to-like error messages are not passed from the incoming SIP leg to the outgoing SIP leg.

Cisco IOS Release 12.4(9)T and before:

- Topology and address hiding is not supported.

Cisco IOS Release 12.4(9)T and later releases:

- Media flow-around for Delayed-Offer to Early-Offer audio and video calls 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.

ptime attributes

- SIP gateway supports one ptime attribute per media line.
- Cisco UBE supports ptime attribute when one codec is offered. The ptime attribute is not sent when multiple codecs are offered by the Cico UBE.
- The default behavior of the Cisco UBE is to select the minimum ptime value from the offer and prefer. Results are unpredictable when dissimilar networks with different packetization time periods are connected.

Information About Configuring SIP-to-SIP Connections on a Cisco Unified Border Element

Note When you configure SIP on a router, the ports on all its interfaces are open by default. This makes the router vulnerable to malicious attackers who can execute toll fraud across the gateway if the router has a public IP address and a public switched telephone network (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.

- Delayed-Offer to Early-Offer audio calls are supported.
- Delayed-Offer to Delayed-Offer calls are supported.
- Delayed-Offer to Delayed-Offer video calls are supported in Cisco IOS Release 12.4(15)XY and later.
- Delayed-Offer to Delayed-Offer audio calls are supported in Cisco IOS Release 12.4(15)T and later.
- Early-Offer to Early-Offer for audio calls are supported.
- Early-Offer to Early-Offer, Delayed-Offer to Early-Offer video calls are supported in 12.4(15)XZ and later.
- Fax relay is enabled by default for all systems. No further configuration is needed.
- Like-to-like dtmf, codec and fax are supported.
- Like-to-like error messages are not passed from the incoming SIP leg to the outgoing SIP leg. Error messages are passed through Cisco Unified BE when the **header-passing error-passthru** command is configured in Cisco IOS Release 12.4(15) T and later.

- Media flow-around (except for Delayed-Offer to Early-Offer audio and video calls) in Cisco IOS Release 12.4(9)T and later.
- reINVITE pass-through for Session Refresh is supported.
- SIP-to-SIP Video (including Delayed-Offer to Delayed-Offer, Early-Offer to Early-Offer, Delayed-Offer to Early-Offer calls) are supported.
- SRTP-to-SRTP support for SIP-to-SIP calls is supported.

How to Configure SIP-to-SIP Gateway Features

The following section provides configuration information for the following SIP-to-SIP features.

SIP-to-SIP Basic Functionality for Session Border Controller (SBC)

SIP-to-SIP Basic Functionality for SBC for Cisco UBE provides termination and reorigination of both signaling and media between VoIP and video networks using SIP signaling in conformance with RFC3261. The SIP-to-SIP protocol interworking capabilities of the Cisco Unified Border Element (Cisco UBE) support the following:

- Basic voice calls (Supported audio codecs include: G.711, G.729, G.728, G.726, G.723, G.722, AAC_LD, iLBC. Video codecs: H.263, and H.264)
- Codec transcoding
- Calling/called name and number
- DTMF relay interworking
  - SIP RFC 2833 <–> SIP RFC 2833
  - SIP Notify <–> SIP Notify
- Interworking between SIP early-media and SIP early-media signaling
- Interworking between SIP delayed-media and SIP delayed-media signaling
- RADIUS call-accounting records
- RSVP synchronized with call signaling
- SIP-SIP Video calls
- TCL IVR 2.0 for SIP, including media playout and digit collection (RFC 2833 DTMF relay)
- T.38 fax relay and Cisco fax relay
- UDP and TCP transport

SIP-to-SIP Extended Feature Functionality for Session Border Controller (SBC)

Enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP UAs. New SIP-to-SIP features available include:

- Call Admission Control (based on CPU, memory, total calls)
- Delayed Media Call
- ENUM support
- Configuring SIP Error Message Pass Through
- Interoperability with Cisco Unified Communications Manager 5.0 and BroadSoft.
- Lawful Intercept
- Media Inactivity
- Modem passthrough
- TCP and UDP interworking
- Tcl scripts with SIP NOTIFY VoiceXML with SIP-to-SIP
- Transport Layer Security (TLS)

SIP-to-SIP Supplementary Feature Interworking for Session Border Controller (SBC)

Provides enhanced termination and re-origination of signaling and media between VoIP and Video Networks in conformance with RFC3261. New SIP-to-SIP capabilities offered in this release on the Cisco 28xx, 38xx, 5350XM and 5400XM include:

- iLBC Codec

  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
- G.711 Inband DTMF to RFC 2833
- Session refresh
- SIP-to-SIP Supplementary Services
  - Refer/302 Based Supplementary Services Supported from 12.4(9)T onwards
  - ReInvite Based Supplementary Services Supported from 12.4(15)XZ

SIP-to-SIP Supplementary Services for Session Border Controller (SBC)

This chapter describes the SIP-to-SIP supplementary service features for SBC. The SIP-to-SIP supplementary services feature enhances terminating and re-originating both signaling and media between VoIP and Video networks by supporting the following features:

- IP Address Hiding in all SIP messages including supplementary services
- Media
  - Media Flow Around
- Support on Cisco AS5350XM and Cisco AS5400XM

- SIP-to-SIP Supplementary services using REFER/3xx method. The following features are enabled by default.
  - Message Waiting Indication
  - Call Waiting
  - Call Transfer (Blind, Consult, Alerting)
  - Call Forward (All, Busy, No Answer)
  - Distinctive Ringing
  - Call Hold/Resume
  - Music on Hold
- Hosted NAT Traversal for SIP

Configuring IP Address-Hiding

Configuring address-hiding hides signaling and media peer addresses from the endpoints, especially for supplemental services when the Cisco Unified BE passes REFER/3xx messages from leg to leg. Configuring the address hiding feature ensures that the Cisco Unified BE is the only point of signaling and media entry/exit in all scenarios. To enable address-hiding in all SIP messages, perform the steps in this section.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(9)T or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

When supplementary services are configured the endpoint sends messages to the SBC, this is then forwarded to the peer endpoint. Address-hiding is preserved during this message forwarding

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **address-hiding**
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	**address-hiding** **Example:** `Router(conf-voi-serv)# address-hiding`	Hides signaling and media peer addresses from the endpoints.
Step 5	**exit** **Example:** `Router(conf-voi-serv)# exit`	Exits the current mode.

Configuring SIP-to-SIP Connections on a Cisco Unified Border Element

To configure SIP-to-SIP connection types, perform the steps in this section.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.3(1) or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

- Connections are disabled by default in Cisco IOS images that support the Cisco UBE.
- This chapter covers only those features that require a unique configuration in order to support the Cisco UBE. For information on those H.323 gateway features not mentioned in this chapter, see the *Cisco IOS Voice, Video, and Fax Configuration Guide*.

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice service voip**
4. **allow-connections**
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 sip to sip`	Allows connections between specific types of endpoints in an Cisco UBE. 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.

Configuring Delayed-Offer to Early-Offer for SIP Audio 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.

Prerequisites

- To enable this feature, you must have Cisco IOS Release 12.4(15)XZ or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

- 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 on a Cisco Unified Border Element" section on page 190 of this 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 1 shows a list of protocol interworking for SIP.

Table 1 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

How to Configure Delayed-Offer to Early-Offer for SIP Audio Calls

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 192
- Configuring Delayed-Offer to Early-Offer for SIP Audio Calls for a Dial-Peer, page 193

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** **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 sip to sip`	Allows connections between specific types of endpoints in an Cisco UBE. 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	**early-offer forced** **Example:** `Router(config-voi-serv)# early-offer forced`	Enables SIP Delayed-Offer to Early-Offer globally.
Step 6	**exit** **Example:** `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** **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	**voice-class sip early-offer forced** **Example:** `Router(config-dial-peer)# voice-class sip early-offer forced`	Forcefully send Early-Offer
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring Call Escalation from Voice to Video

The Call Escalation from Voice to Video feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. The call initially starts as an audio-only call. When the call is in progress, media renegotiation results in a video stream being added to the call, leading to call escalation from an audio-only call to an audio and video call.

To configure call escalation for SIP-to-SIP calls from voice calls to video, perform the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type* **to** *to-type*
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **session protocol sipv2**
8. **codec transparent**
9. **end**

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-srv)# allow-connections sip to sip`	Allows connections between SIP endpoints in a VoIP network.
Step 5	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits VoIP voice service configuration mode and returns to global configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	Enters dial-peer voice configuration mode for the specified VoIP dial peer.
Step 7	**session protocol sipv2** **Example:** `Router(config-dial-peer)# session protocol sipv2`	Enters the session protocol type as SIP.
Step 8	**codec transparent** **Example:** `Router(config-dial-peer)# codec transparent`	Specifies the voice codec rate of speech for a dial peer. • **transparent**—Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element (UBE). The **transparent** keyword is available only on the Cisco 2600, 3600, 7200, and 7500 series routers.
Step 9	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial-peer voice configuration mode.

Configuring SIP Error Message Pass Through

The SIP error message pass through feature allows a received error response from one SIP leg to pass transparently over to another SIP leg. This functionality will pass SIP error responses that are not yet supported on the Cisco UBE or will preserve the Q.850 cause code across two sip call-legs.

SIP error responses that are not supported on the Cisco UBE include: 300—Multiple choices, 301—Moved permanently, and 485—Ambiguous

Pre-leg SIP error responses that are not transparently passed though include:

Error code received	Corresponding error reported on the peer leg
400—Bad request	500—Internal error
401—Unauthorized	503—Service unavailable
406—Not acceptable	500—Internal error
407—Authentication required	503—Service unavailable
413—Request message body too large	500—Internal error
414—Request URI too large	500—Internal error
416—Unsupported URI scheme	500—Internal error
423—Interval too brief	500—Internal error
482—Loop detected	500—Internal error
483—Too many hops	500—Internal error
488—Not acceptable media (applicable only when the call is transcoded)	500—Internal error

Prerequisites

To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

- Configuring SIP error header passing in at the dial-peer level is not supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voice**
4. **sip**
5. **header-passing error-pass through**
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	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**header-passing error-pass through** **Example:** `Router(conf-serv-sip)#header-passing error-pass through`	Passes received error responses from one SIP leg to pass transparently to another SIP leg.
Step 6	**exit** **Example:** `Router(config-serv-sip) exit`	Exit SIP configuration mode.

Configuring Cisco UBE for Unsupported Content Pass-through

This feature introduces the ability to configure the Cisco UBE to pass through end to end headers at a global or dial-peer level, that are not processed or understood in a SIP trunk to SIP trunk scenario. The pass through functionality includes all or only a configured list of unsupported or non-mandatory SIP headers, and all unsupported content/MIME types.

The Cisco Unified Border Element does not support end-to-end media negotiation between the two endpoints that establish a call session through the Cisco Unified Border Element. This is a limitation when the endpoints intend to negotiate codec/payload types that the Cisco Unified Border Element does not process, because currently, unsupported payload types will never be negotiated by the Cisco Unified Border Element. Unsupported content types include text/plain, image/jpeg and application/resource-lists+xml. To address this problem, SDP is configured to pass through transparently at the Cisco Unified Border Element, so that both the remote ends can negotiate media independently of the Cisco Unified Border Element.

SDP pass-through is addressed in two modes:

- Flow-through: Cisco Unified Border Element plays no role in the media negotiation, it blindly terminates and re-originates the RTP packets irrespective of the content type negotiated by both the ends. This supports address hiding and NAT traversal.
- Flow-around: Cisco Unified Border Element neither plays a part in media negotiation, nor does it terminate and re-originate media. Media negotiation and media exchange is completely end-to-end.

Prerequisites for Cisco UBE for Unsupported Content Pass-through

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- Configuring the **media flow-around** command is required for SDP pass-through. When flow-around is not configured, the flow-through mode of SDP pass-through will be functional.
- When the dial-peer media flow mode is asymmetrically configured, the default behavior is to fallback to SDP pass-through with flow-through.

Restrictions for Cisco UBE for Unsupported Content Pass-through

When SDP pass-through is enabled, some of interworking that the Cisco Unified Border Element currently performs cannot be activated. These features include:

- Delayed Offer to Early Offer Interworking
- Supplementary Services with triggered Invites
- DTMF Interworking scenarios
- Fax Interworking/QoS Negotiation
- Transcoding

To enable Cisco UBE Unsupported Content Pass-through perform the steps in this section. This section contains the following subsections:

- Configuring Cisco UBE for Unsupported Content Pass-through at the Global Level, page 198
- Configuring Cisco UBE for Unsupported Content Pass-through at the Dial Peer Level, page 199

Configuring Cisco UBE for Unsupported Content Pass-through at the Global Level

To configure Unsupported Content Pass-through on an Cisco Unified Border Element at the global level, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **pass-thru** {**content** {**sdp** | **unsupp**} | **headers** {**unsupp** | *list tag*}}
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	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**pass-thru** {**content** {**sdp** | **unsupp**} | **headers** {**unsupp** | **list** *tag*}} **Example:** `Router(conf-serv-sip)# pass-thru {content {sdp | unsupp} | headers {unsupp | list <tag>}}`	Passes the SDP transparently from in-leg to the out-leg with no media negotiation.
Step 6	**exit** **Example:** `Router(conf-voi-serv)# exit`	Exits the current mode.

Configuring Cisco UBE for Unsupported Content Pass-through at the Dial Peer Level

To configure Unsupported Content Pass-through on an Cisco Unified Border Element at the dial-peer level, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **voice-class sip pass-thru**{{**headers** | **content**} {**content** {**unsupp** | **sdp**}}
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 22 voip`	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	**voice-class sip pass-thru**{{**headers** | **content**} {**content** {**unsupp** | **sdp**} **Example:** `Router (conf-dial-peer)# voice-class sip pass-thru headers`	`Passes the SDP transparently from in-leg to the out-leg with no media negotiation.`
Step 5	**exit** **Example:** `Router(conf-voi-serv)# exit`	Exits the current mode.

Configuring Cisco UBE for STUN and DTLS Pass-through

Cisco TelePresence System (CTS) endpoints send and receive Session Traversal Utilities for NAT (STUN) and Datagram Transport Layer Security (DTLS) packets. STUN packets are sent to open and refresh firewall pinholes. DTLS handshakes are performed to establish the Secure Real-Time Transport Protocol (SRTP) security parameters for secure CTS calls.

This feature enables Cisco Unified Border Element (Cisco UBE) to support STUN and DTLS packet pass-through, thereby adding support for secure CTS calls through Cisco UBE. However, the feature is generic and is supported for any endpoint that sends STUN or DTLS packets, including Trusted Relay Point (TRP).

Note The configuration for STUN and DTLS pass-through on Cisco UBE is enabled by default and requires no specific configuration.

However, to enable STUN and DTLS pass-through for Cisco TelePresence System (CTS) calls, perform the following tasks:

- Configuring RTCP Report Generation on Cisco UBE, page 271 (optional)
- Troubleshooting Tips, page 272 (optional)

Restrictions

- STUN and DTLS pass-through over IPv6 is not supported.
- DTLS pass-through is not supported for T.38 fax and modem relay calls.
- STUN and DTLS pass-through is not supported when Cisco UBE inserts a Digital Signal Processor (DSP) for transcoding interworkings such as SRTP-RTP, dual-tone multi-frequency (DTMF), and so on.

Configuring STUN and DTLS Pass-through for CTS Calls at the Global Level

Perform this task to configure STUN and DTLS pass-through for CTS calls at the global level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **rtp-ssrc multiplex**
5. **allow-connections** *from-type* **to** *to-type*
6. **sip**
7. **rel1xx disable**
8. **header-passing error-passthru**
9. **end**

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 voice service configuration mode.
Step 4	**rtp-ssrc multiplex** **Example:** `Router(conf-voi-serv)# rtp-ssrc multiplex`	Uses the global rtp-ssrc mux CLI setting.

	Command or Action	Purpose
Step 5	**allow-connections** *from-type* **to** *to-type* **Example:** `Router(conf-voi-serv)# allow-connections sip to sip`	Allows connections between SIP endpoints in a VoIP network.
Step 6	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service SIP configuration mode.
Step 7	**rel1xx disable** **Example:** `Router(conf-serv-sip)# rel1xx disable`	Disables the use of reliable provisional responses.
Step 8	**header-passing error-passthru** **Example:** `Router(conf-serv-sip)# header-passing error-passthru`	Enables SIP error response pass-through functionality.
Step 9	**end** **Example:** `Router(conf-serv-sip)# end`	Exits voice service SIP configuration mode and returns to privileged EXEC mode.

Configuring STUN and DTLS Pass-through for CTS Calls at the Dial Peer Level

Perform this task to configure STUN and DTLS pass-through for CTS calls at the dial-peer level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *voice-dial-peer-tag* **voip**
4. **destination-pattern** *E.164-standard-number*
5. **rtp payload-type cisco-codec-fax-ind** *number*
6. **rtp payload-type cisco-codec-aacld** *number*
7. **rtp payload-type cisco-codec-video-h264** *number*
8. **session protocol sipv2**
9. **session target ipv4:***destination-address*
10. **incoming called-number** *E.164-standard-number*
11. **playout-delay minimum low**
12. **codec transparent**
13. **no vad**
14. **end**

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** *voice-dial-peer-tag* **voip** **Example:** `Router(config)# dial-peer voice 234 voip`	Enters dial peer voice configuration mode.
Step 4	**destination-pattern** *E.164-standard-number* **Example:** `Router(config-dial-peer)# destination-pattern 3305550033`	Configures the specified E.164 telephone number to be used for a dial peer.
Step 5	**rtp payload-type cisco-codec-fax-ind** *number* **Example:** `Router(config-dial-peer)# rtp payload-type cisco-codec-fax-ind 110`	Identifies the payload type of an RTP packet. • **cisco-codec-fax-ind** *number*—Cisco codec fax indication. Range: 96 to 127. Default: 96.
Step 6	**rtp payload-type cisco-codec-aacld** *number* **Example:** `Router(config-dial-peer)# rtp payload-type cisco-codec-aacld 96`	Identifies the payload type value in the dynamic range. • **cisco-codec-aacld** *number*—Cisco codec AACLD. Range: 96 to 127. Default: 96.
Step 7	**rtp payload-type cisco-codec-video-h264** *number* **Example:** `Router(config-dial-peer)# rtp payload-type cisco-codec-video-h264 112`	Identifies the payload type of an RTP packet. • **cisco-codec-video-h264** *numbe*r—RTP video codec H.264 payload type. Range: 96 to 127. Default: 119.
Step 8	**session protocol sipv2** **Example:** `Router(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the packet network. • **sipv2**—Dial peer uses the Internet Engineering Task Force (IETF) Session Initiation Protocol (SIP).
Step 9	**session target ipv4:***destination-address* **Example:** `Router(config-dial-peer)# session target ipv4:192.168.1.2:5019`	Specifies a network-specific destination for a dial peer to receive calls from the current dial peer. • **ipv4:***destination-address*—IP address of the dial peer to receive calls. The colon is required.

	Command or Action	Purpose
Step 10	**incoming called-number** *E.164-standard-number* **Example:** `Router(config-dial-peer)# incoming called-number 7705550077`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.
Step 11	**playout-delay minimum low** **Example:** `Router(config-dial-peer)# playout-delay minimum low`	Tunes the playout buffer on digital signal processors (DSPs) to accommodate packet jitter caused by switches in the WAN. • **minimum**—Lower limit of the jitter buffer, or the lowest value to which the adaptive delay is set, in milliseconds. Value is as follows: • **low**—10 ms. Use when there are low jitter conditions in the network.
Step 12	**codec transparent** **Example:** `Router(config-dial-peer)# codec transparent`	Specifies the voice coder rate of speech for a dial peer. • **transparent**—Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element. **Note** The transparent keyword is available only on the Cisco 2600, 3600, 7200, and 7500 series routers.
Step 13	**no vad** **Example:** `Router(config-dial-peer)# no vad`	Disables voice activity detection (VAD) for the calls using a particular dial peer.
Step 14	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial peer voice configuration mode and returns to privileged EXEC mode.

Troubleshooting Tips

The following **debug** commands display the details about STUN and DTLS packets sent and received:

- **debug ccsip messages**—Shows SIP messages.

```
Router# debug ccsip messages

SIP Call messages tracing is enabled
```

- **debug ccsip error**—Shows SIP Service Provider Interface (SPI) errors.

```
Router# debug ccsip error

SIP Call error tracing is enabled
```

- **debug voip rtp session event**—Shows debugging related to RTP named event packets.

```
Router# debug voip rtp session event

*Nov 26 12:10:06.558: voip_rtp_is_media_service_pak: Received DTLS packet
              (src ip=192.16.2.2, src port=19312, dst ip=192.16.2.1, dst port=16386
               pdu size=111, pdu=0x16, switching ctx=interrupt)
```

```
*Nov 26 12:10:06.558: voip_rtp_send_service_pak: Sending DTLS packet
              (src ip=192.16.2.1, src port=17958, dst ip=192.16.2.3, dst port=5020
               pdu size=111, pdu=0x16, switching ctx=interrupt)
*Nov 26 12:10:07.014: voip_rtp_is_media_service_pak: Received STUN packet
              (src ip=192.16.2.2, src port=19210, dst ip=192.16.2.1, dst port=16910
               pdu size=20, pdu=0x00, switching ctx=interrupt)
*Nov 26 12:10:07.014: voip_rtp_send_service_pak: Sending STUN packet
              (src ip=192.16.2.1, src port=17894, dst ip=192.16.2.3, dst port=5022
               pdu size=20, pdu=0x00, switching ctx=interrupt)
```

Note The **debug voip rtp session event** command should be enabled only for troubleshooting purposes.

Configuring Media Flow-Around

This feature adds media flow-around capability on the Cisco Unified Border Element 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 dialpeer, 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 UBE 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 UBE. The Cisco UBE continues to handle routing and billing functions.

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, page 206
- Configuring Media Flow-Around at the Global Level, page 207
- Configuring Media Flow-Around for a Dial Peer, page 207
- Configuring Delayed-Offer to Early-Offer Media Flow-Around at the Global Level, page 208
- Configuring Delayed-Offer to Early-Offer Media Flow-Around for a Dial-Peer, page 210
- Configuring Delayed-Offer to Early-Offer Media Flow-Around for High-Density Transcoding Calls, page 211

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- 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 on a Cisco Unified Border Element" section on page 190 of this chapter.

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**
4. **media flow-around**
5. **dial-peer voice 2 voip**
6. **voice-class media 1**
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** *tag* **voip** **Example:** `Router(config-class)# dial-peer voice 2 voip`	Enters dial-peer configuration mode and assign an identification tag for VoIP.
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-dial-peer)# exit`	Exit dial-peer 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 voice 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 dial peer voice configuration and returns to privileged EXEC mode.

Configuring Delayed-Offer to Early-Offer Media Flow-Around at the Global Level

Perform this task to configure delayed-offer (DO) to early-offer (EO) media flow-around at the voice service configuration mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media flow-around**
5. **sip**
6. **early-offer forced**

7. **exit**
8. **exit**
9. **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 voice service configuration mode.
Step 4	**media flow-around** **Example:** `Router(config-voi-serv)# media flow-around`	Enables media flow-around.
Step 5	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 6	**early offer-forced** **Example:** `Router(config-serv-sip)# early offer-forced`	Forcefully sends SIP EO invites on the Out-Leg(OL).
Step 7	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode and returns to voice service configuration mode.
Step 8	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice service configuration mode and returns to global configuration mode.
Step 9	**exit** **Example:** `Router(config# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring Delayed-Offer to Early-Offer Media Flow-Around for a Dial-Peer

Perform this task to configure DO to EO Media Flow-Around for an individual dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **media flow-around**
5. **voice class sip early-offer forced**
6. **exit**
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	**dial-peer voice** *number* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	Enters dial peer voice 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	**voice class sip early-offer forced** **Example:** `Router(config-dial-peer)# voice class sip early-offer forced`	Forcefully sends SIP EO invites on the Out-Leg.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode and returns to global configuration mode.
Step 7	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring Delayed-Offer to Early-Offer Media Flow-Around for High-Density Transcoding Calls

Perform this task to configure Delayed-Offer to Early-Offer Media transcoding high-density calls.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media transcoder high-density**
5. **sip**
6. **early offer-forced**
7. **exit**
8. **exit**
9. **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 voice service configuration mode.
Step 4	**media transcoder high-density** **Example:** `Router(config-voi-serv)# media transcoder high-density`	Enables media transcoder high-density for transcoding high-density media calls.
Step 5	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 6	**early offer-forced** **Example:** `Router(config-serv-sip)# early offer-forced`	Forcefully sends SIP EO invites on the Out-Leg.

	Command or Action	Purpose
Step 7	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode and returns to voice service configuration mode.
Step 8	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice service configuration mode and returns to global configuration mode.
Step 9	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring Media Antitrombone

Media Trombones are media loops in a SIP entity due to call transfer or call forward. Media loops in Cisco UBE are not detected because Cisco UBE looks at both call types as individual calls and not calls related to each other.

Antitromboning is a media signaling service in SIP entity to overcome the media loops. Antitrombone service has to be enabled only when no media interworking is required in both the out-legs.

To specify media antitrombone for voice class, all VoIP calls, or individual dial peers, perform the tasks in the following sections:

- Configuring Media Antitrombone for a Voice Class, page 213 (Required)
- Configuring Media Antritrombone at the Global Level, page 214 (Required)
- Configuring Media Antitrombone for a Dial Peer, page 215 (Required)

Prerequisites

To enable this feature, you must have Cisco IOS Release 15.1(3)T or a later release installed on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

- When media antitrombone service is activated, Cisco UBE does not perform supplementary services such as handling REFER-based call transfers or media services such as SRTP, SNR and call transfers.
- Video codecs are not supported for the normal media handling because the SIP Cisco IOS gateway infrastructure does not support flow-through and flow-around for video.
- Antitrombone will not work if one call leg is flow-through and another call leg is flow-around. Similarly, antitrombone will not work if one call leg is SDP pass-through and another call leg is SDP normal.

- H.323 is not supported.
- Delayed-offer to early-offer (DO-EO) video media flow around is not supported.

Configuring Media Antitrombone for a Voice Class

Perform this task to configure antitrombone service for a voice class.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class media** *tag*
4. **media anti-trombone**
5. **exit**
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 class media** *tag* **Example:** `Router(config)# voice class media 1`	Enters voice class configuration mode and assigns an identification tag for a media voice class.
Step 4	**media anti-trombone** **Example:** `Router(config-class)# media anti-trombone`	Configures media antitrombone service.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer configuration mode and enters global configuration mode.
Step 6	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and enters privileged EXEC mode.

Configuring Media Antritrombone at the Global Level

Perform this task to configure media antitrombone service at the voice service configuration mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media anti-trombone**
5. **exit**
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 voice service configuration mode.
Step 4	**media anti-trombone** **Example:** `Router(config-voi-serv)# media anti-trombone`	Configures media antitrombone service.
Step 5	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice service configuration mode and returns to global configuration mode.
Step 6	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring Media Antitrombone for a Dial Peer

Perform this task to configure media antitrombone at individual dial peer level.

Restrictions

- If both incoming and outgoing dial peers are configured, you must specify the transparent codec on the incoming dial peer.
- The **media anti-trombone** command needs to be enabled for all related dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **media anti-trombone**
5. **exit**
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	**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 anti-trombone** **Example:** `Router(config-dial-peer)# media antri-trombone`	Configures media antitrombone service.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer configuration mode and enters global configuration mode.
Step 6	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and enters privileged EXEC mode.

Configuring DTMF Relay Digit-Drop on a Cisco Unified Border Element

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 the **dtmf-relay h245-alphanumeric** or **dtmf-relay h245-signal** command. This feature can 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).

To configure DTMF relay digit drop on an Cisco UBE with Cisco Unified Communications Manager, perform the steps in this section.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(4)T or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

- You should not configure digit-drop for inband to and from rtp-nte dtmf conversion (this involves transcoder), the digit-drop CLI prevents sending rtp-nte packets from the RTP lib.
- 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.
- Cisco IOS MTP should be used when the Cisco UBE does DTMF interworking between inband G.711 voice and RFC 2833 with Cisco Communication Manager (CCM) SIP trunk.

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.
Step 4	**dtmf-relay** [**cisco-rtp**] [**h245-alphanumeric**][**h245-signal**] [**rtp-nte** [**digit-drop**]] **Example:** `Router (config-dial-peer)# dtmf-relay rtp-nte digit-drop`	Forwards DTMF tones. Keywords are as follows: • **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. **Note** The **digit-drop** keyword is available only when the **rtp-nte** keyword is configured.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

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 Unified BE:

```
.
.
.
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 Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls Feature

The Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls feature provides dynamic payload type interworking for dual tone multifrequency (DTMF) and codec packets for Session Initiation Protocol (SIP) to SIP calls.

Based on this feature, the Cisco Unified Border Element interworks between different dynamic payload type values across the call legs for the same codec. Also, Cisco UBE supports any payload type value for audio, video, named signaling events (NSEs), and named telephone events (NTEs) in the dynamic payload type range 96 to 127.

Symmetric and Asymmetric Calls

Cisco UBE supports dynamic payload type negotiation and interworking for all symmetric and asymmetric payload type combinations. A call leg on Cisco UBE is considered as symmetric or asymmetric based on the payload type value exchanged during offer answer with the endpoint:

- A symmetric endpoint accepts and sends the same payload type.
- An asymmetric endpoint can accept and send different payload types.

Default Behavior

The Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature is enabled by default for a symmetric call. An offer is sent with a payload type based on the dial-peer configuration. The answer is sent with the same payload type as was received in the incoming offer. When the payload type values negotiated during the signaling are different, the Cisco UBE changes the Real-Time Transport Protocol (RTP) payload value in the VoIP to RTP media path.

CLI Behavior

The Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature is not enabled by default for an asymmetric call leg. You must use the **asymmetric payload** command to configure this feature to support asymmetric call legs. The dynamic payload type value is passed across the call legs, and the RTP payload type interworking is not required. The RTP payload type handling is dependent on the endpoint receiving them.

Prerequisites

To enable this feature, you must have Cisco IOS Release 15.0(1)XA or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

The Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature is not supported for the following:

- H323-to-H323 and H323-to-SIP calls.
- All transcoded calls.
- Secure Real-Time Protocol (SRTP) pass-through calls.
- Flow-around calls.
- Asymmetric payload types are not supported on early-offer (EO) call leg in a delayed-offer to early-offer (DO-EO) scenario.
- Multiple *m* lines with the same dynamic payload types, where *m* is:

  m = audio <media-port1> RTP/AVP XXX
  m = video <media-port2> RTP/AVP XXX

Configuring Dynamic Payload Support at the Global Level

Perform this task to configure the Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature at the global level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **asymmetric payload** {**dtmf** | **dynamic-codecs** | **full** | **system**}
6. **end**

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 voice service configuration mode.

	Command or Action	Purpose
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service SIP configuration mode.
Step 5	**asymmetric payload** {**dtmf** | **dynamic-codecs** | **full** | **system**} **Example:** `Router(conf-serv-sip)# asymmetric payload full`	Configures global SIP asymmetric payload support. **Note** The **dtmf** and **dynamic-codecs** keywords are internally mapped to the **full** keyword to provide asymmetric payload type support for audio and video codecs, DTMF, and NSEs.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Exits voice service SIP configuration mode and enters privileged EXEC mode.

Configuring Dynamic Payload Support for a Dial Peer

Perform this task to configure Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature for a dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip asymmetric payload** {**dtmf** | **dynamic-codecs** | **full** | **system**}
5. **end**

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 77 voip`	Enters dial peer voice configuration mode.

	Command or Action	Purpose
Step 4	**voice-class sip asymmetric payload** {**dtmf** | **dynamic-codecs** | **full** | **system**} **Example:** `Router(config-dial-peer)# voice-class sip asymmetric payload full`	Configures the dynamic SIP asymmetric payload support feature. **Note** The **dtmf** and **dynamic-codecs** keywords are internally mapped to the **full** keyword to provide asymmetric payload type support for audio and video codecs, DTMF, and NSEs.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	(Optional) Exits dial peer voice configuration mode and enters privileged EXEC mode.

Troubleshooting the Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls Feature

Use the following commands to debug any errors that you may encounter when you configure the Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature.

- **debug ccsip all**
- **debug voip ccapi inout**
- **debug voip rtp**

Verifying Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls Feature

This task shows how to display information to verify Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls configuration. These **show** commands need not be entered in any specific order.

SUMMARY STEPS

1. **enable**
2. **show call active voice compact**
3. **show call active voice**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**show call active voice compact** **Example:** `Router# show call active voice compact`	(Optional) Displays a compact version of call information.

	Command or Action	Purpose

Step 3	`show call active voice`
	
	Example:
	`Router# show call active voice`
		(Optional) Displays call information for voice calls in progress.

Enabling In-Dialog OPTIONS to Monitor Active SIP Sessions

The two common methods to determine whether a SIP session is active; RTP/RTCP media inactivity timer and session timer have limitations when used with the Cisco UBE. The media inactivity (rtp/rtcp) method will not work if flow around mode is configured as the media is sent directly between endpoints without going through the Cisco UBE and session timer cannot be used if the SIP endpoint does not support session timer.

The in-dialog OPTIONS refresh feature introduces a refresh mechanism that addresses these two scenarios, and can be used on SIP-to-SIP and SIP-to-H.323 calls. The refresh with OPTIONS method is meant to only be hop-to-hop, and not end-to-end. Since session timer achieves similar results, the OPTIONs refresh/ping will not take affect when session timer is negotiated. The behavior on the H.323 endpoint is as if it was a TDM-SIP call. The generating in-dialog OPTIONS is enabled at the global level or dialpeer level. The system default setting is disabled. This feature can be use by both a TDM voice gateway and an Cisco UBE.

To enable in-dialog OPTIONS at the global level, or individual dial peers, perform the steps in this section. This section contains the following subsections:

Methods to Determine Active SIP Sessions

RTP/RTCP

The SIP Media Inactivity Timer enables Cisco gateways to monitor and disconnect VoIP calls if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period.

Session Timer

The SIP Session Timer periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests are sent during an active call leg to allow user agents (UA) or proxies to determine the status of a SIP session. The re-INVITES ensure that active sessions stay active and completed sessions are terminated.

Prerequisites

To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Enabling In-dialog OPTIONS at the Global Level

To enable in-dialog OPTIONS at the global level, perform the steps in this section.

Note The global system default setting is disable.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **options-ping 90**
6. **exit**
7. **end**

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	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**options-ping** **Example:** `Router(conf-serv-sip)# options-ping 90`	Enables in-dialog OPTIONS. OPTIONS transactions are sent, in seconds.

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.
Step 7	**end** **Example:** `Router(config-voi-srv)# end`	Returns to privileged EXEC mode.

Enabling in-dialog OPTIONS for a Dial-Peer

To enable in-dialog OPTIONS for an individual dial-peer, perform the steps in this section.

Restrictions

When configuring in-dialog OPTIONS at the dial-peer level OPTIONS must be configured on both incoming and outgoing dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice 1 voip**
4. **voice-class sip options-ping**
5. **exit**
6. **end**

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.

	Command or Action	Purpose
Step 4	**voice-class sip options-ping** **Example:** `Router(config-voip-peer)# voice-class sip options-ping 65`	Enables intervals OPTIONS transactions to be sent, in seconds.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.
Step 6	**end** **Example:** `Router(config-voi-srv)# end`	Returns to privileged EXEC mode.

Configuring Library Based RTCP Media Inactivity Timer

Restrictions

- No dsp based media inactivity is supported in CUBE

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip rtcp report interval**
4. **gateway**
5. **media-inactivity-criteria rtcp**
6. **timer receive-rtcp 5**
7. **exit**
8. **end**

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	`ip rtcp report interval value`
	
	Example:
	`Router(config)#ip rtcp report interval 9000`
		Set the average reporting interval between subsequent Real-Time Control Protocol (RTCP) report transmissions.
		• *value*—Average interval for RTCP report transmissions, in ms. Range is 1 to 65535. Default is 5000.

Step 4	**gateway** **Example:** `Router(config)# gateway`	Enable the H.323 VoIP gateway.

Step 5	`media-inactivity-criteria rtcp`
	
	Example:
	`Router (config-gateway)#media-inactivity-criteria rtcp`
		Specifies the mechanism for detecting media inactivity (silence) on a voice call.

Step 6	`timer receive-rtcp timer`
	
	Example:
	`Router (config-gateway)#timer receive-rtcp 5`
		Enables the RTCP timer and to configures a multiplication factor for the RTCP timer interval
		• timer—Multiples of the RTCP report transmission interval. Range is 0 to 1000. Default is 0. Recommended value is 5.

Step 7	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.
Step 8	**end** **Example:** `Router(config-voi-srv)# end`	Returns to privileged EXEC mode.

Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints

The Out-of-dialog (OOD) Options Ping feature provides a keepalive mechanism at the SIP level between any number of destinations. A generic heartbeat mechanism allows Cisco Unified Border Element to monitor the status of SIP servers or endpoints and provide the option of busying-out a dial-peer upon total heartbeat failure. When a monitored endpoint heartbeat fails, the dial-peer is busied out. If an alternate dial-peer is configured for the same destination pattern, the call is failed over to the next preferred dial peer, or else the on call is rejected with an error cause code.

The response to options ping will be considered unsuccessful and dial-peer will be busied out for following scenarios:

Table 2 Error Codes that busyout the endpoint

Error Code	Description
503	service unavailable
505	sip version not supported
no response	i.e. request timeout

All other error codes, including 400 are considered a valid response and the dial peer is not busied out.

Note The purpose of this feature is to determine if the SIP session protocol on the endpoint is UP and available to handle calls. It may not handle OPTIONS message but as long as the SIP protocol is available, it should be able to handle calls.

When a dial-peer is busied out, Cisco Unified Border Element continues the heartbeat mechanism and the dial-peer is set to active upon receipt of a response.

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- The following are required for OOD Options ping to function. If any are missing, the Out-of-dialog (OOD) Options ping will not be sent and the dial peer is reset to the default active state.
  - Dial-peer should be in active state
  - Session protocol must be configured for SIP
  - Configure Session target or outbound proxy must be configured. If both are configured, outbound proxy has preference over session target.

Restrictions

- The Cisco Unified Border Element OOD Options ping feature can only be configured at the VoIP Dial-peer level.
- All dial peers start in an active (not busied out) state on a router boot or reboot.
- If a dial-peer has both an outbound proxy and a session target configured, the OOD options ping is sent to the outbound proxy address first.
- Though multiple dial-peers may point to the same SIP server IP address, an independent OOD options ping is sent for each dial-peer.
- If a SIP server is configured as a DNS hostname, OOD Options pings are sent to all the returned addresses until a response is received.
- Configuration for Cisco Unified Border Element OOD and TDM Gateway OOD are different, but can co-exist.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip options-keepalive**
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 200 voip`	Enters dial-peer configuration mode for the VoIP peer designated by tag.
Step 4	**voice-class sip options-keepalive** {**up-interval** *seconds* | **down-interval** *seconds* | **retry** *retries*} **Example:** `Router(config-dial-peer)# voice-class sip options-keepalive up-interval 12 down-interval 65 retry 3`	Monitors connectivity between endpoints. • **up-interval seconds** — Number of up-interval seconds allowed to pass before marking the UA as unavailable.The range is 5-1200. The default is 60. • **down-interval seconds** — Number of down-interval seconds allowed to pass before marking the UA as unavailable.The range is 5-1200. The default is 30. • **retry retries** — Number of retry attempts before marking the UA as unavailable. The range is 1 to 10. The default is 5 attempts.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Troubleshooting Tips

The following commands can help troubleshoot the OOD Options Ping feature:

- **debug ccsip all**—shows all Session Initiation Protocol (SIP)-related debugging.
- **show dial-peer voice x**—shows configuration of keepalive information.

```
Router# show dial-peer voice | in options
voice class sip options-keepalive up-interval 60 down-interval 30 retry 5
voice class sip options-keepalive dial-peer action  = active
```

- **show dial-peer voice summary**—shows Active or Busyout dial-peer status.

```
Router# show dial-peer voice summary

           AD                      PRE PASS
TAG TYPE  MIN  OPER PREFIX    DEST-PATTERN KEEPALIVE

111 voip  up     up               0 syst   active
9   voip  up    down              0 syst   busy-out
```

Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure

Cisco Unified Border Element (Cisco UBE) provides an option to configure the error response code when a dial peer is busied out because of an Out-of-Dialog OPTIONS ping failure.

The OPTIONS ping mechanism monitors the status of a remote Session Initiation Protocol (SIP) server, proxy or endpoints. Cisco UBE monitors these endpoints periodically. When there is no response from these monitored endpoints, the configured dial peer is busied out. If the dial-peer endpoint is busied out due to an OPTIONS ping failure, the call is passed on to the next dial-peer endpoint if an alternate dial peer is configured for the same destination. Otherwise the error response 404 is sent. This feature provides the option of configuring the error response code to reroute the call. Therefore when a dial peer is busied out due to the OPTIONS ping failure, the SIP error code configured in the inbound dial-peer is sent as a response.

To configure the SIP error code response, perform the following tasks:

- "Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure at the Global Level" section on page 229 (required)
- Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure at the Dial Peer Level, page 231 (required)

Prerequisites

- To enable this feature, you must have Cisco IOS Release 15.0(1)XA or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- The Cisco UBE Out-of-Dialog (OOD) OPTIONS Ping for Specified SIP Servers or Endpoints feature should be configured before configuring this error response code for a ping OPTIONS failure.

Restrictions

The error code configuration will not have any effect if it is configured on the outbound dial peer.

Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure at the Global Level

Table 3 describes the SIP error codes.

Table 3 SIP Error Codes

Error Code Number	Description
400	Bad Request
401	Unauthorized
402	Payment Required
403	Forbidden
404	Not Found
408	Request Timed Out
416	Unsupported URI
480	Temporarily Unavailable
482	Loop Detected
484	Address Incomplete
486	Busy Here
487	Request Terminated
488	Not Acceptable Here
500–599	SIP 5xx—Server/Service Failure
500	Internal Server Error
502	Bad Gateway
503	Service Unavailable
600–699	SIP 6xx—Global Failure

To configure the error response code for the OPTIONS ping failure to support the Cisco Unified Border Element at the global level, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **error-code-override options-keepalive failure** *sip-status-code-number*
6. **end**

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 voice service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service SIP configuration mode.
Step 5	**error-code-override options-keepalive failure** *sip-status-code-number* **Example:** `Router(conf-serv-sip)# error-code-override options-keepalive failure 402`	Configures the specified SIP error code number. • *sip-status-code-number*—SIP status code to be sent for an options keepalive failure. Range: 400 to 699. Default: 503. • Table 3 provides more details about these error codes.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Exits voice service SIP configuration mode and returns to privileged EXEC mode.

Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure at the Dial Peer Level

To configure the error response code for the OPTIONS ping failure to support the Cisco Unified Border Element at the dial-peer level, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *voice-dial-peer-tag* **voip**
4. **voice-class sip error-code-override options-keepalive failure** {*sip-status-code-number* | **system**}
5. **end**

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** *voice-dial-peer-tag* **voip** **Example:** `Router(config)# dial-peer voice 234 voip`	Enters dial peer voice configuration mode.
Step 4	**voice-class sip error-code-error-override options-keepalive failure** {*sip-status-code-number* | **system**} **Example:** `Router(config-dial-peer)# voice-class sip error-code-override options-keepalive failure 500`	Configures the specified SIP error code number. • *sip-status-code-number* —SIP status code to be sent for an options keepalive failure. Range: 400 to 699. Default: 503. • Table 3 provides more details about these error codes. **Note** If the **system** keyword is configured, the global level configuration will override the dial-peer configuration.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial peer voice configuration mode and returns to privileged EXEC mode.

Troubleshooting Tips

The following debug commands display any error that occurs with the error code response:

- **debug ccsip messages—**shows SIP messages.

```
Router# debug ccsip messages

SIP Call messages tracing is enabled
```

- **debug ccsip all**—shows all SIP-related debugging.

```
Router# debug ccsip all

This may severely impact system performance. Continue? [confirm]
All SIP Call tracing is enabled
```

Configuring SIP Parameters

The SIP Parameters feature allow customers to add, remove, or modify the SIP parameters in the SIP messages going out of a border element. The SIP message is generated from the standard signaling stack, but runs the message through a parser which can add, delete or modify specific parameters. This allows interoperability with additional third party devices that require specific SIP message formats. All SIP methods and responses are supported, profiles can be added either in dial-peer level or global level. Basic Regular Expression support would be provided for modification of header values. SDP parameters can also be added, removed or modified.

This feature is applicable only for outgoing SIP messages. Changes to the messages are applied just before they are sent out, and the SIP SPI code does not remember the changes. Because there are no restrictions on the changes that can be applied, users must be careful when configuring this feature – for example, the call might fail if a regular expression to change the To tag value is configured.

The **all** keyword is used to apply rules on all requests and responses.

Prerequisites

To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

Restrictions

- This feature applies to outgoing SIP messages.
- This feature is disabled by default.
- Removal of mandatory headers is not supported.
- This feature allows removal of entire MIME bodies from SIP messages. Addition of MIME bodies is not supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **voice-class sip profiles** *group-number*
5. **response** *option* **sip-header** *option* **ADD** *word* **CR**
6. **exit**
7. **end**

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** *number* **voip** **Example:** `Router(config)# voice service 1 voip`	Enters VoIP voice-service configuration mode.
Step 4	**voice-class sip-profiles** *group-number* **Example:** `Router(config)# voice-class sip profiles 42`	Establishes individual sip profiles defined by a group-number. Valid group-numbers are from 1 to 1000.
Step 5	**response** *option* **sip-header** *option* ADD *word* CR **Example:** `Router(config)# request INVITE sip-header supported remove`	Add, change, or delete any SIP or SDP header in voice class or sip-profile submode.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.
Step 7	**end** **Example:** `Router(config-voi-srv)# end`	Returns to privileged EXEC mode.

Example

```
!
!
!
voice service voip
allow-connections sip to sip
redirect ip2ip
sip
early-offer forced
midcall-signaling passthru
sip-profiles 1
!
!
!
voice class sip-profiles 1
request INVITE sip-header Supported remove
```

```
request INVITE sip-header Min-SE remove
request INVITE sip-header Session-Expires remove
request INVITE sip-header Unsupported modify "Unsupported:" "timer"
!
!
!
```

Configurable SIP Parameters via DHCP

The Configurable SIP Parameters via DHCP feature allows a Dynamic Host Configuration Protocol (DHCP) server to provide Session Initiation Protocol (SIP) parameters via a DHCP client. These parameters are used for user registration and call routing.

The DHCP server returns the SIP Parameters via DHCP options 120 and 125. These options are used to specify the SIP user registration and call routing information. The SIP parameters returned are the SIP server address via Option 120, and vendor-specific information such as the pilot, contract or primary number, an additional range of secondary numbers, and the SIP domain name via Option 125.

In the event of changes to the SIP parameter values, this feature also allows a DHCP message called DHCPFORCERENEW to reset or apply a new set of values.

The SIP parameters provisioned by DHCP are stored, so that on reboot they can be reused.

Prerequisites for Configurable SIP Parameters via DHCP

- To enable this feature, you must have Cisco IOS Release12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- A DHCP interface has to be associated with SIP before configurable SIP parameters via DHCP can be enabled.

Restrictions for Configurable SIP Parameters via DHCP

- DHCP Option 120 is the standard DHCP option (RFC3361) to get a SIP server address, and this can be used by any vendor DHCP server. Only one address is supported, which is in the IPv4 address format. Multiple IPv4 address entries are not supported. Also, there is no support for a DNS name in this or for any port number given behind the IPv4 address.
- DHCP Option 125 (RFC 3925) provides vendor-specific information and its interpretation is associated with the enterprise identity. The primary and secondary phone numbers and domain are obtained using Option 125, which is vendor-specific. As long as other customers use the same format as in the Next Generation Network (NGN) DHCP specification, they can use this feature.
- A primary or contract number is required in suboption 202 of DHCP Option 125. There can be only one instance of the primary number and not multiple instances.
- Multiple secondary or numbers in suboption 203 of DHCP Option 125 are supported. Up to five numbers are accepted and the rest ignored. Also, they have to follow the contract number in the DHCP packet data.
- Authentication is not supported for REGISTER and INVITE messages sent from a Cisco Unified Border Element that uses DHCP provisioning
- The DHCP provisioning of SIP Parameters is supported only over one DHCP interface.

- The DHCP option is available only to be configured for the primary registrar. It will not be available for a secondary registrar.

Information About Configurable SIP Parameters via DHCP

To perform basic Configurable SIP Parameters via DHCP configuration tasks, you should understand the following concepts:

- Cisco Unified Border Element Support for Configurable SIP Parameters via DHCP, page 236
- DHCP to Provision SIP Server, Domain Name, and Phone Number, page 236
- DHCP-SIP Call Flow, page 237
- DHCP Message Details, page 238

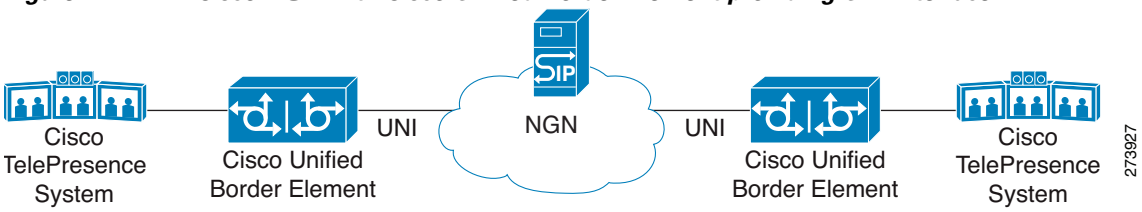
Cisco Unified Border Element Support for Configurable SIP Parameters via DHCP

The Cisco Unified Border Element provides the support for the DHCP provisioning of the SIP parameters.

The NGN is modeled using SIP as a VoIP protocol. In order to connect to NGN, the User to Network Interface (UNI) specification is used. Cisco TelePresence Systems (CTS), consisting of an IP Phone, a codec, and Cisco Unified Communications Manager, are required to internetwork over the NGN for point-to-point and point-to-multipoint video calls. Because Cisco Unified Communications Manager does not provide a UNI interface, there has to be an entity to provide the UNI interface. The Cisco Unified Border Element provides the UNI interface and has several advantages such as demarcation, delayed offer to early offer, and registration.

Figure 1 shows the Cisco Unified Border Element providing the UNI interface for the NGN.

Figure 1 Cisco NGN with Cisco Unified Border Element providing UNI interface



DHCP to Provision SIP Server, Domain Name, and Phone Number

NGN requires Cisco Unified Border Element to support DHCP (RFC 2131 and RFC 2132) to provision the following:

- IP address for Cisco Unified Border Element's UNI interface facing NGN
- SIP server address using option 120
- Option 125 vendor specific information to get:
  - Pilot number (also called primary or contract number), there is only one pilot number in DHCPACK, and REGISTER is done only for the pilot number
  - Additional numbers, or secondary numbers, are in DHCPACK; there is no REGISTER for additional numbers
  - SIP domain name

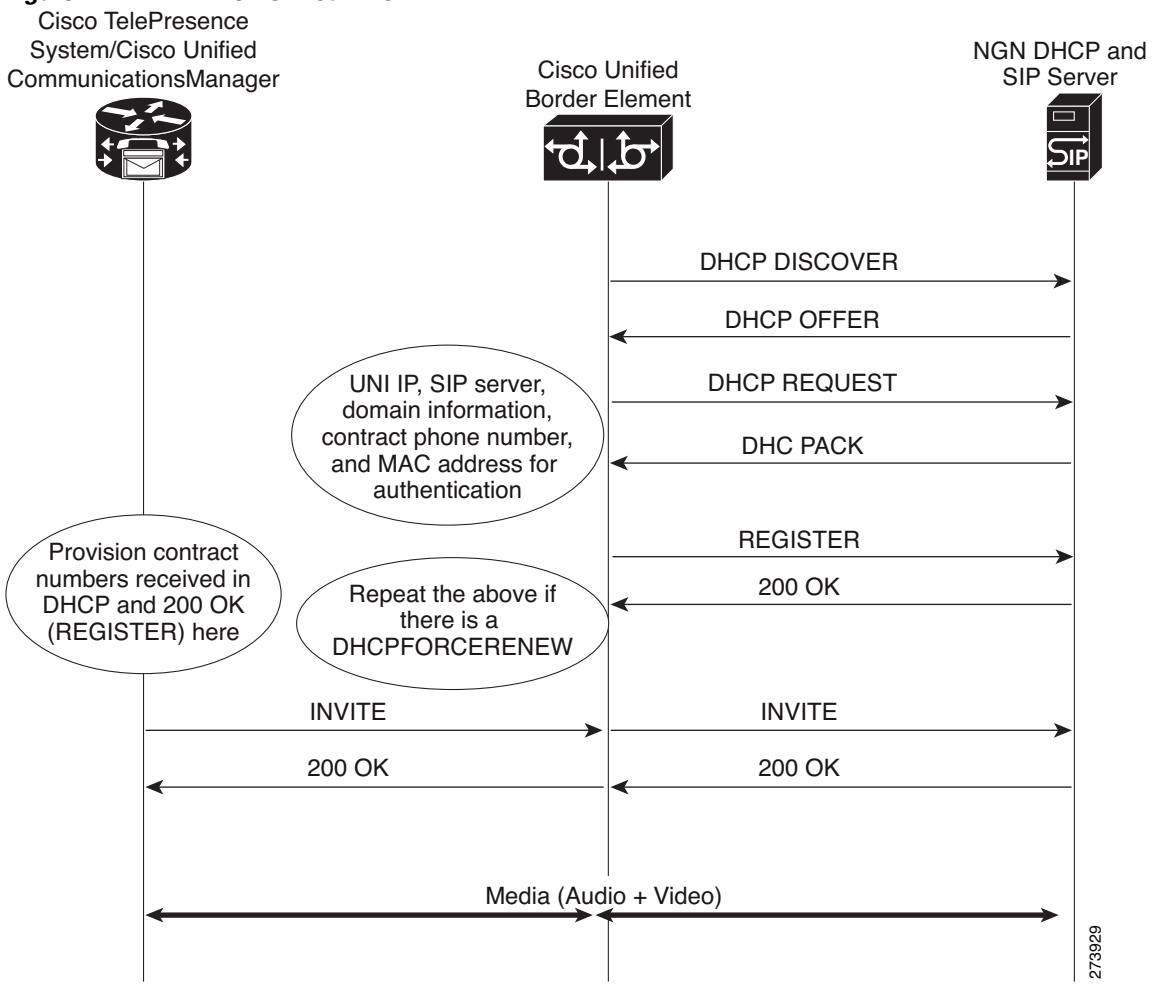
- DHCPFORCERENEW to reset or apply a new set of SIP parameters (RFC 3203)

DHCP-SIP Call Flow

The following scenario shows the DHCP messages involved in provisioning information such as the IP address for UNI interface, and SIP parameters including the SIP server address, phone number, and domain name, along with how SIP messages use the provisioned information.

Figure 2 shows the DHCP and SIP messages involved in obtaining the SIP parameters and using them for REGISTER and INVITE.

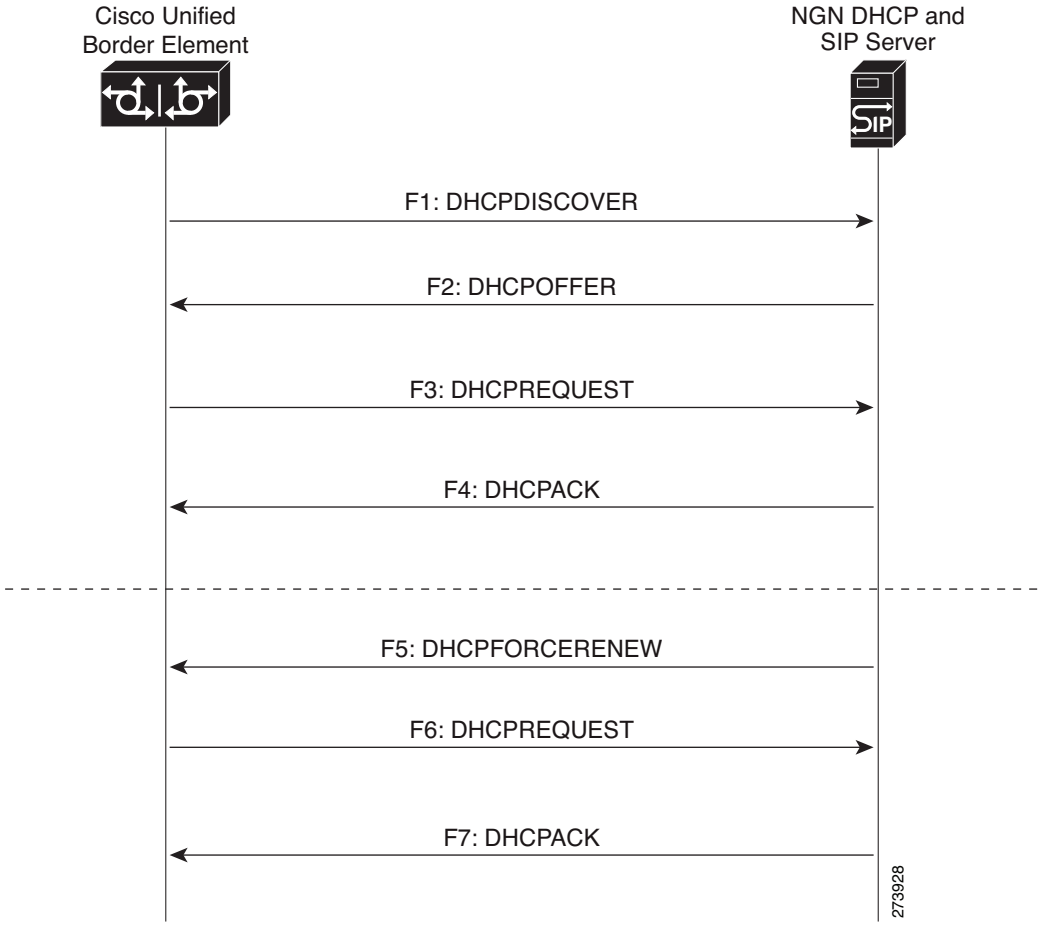
Figure 2 DHCP-SIP Call Flow



DHCP Message Details

The DHCP call flow involved in obtaining Cisco Unified Border Element provision information, including the IP address for UNI interface and SIP information such as phone number, domain, and SIP server, is shown in Figure 2.

Figure 3 DHCP Message Details



The DHCP messages involved in provisioning the SIP parameters are described in Steps 1 to 6.

1. F1: The Cisco Unified Border Element DHCP client sends a DHCPDISCOVER message to find the available NGN DHCP servers on the network and obtain a valid IPv4 address. The Cisco Unified Border Element DHCP client identity (computer name) and MAC address are included in this message.
2. F2: The Cisco Unified Border Element DHCP client receives a DHCPOFFER message from each available NGN DHCP server. The DHCPOFFER message includes the offered DHCP server's IPv4 address, the DHCP client's MAC address, and other configuration parameters.
3. F3: The Cisco Unified Border Element DHCP client selects an NGN DHCP server and its IPv4 address configuration from the DHCPOFFER messages it receives, and sends a DHCPREQUEST message requesting its usage. Note that this is where Cisco Unified Border Element requests SIP server information via DHCP Option 120 and vendor- identifying information via DHCP Option 125.

4. F4: The chosen NGN DHCP server assigns its IPv4 address configuration to the Cisco Unified Border Element DHCP client by sending a DHCPACK message to it. The Cisco Unified Border Element DHCP client receives the DHCPACK message. This is where the SIP server address, phone number and domain name information are received via DHCP options 120 and 125. The Cisco Unified Border Element will use the information for registering the phone number and routing INVITE messages to the given SIP server.
5. F5: When NGN has a change of information or additional information (such as changing SIP server address from 1.1.1.1 to 2.2.2.2) for assigning to Cisco Unified Border Element, the DHCP server initiates DHCPFORCERENEW to the Cisco Unified Border Element. If the authentication is successful, the Cisco Unified Border Element DHCP client accepts the DHCPFORCERENEW and moves to the next stage of sending DHCPREQUEST. Otherwise DHCPFORCERENEW is ignored and the current information is retained and used.
6. F6 and F7: In response to DHCPFORCERENEW, similar to steps F3 and F4, the Cisco Unified Border Element requests DHCP Options 120 and 125. Upon getting the response, SIP will apply these parameters if they are different by sending an UN-REGISTER message for the previous phone number and a REGISTER message for the new number. Similarly, a new domain and SIP server address will be used. If the returned information is the same as the current set, it is ignored and hence registration and call routing remains the same.

How to Configure SIP Parameters via DHCP

To configure SIP parameters via DHCP, perform the following tasks:

- Configuring the DHCP Client, page 239 (Required)
- Enabling the SIP Configuration, page 241 (Required)
- Configuring a SIP Outbound Proxy Server, page 242 (Required)
- Enabling Forced Update of SIP Parameters via DHCP, page 245 (Required)

Configuring the DHCP Client

To receive the SIP configuration parameters the Cisco Unified Border Element has to act as a DHCP client. This is because in the NGN network, a DHCP server pushes the configuration to a DHCP client. Thus the Cisco Unified Border Element must be configured as a DHCP client.

Perform this task to configure the DHCP client.

Prerequisites

You must configure the **ip dhcp client** commands before entering the **ip address dhcp** command on an interface to ensure that the DHCPDISCOVER messages that are generated contain the correct option values. The **ip dhcp client** commands are checked only when an IP address is acquired from DHCP. If any of the **ip dhcp client** commands are entered after an IP address has been acquired from DHCP, the DHCPDISCOVER messages' correct options will not be present or take effect until the next time the router acquires an IP address from DHCP. This means that the new configuration will only take effect after either the **ip address dhcp** command or the **release dhcp** and **renew dhcp** EXEC commands have been configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** ***type number***
4. **ip address dhcp**
5. **ip dhcp client request sip-server-address**
6. **ip dhcp client request vendor-identifying-specific**
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	**interface** ***type number*** **Example:** `Router(config)# interface gigabitethernet 0/0`	Configures an interface type and enters interface configuration mode.
Step 4	**ip dhcp client request sip-server-address** **Example:** `Router(config-if)# ip dhcp client request sip-server-address`	Configures the DHCP client to request a SIP server address from a DHCP server.
Step 5	**ip dhcp client request vendor-identifying-specific** **Example:** `Router(config-if)# ip dhcp client request vendor-identifying-specific`	Configures the DHCP client to request vendor-specific information from a DHCP server.
Step 6	**ip address dhcp** **Example:** `Router(config-if)# ip address dhcp`	Acquires an IP address on the interface from the DHCP.
Step 7	**exit** **Example:** `Router(config-if)# exit`	Exits the current mode.

Enabling the SIP Configuration

Enabling the SIP configuration allows the Cisco Unified Border Element to use the SIP parameters received via DHCP for user registration and call routing.

Perform this task to enable the SIP configuration.

Prerequisites

The **dhcp interface** command has to be entered to declare the interface before the **registrar** and **credential** commands are entered.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** *type number*
4. **sip-ua**
5. **dhcp interface** *type number*
6. **registrar dhcp expires** *seconds* **random-contact refresh-ratio** *seconds*
7. **credentials dhcp password** [**0** | **7**] *password* **realm** *domain-name*
8. **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	**interface** ***type number*** **Example:** `Router(config)# interface gigabitethernet 0/0`	Configures an interface type and enters interface configuration mode.
Step 4	**sip-ua** **Example:** `Router(config-if)# sip-ua`	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
Step 5	**dhcp interface** *type number* **Example:** `Router(sip-ua)# dhcp interface gigabitethernet 0/0`	Assigns a specific interface for DHCP provisioning of SIP parameters. • Multiple interfaces on the CUBE can be configured with DHCP—this command specifies the DHCP interface used with SIP.
Step 6	**registrar dhcp expires** *seconds* **random-contact refresh-ratio** *seconds* **Example:** `Router(sip-ua)# registrar dhcp expires 100 random-contact refresh-ratio 90`	Registers E.164 numbers on behalf of analog telephone voice ports (FXS) and IP phone virtual voice ports (EFXS) with an external SIP proxy or SIP registrar server. • **expires** *seconds*—Specifies the default registration time, in seconds. Range is 60 to 65535. Default is 3600. • **refresh-ratio** *seconds*—Specifies the refresh-ratio, in seconds. Range is 1 to 100 seconds. Default is 80.
Step 7	**credentials dhcp password** [**0** | **7**] *password* **realm** *domain-name* **Example:** `Router(sip-ua)# credentials dhcp password cisco realm cisco.com`	Sends a SIP registration message from a Cisco Unified Border Element in the UP state.
Step 8	**exit** **Example:** `Router(sip-ua)# exit`	Exits the current mode.

Troubleshooting Tips

To display information on DHCP and SIP interaction when SIP parameters are provisioned by DHCP, use the **debug ccsip dhcp** command in privileged EXEC mode.

Configuring a SIP Outbound Proxy Server

An outbound-proxy configuration sets the Layer 3 address (IP address) for any outbound REGISTER and INVITE SIP messages. The SIP server can be configured as an outbound proxy server in voice service SIP configuration mode or dial peer configuration mode. When enabled in voice service SIP configuration mode, all the REGISTER and INVITE messages are forwarded to the configured outbound proxy server. When enabled in dial-peer configuration mode, only the messages hitting the defined dial-peer will be forwarded to the configured outbound proxy server.

The configuration tasks in each mode are presented in the following sections:

- Configuring a SIP Outbound Proxy Server in Voice Service VoIP Configuration Mode, page 243
- Configuring a SIP Outbound Proxy Server and Session Target in Dial Peer Configuration Mode, page 244

Perform either of these tasks to configure the SIP server as a SIP outbound proxy server.

Configuring a SIP Outbound Proxy Server in Voice Service VoIP Configuration Mode

Perform this task to configure the SIP server as a SIP outbound proxy server in voice service SIP configuration mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **outbound-proxy dhcp**
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 voice service VoIP configuration mode and specifies VoIP as the voice-encapsulation type.
Step 4	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters voice service SIP configuration mode.
Step 5	**outbound-proxy dhcp** **Example:** `Router(conf-serv-sip)# outbound-proxy dhcp`	Configures the DHCP client to request a SIP server address from a DHCP server.
Step 6	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits the current mode.

Configuring a SIP Outbound Proxy Server and Session Target in Dial Peer Configuration Mode

Perform this task to configure the SIP server as a SIP outbound proxy server in dial peer configuration mode.

Restrictions

SIP must be configured on the dial pier before DHCP is configured. Therefore the **session protocol sipv2** command must be executed before the **session target dhcp** command. DHCP is supported only with SIP configured on the dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *number* voip**
4. **session protocol sipv2**
5. **voice-class sip outbound-proxy dhcp**
6. **session target dhcp**
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	**dial-peer voice *number* voip** **Example:** `Router(config)# dial-peer voice 10 voip`	Defines a dial peer, specifies VoIP as the method of voice encapsulation, and enters dial peer configuration mode.
Step 4	**session protocol sipv2** **Example:** `Router(config-dial-peer)# session protocol sipv2`	Enters the session protocol type as SIP.
Step 5	**voice-class sip outbound-proxy dhcp** **Example:** `Router(config-dial-peer)# voice-class sip outbound-proxy dhcp`	Configures the SIP server received from the DHCP server as a SIP outbound proxy server.

	Command or Action	Purpose
Step 6	**session target dhcp** **Example:** `Router(config-dial-peer)# session target dhcp`	Specifies that the DHCP protocol is used to determine the IP address of the session target.
Step 7	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Enabling Forced Update of SIP Parameters via DHCP

In the event of changes to the SIP parameter values, a DHCP message called DHCPFORCERENEW can reset or apply a new set of values. The NGN can add or change phone number, SIP server address and domain name by sending DHCPFORCERENEW. When the SIP server receives the SIP parameter values, it compares the existing values to see if they are the same or if they have changed. If they are the same, the existing SIP parameters continue to be used. If they are different, the current phone number is unregistered and the new one registered, and the new SIP server address and domain name are used.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.

The DHCP provisioning of SIP parameters must be enabled.

This feature provides the ability for a DHCP server to add or change SIP signaling configuration and routing information related parameters via DHCP FORCERENEW. The DHCP client in IOS is required to restart REGISTRATION and use updated parameters for subsequent SIP dialogs

- Commands Required to turn on the feature.
  - dhcp interface <intf>
  - registrar dhcp
  - credentials dhcp password <password> realm <realm>

Restrictions

- DHCP Option 120 is the standard DHCP option (RFC3361) to get an SIP server address, and this can be used by any vendor DHCP server. Only one address is supported, which is in the IPv4 address format. Multiple IPv4 address entries are not supported. Additionally, a DNS name and any port number given behind the IPv4 address is not supported.
- DHCP Option 125 (RFC3925) provides vendor specific information. Its interpretation is tied up with the enterprise id. The primary and secondary phone numbers and domain are obtained using option 125 which is vendor specific. As long as other customers use the same format as in the NGN DHCP specification, they can leverage this feature.
- The presence of the primary number in sub-option 202 of DHCP option 125 is mandatory. There can only be one instance of the primary number and not multiple instances.

- Multiple secondary numbers in sub-option 203 of DHCP option 125 are supported. Up to five numbers are accepted and the rest are ignored. Also, they have to follow behind the primary number in the DHCP packet data.
- Authentication is not supported for REGISTER and INVITE messages sent from a CUBE that uses DHCP provisioning.
- The DHCP provisioning of SIP Parameters is only supported over one DHCP interface.
- The DHCP option is only available to be configured for the primary registrar. It will not be available for a secondary registrar.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip dhcp-client forcerenew**
4. **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	**ip dhcp-client forcerenew** **Example:** `Router> ip dhcp-client forcerenew`	Causes the DHCP server to force an immediate update to DHCP Client.
Step 4	**exit** **Example:** `Router> exit`	Exits the current mode.

Configuration Examples for Configurable SIP Parameters via DHCP

This section contains the following configuration examples:

- Configuring the DHCP Client: Example, page 247
- Enabling the SIP Configuration: Example, page 247
- Configuring a SIP Outbound Proxy Server in Voice Service VoIP Configuration Mode: Example, page 247
- Configuring a SIP Outbound Proxy Server in Dial Peer Configuration Mode: Example, page 247

- Enabling Forced Update of SIP Parameters via DHCP: Example, page 248

Configuring the DHCP Client: Example

The following is an example of how to enable the DHCP client:

```
Router> enable
Router# configure terminal
Router(config)# interface gigabitethernet 1/1
Router(config-if)# ip dhcp client request sip-server-address
Router(config-if)# ip dhcp client request vendor-identifying-specific
Router(config-if)# ip address dhcp
Router(config-if)# exit
```

Enabling the SIP Configuration: Example

The following is an example of how to enable the SIP configuration:

```
Router> enable
Router# configure terminal
Router(config)# interface gigabitethernet 1/0
Router(config-if)# sip-ua
Router(sip-ua)# dhcp interface gigabitethernet 1/0
Router(sip-ua)# registrar dhcp expires 90 random-contact refresh-ratio 90
Router(sip-ua)# credentials dhcp password cisco realm cisco.com
Router(sip-ua)# exit
```

Configuring a SIP Outbound Proxy Server in Voice Service VoIP Configuration Mode: Example

The following is an example of how to configure a SIP outbound proxy in voice service SIP configuration mode:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# outbound-proxy dhcp
Router(config-serv-if)# exit
```

Configuring a SIP Outbound Proxy Server in Dial Peer Configuration Mode: Example

The following is an example of how to configure a SIP outbound proxy in dial peer configuration mode:

```
Router> enable
Router# configure terminal
Router(config)# dial-peer voice 11 voip
Router(config-dial-peer)# session protocol sipv2
Router(config-dial-peer)# voice-class sip outbound-proxy dhcp
Router(config-dial-peer)# session target dhcp
Router(config-dial-peer)# exit
```

Enabling Forced Update of SIP Parameters via DHCP: Example

```
The following is an example of how to enable forced update of SIP parameters via DHCP:

Router> enable
Router# configure terminal
Router(config)# ip dhcp-client forcerenew
Router(config)# exit
```

Configuring SIP Listening Port

To manually change the SIP listen port for UDP/TCP/TLS calls, perform the steps in this section:

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- Configure the **shutdown** command in sip configuration mode first. This ensures that there are no active calls when the SIP listen port is changed. If SIP service is not shutdown, the listen-port command flashes an error message saying "shutdown SIP service before changing SIP listen port".
- This feature is applicable for both incoming and outgoing call SIP.
- The IP-to-IP gateway port number defined in global configuration will be used for both IN leg and OUT leg.

Restrictions

- Configuring SIP listening port on a dial-peer basis is not supported.
- Configuring the same listening port for both UDP/TCP and TLS is not supported.
- Configuring SIP listen port to a port that is already in use is not supported, and results in an error message.
- Changing the SIP listening port when Transport Process (TCP/UDP/TLS) services are shutdown, will not close or reopen the port. The only result is that the new port number is updated. The new port is bound when transport services (TCP/UDP/TLS) is enabled.
- Both **secure** and **non-secure** keywords are supported on Crypto images
- The **non-secure** keyword is supported on non-Crypto images.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **listen-port** {**non-secure** | **secure**} *port-number*

6. **exit**
7. **end**

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	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**listen-port** {**non-secure** | **secure**} *port-number* **Example:** `Router (config-voip-peer)# listen-port secure 3000`	Port number. Range: 1 to 65535. The default for UDP/TCP is 5060, the default for TLS is 5061. **Image Support** • The **secure** and **non-secure** keywords are supported on Crypto images. • The **non-secure** keyword is supported on non-Crypto images.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.
Step 7	**end** **Example:** `Router(config-voi-srv)# end`	Returns to privileged EXEC mode.

Configuring Bandwidth Parameters for SIP Calls

This feature provides a CLI command that is configured under each dialpeer that is triggered when an outbound SIP call is made using this dialpeer. The configured value for the Bandwidth command overwrite the default bandwidth that is determined by the codec selected. This command is helpful to allow the bandwidth to be signalled independent of the specific codec used

To manually change the SIP listen port for UDP/TCP/TLS calls, perform the steps in this section:

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- Configure the **shutdown** command in sip configuration mode first. This ensures that there are no active calls when the SIP listen port is changed. If SIP service is not shutdown, the listen-port command flashes an error message saying "shutdown SIP service before changing SIP listen port".
- This feature is applicable for both incoming and outgoing call SIP.
- The Cisco Unified BE port number defined in global configuration will be used for both IN leg and OUT leg.

Restrictions

- Configuring SIP listening port on a dial-peer basis is not supported.

Configuring Support for Session Refresh with Reinvites

Configuring support for session refresh with reinvites expands the ability of the Cisco Unified BE to receive a REINVITE message that contains either a session refresh parameter or a change in media via a new SDP and ensure the session does not time out. The **midcall-signaling** command distinguishes between the way a Cisco Unified Communications Express and Cisco Unified Border Element releases signaling messages. Most SIP-to-SIP video and SIP-to-SIP ReInvite-based supplementary services features require the Configuring Session Refresh with Reinvites feature to be configured.

Cisco IOS Release 12.4(15)XZ and Earlier Releases

Session refresh support via OPTIONS method. For configuration information, see the "Enabling In-Dialog OPTIONS to Monitor Active SIP Sessions" section on page 222.

Cisco IOS Release 12.4(15)XZ and Later Releases

Cisco Unified BE transparently passes other session refresh messages and parameters so that UAs and proxies can establish keepalives on a call.

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- The **allow-connections sip to sip** command must be configured before you configure the Session refresh with Reinvites feature. For more information and configuration steps see the "Configuring SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 190.

Restrictions

- SIP-to-SIP video calls and SIP-to-SIP ReInvite-based supplementary services fail if the **midcall-signaling** command is not configured.

Note The following features function if the **midcall-signaling** command is not configured: session refresh, fax, and refer-based supplementary services.

- Configuring Session Refresh with Reinvites is for SIP-to-SIP calls only. All other calls (H323-to-SIP, and H323-to-H323) do not require the **midcall-signaling** command be configured
- Configuring the Session Refresh with Reinvites feature on a dial-peer basis is not supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **midcall-signaling passthru**
6. **exit**
7. **end**

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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**midcall-signaling passthru** **Example:** `Router(conf-serv-sip)# midcall-signaling passthru`	Passes SIP messages from one IP leg to another IP leg.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.
Step 7	**end** **Example:** `Router(conf-serv-sip) end`	Returns to privileged EXEC mode.

Sending a SIP Registration Message from a Cisco Unified Border Element

The **credentials** command allows you to send a SIP registration message from a Cisco Unified Border Element in the UP state. Registration can include numbers, number ranges (such as E.164-numbers), or text information.

Before Cisco IOS Release12.4(24)T, a POTS dial peer was required to register numbers from a Cisco Unified Border Element in the UP state. The **credentials** command is modified in Release 12.4(24s)T to allow for registration of the E.164-numbers, if there is no POTS dial peer.

Prerequisites

- To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- Configure a registrar in sip user-agent configuration mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. sip-uaF
4. **credentials username** *username* **password** *password* **realm** *domain-name*

5. exit
6. end

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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters sip user-agent configuration mode.
Step 4	**credentials username** *username* **password** *password* **realm** *domain-name* **Example:** `Router(config-sip-ua)# credentials username alex password test realm cisco.com`	Enters SIP digest credentials in sip-ua configuration mode.
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 6	**end** **Example:** `Router(config)# end`	Returns to privileged EXEC mode.

Configuring Adjustable Timers for Registration Refresh and Retries

Configuring Adjustable Timers for Registration Refresh and Retries provides the ability for IOS software to refresh the REGISTER at a configurable fraction of the expiry timer specified in the 200 OK response of the REGISTER request. The feature also provides the ability to retransmit REGISTER upon receiving failure responses as per the min-expires header value in a “423 interval too brief” response, or retry-after if header value if present or terminal re-registration interval if retry-after header value is absent in 4xx/5xx/6xx responses. Additionally, the ability to retransmit REGISTER per Timer E up to 32 seconds, and at a command line interface controlled random interval thereafter.

This feature addresses the UNI SIP registration specification requirements on Cisco Unified Border Element to interwork CTS over NGN and includes the following are SIP registration enhancements:

423 Interval Too Brief Response Handling

Cisco Unified Border Element retransmits the REGISTER request with the received Min-Expires value in the 423 response. The retransmit interval is the same as the configured REGISTER refresh ratio.

If the registration response from the REGISTRAR server is a “423 Interval Too Brief”, the configured registration expires time-value sent in the REGISTER message does not apply. The 423 response contains the acceptable expires time value in the Min-Expires header. The newly received time value is then used in the Expires header when the next registration refresh request is sent.

4xx/5xx/6xx Error Response Handling (Except 423)

If the registration response from the REGISTRAR server is a 4xx/5xx/6xx (except 423) message, an error has occurred. The retransmit interval uses the value in the Retry-After header if present in the 4xx/5xx/6xx response. The only supported Retry-After header format is ‘Retry-After:1800’. If “Retry-After” header is not present in the error response, the configured refresh ratio and “Expires” time value will be used to calculate the interval between the sending of the next REGISTER message or it will be the default retransmit interval.

Configurable REGISTER Refresh Ratio

The Cisco Unified Border Element sends REGISTER refresh at 40% to 50% of the expiry time as specified in 200 OK response of REGISTER request. Use the refresh-ratio keyword to configure the REGISTER refresh ratio. If the refresh-ratio option is not configured, the default REGISTER refresh ratio is 80% of the expiry timer. The minimum refresh interval is one minute.

No REGISTER Response Handling

The Cisco Unified Border Element handles no response to REGISTER by retransmitting at intervals Timer E for up to a maximum of 32 seconds. If no REGISTER response is received from the REGISTRAR server, the REGISTER message will be retransmitted. By configuring the **retry register** command to 10, the Cisco Unified Border Element retransmits the REGISTER (starting at 500 ms) and continues to retransmit at double the rate, to a maximum of 4 seconds. The default REGISTER retransmit count is six retries, after which the Cisco Unified Border Element retries REGISTER request at a random interval (5 to 10 minutes).

There is a two minute interval after which the REGISTER retransmits begin again. The **retry register exhausted-random-interval** command allows the user to set a desired interval after the number of REGISTER retransmits have been exhausted. This also allows the user to set a range in which a number (in minutes) is randomly generated and used as the interval between retransmission exhaustion.

The default REGISTER refresh ratio is eighty percent (80%) of the expiry time. The default REGISTER error retransmit interval is 5% of the configured expiry time or two minutes, whichever is greater.

Random String in REGISTER Contact

Cisco Unified Border Element uses a random string in the Contact header of the REGISTER message. The random string consists of alphanumeric characters. A different random string is generated and used for each number registered.

Prerequisites

To enable this feature, you must have Cisco IOS Release XXX or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the “Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element” section on page 344.

To configure Adjustable Timers for Registration Refresh and Retries, perform the steps in this section:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registrar expires** *seconds* **refresh-ratio** *seconds* **random-contact**
5. **retry register retries exhausted-random-interval minimum** *minutes* **maximum** *minutes*
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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters the SIP user agent (sip-ua) configuration mode to configure SIP-UA related commands.
Step 4	**registrar expires** *seconds* **refresh-ratio** *seconds* **random-contact** **Example:** `Router(config-sip-ua)# registrar expires 60 refresh-ratio 45 random-contact`	Configures the SIP registrar for retry attempts. The keywords are as follows: • **expires**—Registration expires time. Range is 60 to 65535. Default is 3600. • **refresh-ratio**—Registration refresh ratio expressed as a percentage. Valid entries are 1 to 100. The default is 80. • **random-contact**—Random String Contact Header.

	Command or Action	Purpose
Step 5	**retry register** *retries* **exhausted-random-interval minimum** *minutes* **maximum** *minutes* **Example:** `Router(config-sip-ua)# retry register 4 exhausted-random-interval minimum 4 maximum 5`	Sets the total number of SIP register messages that the gateway should send. The keywords are as follows: • *retries*—Total number of register messages that the gateway should send. The range is from 1 to 10. The default is 10 retries. • **exhausted-random-interval**—specifies that the register request is generated within the defined time interval. • **minimum** *minutes*—Sets the minimum time interval, in minutes. • **maximum** *minutes*—Sets the maximum time interval in minutes.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.

Cisco Unified Border Element Support for SRTP-RTP Internetworking

The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature allows secure enterprise-to-enterprise calls. The feature also provides operational enhancements for Session Initiation Protocol (SIP) trunks from Cisco Unified Call Manager and Cisco Unified Call Manager Express. Support for Secure Real-Time Transport Protocol (SRTP)-RTP internetworking between one or multiple Cisco Unified Border Elements is enabled for SIP-SIP audio calls.

Prerequisites for Cisco Unified Border Element Support for SRTP-RTP Internetworking

- To enable this feature, you must have Cisco IOS Release 12.4(22(YB) or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the "Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344.
- The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature is supported in Cisco Unified CallManager 7.0 and later releases.

Restrictions for Cisco Unified Border Element Support for SRTP-RTP Internetworking

The following features are not supported by the Cisco Unified Border Element Support for SRTP-RTP Internetworking feature:

- Voice-class codec
- Call admission control (CAC) support
- Rotary SIP-SIP
- T.38 Fax
- Early offer to delayed offer calls
- Delayed offer to early offer calls

Information About Cisco Unified Border Element Support for SRTP-RTP Internetworking

To configure support for SRTP-RTP internetworking, you should understand the following concepts:

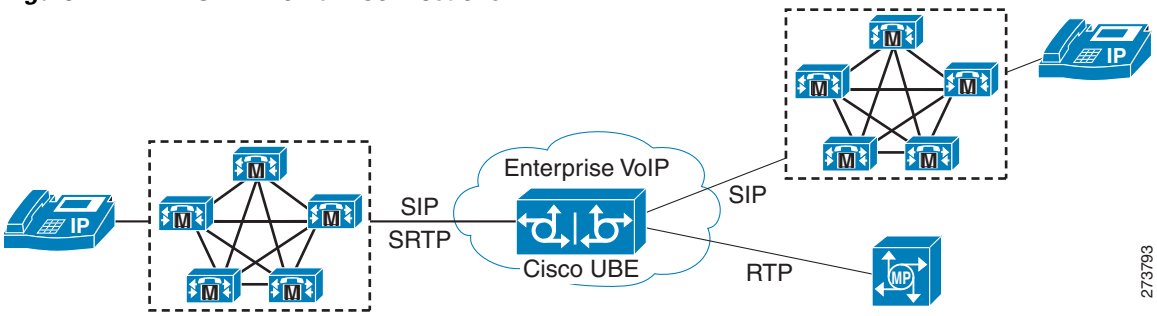
- Cisco Unified Border Element Support for SRTP-RTP Internetworking, page 257
- TLS on the Cisco Unified Border Element, page 258

Cisco Unified Border Element Support for SRTP-RTP Internetworking

The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature connects SRTP Cisco Unified CallManager domains with the following:

- RTP Cisco Unified CallManager domains. Domains that do not support SRTP, or have not been configured for SRTP, as shown in Figure 4.
- RTP Cisco applications or servers. For example, Cisco Unified MeetingPlace, Cisco WebEx, or Cisco Unity, which do not support SRTP, or have not been configured for SRTP, or are resident in a secure data center, as shown in Figure 4.
- RTP to third-party equipment. For example, IP trunks to PBXs or virtual machines, which do not support SRTP.

Figure 4 SRTP Domain Connections



The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature connects SRTP enterprise domains to RTP SIP provider (SP) SIP trunks. SRTP-RTP internetworking connects RTP enterprise networks with SRTP over an external network between businesses. This provides flexible secure business-to-business communications without the need for static IPsec tunnels or the need to deploy SRTP within the enterprise, as shown in Figure 5. SRTP-RTP internetworking also connects SRTP enterprise networks with static IPsec over external networks, as shown in Figure 6.

Figure 5 Secure Business-to-Business Communications

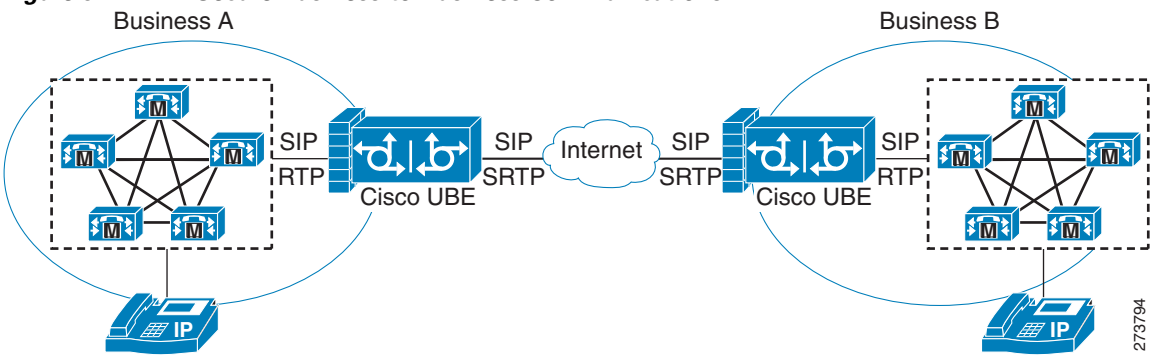
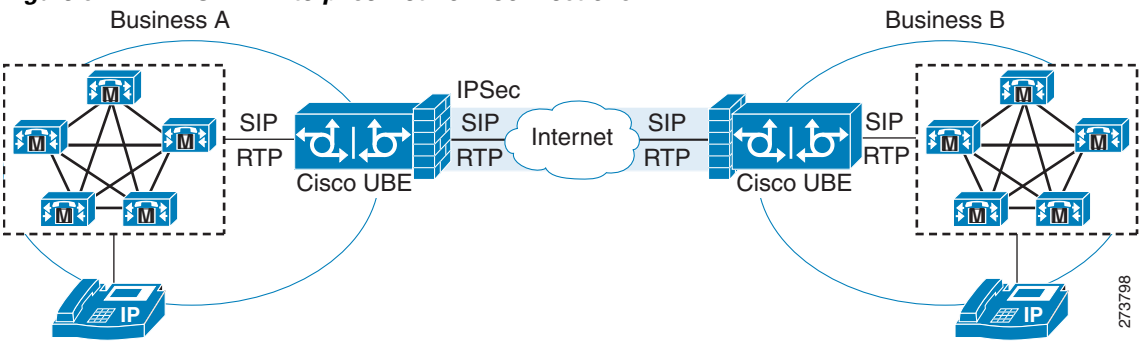


Figure 6 SRTP Enterprise Network Connections



SRTP-RTP internetworking on the Cisco Unified Border Element in a network topology uses single pair key generation. Existing audio and dual-tone multifrequency (DTMF) transcoding is used to support voice calls. SRTP-RTP internetworking support is provided in both flow-through and high-density mode. SRTP-SRTP pass-through is not impacted.

SRTP is configured on one dial peer and RTP is configured on the other dial peer using the **srtp** and **srtp fallback** commands. The dial-peer configuration takes precedence over the global configuration on the Cisco Unified Border Element.

Fallback handling occurs if one of the call endpoints does not support SRTP. The call can fall back to RTP-RTP, or the call can fail, depending on the configuration. Fallback takes place only if the **srtp fallback** command is configured on the respective dial peer. RTP-RTP fall back occurs when no transcoding resources are available for SRTP-RTP internetworking.

TLS on the Cisco Unified Border Element

The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature allows Transport Layer Security (TLS) to be enabled or disabled between the SCCP server and SCCP client. By default TLS is enabled, which provides added protection at transport level and ensures that SRTP keys are not easily accessible. Once TLS is disabled, the SRTP keys are not protected.

SRTP-RTP internetworking is available with normal and universal transcoders. The transcoder on the Cisco Unified Border Element is invoked using SCCP messaging between the SCCP server and the SCCP client. The SCCP messages carry the SRTP keys to the digital signal processor (DSP) farm at the SCCP client. The transcoder can be within the same router or can be located in a separate router. TLS

should be disabled only when the transcoder is located in the same router. To disable TLS, configure the **no** form of the **tls** command in dsp farm profile configuration mode. Disabling TLS improves CPU performance.

How to Configure Cisco Unified Border Element Support for SRTP-RTP Internetworking

This section contains the following task:

- Configuring Cisco Unified Border Element Support for SRTP-RTP Internetworking, page 259 (required)

Configuring Cisco Unified Border Element Support for SRTP-RTP Internetworking

Configuring the Cisco Unified Border Element Support for SRTP-RTP Internetworking feature consists of the following tasks:

- Configuring the Certificate Authority, page 259 (required)
- Configuring a Trustpoint for the Secure Universal Transcoder, page 260 (required)
- Configuring DSP Farm Services, page 262 (required)
- Associating SCCP to the Secure DSP Farm Profile, page 263 (required)
- Registering the Secure Universal Transcoder to the Cisco Unified Border Element, page 266 (required)
- Configuring SRTP-RTP Internetworking Support, page 268 (required)

Configuring the Certificate Authority

Perform the steps described in this section to configure the certificate authority.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip http server**
4. **crypto pki server** *cs-label*
5. **database level complete**
6. **grant auto**
7. **no shutdown**
8. **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	**ip http server** **Example:** `Router(config)# ip http server`	Enables the HTTP server on your IP or IPv6 system, including the Cisco web browser user interface.
Step 4	**crypto pki server** *cs-label* **Example:** `Router(config)# crypto pki server 3854-cube`	Enables a Cisco IOS certificate server and enters certificate server configuration mode. • In the example, 3845-cube is specified as the name of the certificate server.
Step 5	**database level complete** **Example:** `Router(cs-server)# database level complete`	Controls what type of data is stored in the certificate enrollment database. • In the example, each issued certificate is written to the database.
Step 6	**grant auto** **Example:** `Router(cs-server)# grant auto`	Specifies automatic certificate enrollment.
Step 7	**no shutdown** **Example:** `Router(cs-server)# no shutdown`	Reenables the certificate server. • Create and enter a new password when prompted.
Step 8	**exit** **Example:** `Router(cs-server)# exit`	Exits certificate server configuration mode.

Configuring a Trustpoint for the Secure Universal Transcoder

Perform the steps in this section to configure, authenticate, and enroll the trustpoint for the secure universal transcoder.

Prerequisites

Before you configure the trustpoint for the secure universal transcoder, you should configure the certificate authority, as described in the "Configuring the Certificate Authority" section on page 259.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **crypto pki trustpoint** *name*
4. **enrollment url** *url*
5. **serial-number**
6. **revocation-check** *method*
7. **rsakeypair** *key-label*
8. **end**
9. **crypto pki authenticate** *name*
10. **crypto pki enroll** *name*
11. **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	**crypto pki trustpoint** *name* **Example:** `Router(config)# crypto pki trustpoint secdsp`	Declares the trustpoint that the router uses and enters ca-trustpoint configuration mode. • In the example, the trustpoint is named secdsp.
Step 4	**enrollment url** *url* **Example:** `Router(ca-trustpoint)# enrollment url http://10.13.2.52:80`	Specifies the enrollment parameters of a certification authority (CA). • In the example, the URL is defined as http://10.13.2.52:80
Step 5	**serial-number** **Example:** `Router(ca-trustpoint)# serial-number`	Specifies whether the router serial number should be included in the certificate request.
Step 6	**revocation-check** *method* **Example:** `Router(ca-trustpoint)# revocation-check crl`	Checks the revocation status of a certificate. • In the example, the certificate revocation list checks the revocation status.

	Command or Action	Purpose
Step 7	**rsakeypair** *key-label* **Example:** `Router(ca-trustpoint)# rsakeypair 3845-cube`	Specifies which key pair to associate with the certificate. • In the example, the key pair, 3845-cube generated during enrollment is associated with the certificate.
Step 8	**end** **Example:** `Router(ca-trustpoint)# end`	Exits ca-trustpoint configuration mode.
Step 9	**crypto pki authenticate** *name* **Example:** `Router(config)# crypto pki authenticate secdsp`	Authenticates the CA. • Accept the trustpoint CA certificate if prompted.
Step 10	**crypto pki enroll** *name* **Example:** `Router(config)# crypto pki enroll secdsp`	Obtains the certificate for the router from the CA. • Create and enter a new password if prompted. • Request a certificate from the CA if prompted.
Step 11	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode.

Configuring DSP Farm Services

Perform the steps in this section to configure DSP farm services.

Prerequisites

Before you configure DSP farm services, you should configure the trustpoint for the secure universal transcoder, as described in the "Configuring a Trustpoint for the Secure Universal Transcoder" section on page 260.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-card** *slot*
4. **dspfarm**
5. **dsp services dspfarm**
6. Repeat Steps 3,4, and 5 to configure a second voice card.
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-card** *slot* **Example:** `Router(config)# voice-card 0`	Configures a voice card and enters voice-card configuration mode. • In the example, voice card 0 is configured.
Step 4	**dspfarm** **Example:** `Router(config-voicecard)# dspfarm`	Adds a specified voice card to those participating in a DSP resource pool.
Step 5	**dsp services dspfarm** **Example:** `Router(config-voicecard)# dsp services dspfarm`	Enables DSP farm services for a particular voice network module.
Step 6	Repeat Steps 3, 4, and 5 to configure a second voice card.	—
Step 7	**exit** **Example:** `Router(config-voicecard)# exit`	Exits voice-card configuration mode.

Associating SCCP to the Secure DSP Farm Profile

Perform the steps in this section to associate SCCP to the secure DSP farm profile.

Prerequisites

Before you associate SCCP to the secure DSP farm profile, you should configure DSP farm services, as described in the "Configuring DSP Farm Services" section on page 262.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sccp local** *interface-type interface-number*
4. **sccp ccm** *ip-address* **identifier** *identifier-number* **version** *version-number*
5. **sccp**

6. **associate ccm** *identifier-number* **priority** *priority-number*
7. **associate profile** *profile-identifier* **register** *device-name*
8. **dspfarm profile** *profile-identifier* **transcode universal security**
9. **trustpoint** *trustpoint-label*
10. **codec** *codec-type*
11. Repeat Step 10 to configure required codecs.
12. **maximum sessions** *number*
13. **associate application sccp**
14. **no shutdown**
15. **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	**sccp local** *interface-type interface-number* **Example:** `Router(config)# sccp local GigabitEthernet 0/0`	Selects the local interface that SCCP applications (transcoding and conferencing) use to register with Cisco CallManager. • In the example, the following parameters are set: – GigabitEthernet is defined as the interface type that the SCCP application uses to register with Cisco CallManager. – The interface number that the SCCP application uses to register with Cisco CallManager is specified as 0/0.
Step 4	**sccp ccm** *ip-address* **identifier** *identifier-number* **version** *version-number* **Example:** `Router(config)# sccp ccm 10.13.2.52 identifier 1 version 5.0.1`	Adds a Cisco Unified Communications Manager server to the list of available servers. • In the example, the following parameters are set: – 10.13.2.52 is configured as the IP address of the Cisco Unified Communications Manager server. – The number 1 identifies the Cisco Unified Communications Manager server. – The Cisco Unified Communications Manager version is identified as 5.0.1.

	Command or Action	Purpose
Step 5	**sccp** **Example:** `Router(config)# sccp`	Enables the SCCP and its related applications (transcoding and conferencing) and enters SCCP Cisco CallManager configuration mode.
Step 6	**associate ccm** identifier-number **priority** *priority-number* **Example:** `Router(config-sccp-ccm)# associate ccm 1 priority 1`	Associates a Cisco Unified CallManager with a Cisco CallManager group and establishes its priority within the group. • In the example, the following parameters are set: – The number 1 identifies the Cisco Unified CallManager. **Note** The priority must match the order of the call manager group associated with this device within call manager.
Step 7	**associate profile** *profile-identifier* **register** *device-name* **Example:** `Router(config-sccp-ccm)# associate profile 1 register sxcoder`	Associates a DSP farm profile with a Cisco CallManager group. • In the example, the following parameters are set: – The number 1 identifies the DSP farm profile. – Sxcoder is configured as the user-specified device name in Cisco Unified CallManager.
Step 8	**dspfarm profile** *profile-identifier* **transcode universal security** **Example:** `Router(config-sccp-ccm)# dspfarm profile 1 transcode universal security`	Defines a profile for DSP farm services and enters DSP farm profile configuration mode. • In the example, the following parameters are set: – Profile 1 is enabled for transcoding. – Profile 1 is enabled for secure DSP farm services.
Step 9	**trustpoint** *trustpoint-label* **Example:** `Router(config-dspfarm-profile)# trustpoint secdsp`	Associates a trustpoint with a DSP farm profile. • In the example, the trustpoint to be associated with the DSP farm profile is labeled secdsp.
Step 10	**codec** *codec-type* **Example:** `Router(config-dspfarm-profile)# codec g711ulaw`	Specifies the codecs that are supported by a DSP farm profile. • In the example, the g711ulaw codec is specified.
Step 11	Repeat Step 10 to configure required codecs.	—

	Command or Action	Purpose
Step 12	**maximum sessions** *number* **Example:** `Router(config-dspfarm-profile)# maximum sessions 84`	Specifies the maximum number of sessions that are supported by the profile. • In the example, a maximum of 84 sessions are supported by the profile. The maximum number of sessions depends on the number of DSPs available for transcoding.
Step 13	**associate application sccp** **Example:** `Router(config-dspfarm-profile)# associate application sccp`	Associates SCCP to the DSP farm profile.
Step 14	**no shutdown** **Example:** `Router(config-dspfarm-profile)# no shutdown`	Allocates DSP farm resources and associates with the application.
Step 15	**exit** **Example:** `Router(config-dspfarm-profile)# exit`	Exits DSP farm profile configuration mode.

Registering the Secure Universal Transcoder to the Cisco Unified Border Element

Perform the steps in this section to register the secure universal transcoder to the Cisco Unified Border Element. The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature supports both secure transcoders and secure universal transcoders.

Prerequisites

Before you register the secure universal transcoder to the Cisco Unified Border Element, you should associated SCCP to the secure DSP farm profile, as described in the "Associating SCCP to the Secure DSP Farm Profile" section on page 263.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **sdspfarm transcode sessions** *number*
5. **sdspfarm tag** *number device-name*
6. **em logout** *time1 time2 time3*
7. **max-ephones** *max-phones*
8. **max-dn** *max-directory-numbers*
9. **ip source-address** *ip-address*
10. **secure-signaling trustpoint** *label*

11. **tftp-server-credentials trustpoint** *label*
12. **create cnf-files**
13. **no sccp**
14. **sccp**
15. **end**

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	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**sdspfarm transcode sessions** *number* **Example:** `Router(config-telephony)# sdspfarm transcode sessions 84`	Specifies the maximum number of transcoding sessions allowed per Cisco CallManager Express router. • In the example, a maximum of 84 DSP farm sessions are specified.
Step 5	**sdspfarm tag** *number device-name* **Example:** `Router(config-telephony)# sdspfarm tag 1 sxcoder`	Permits a DSP farm to be to registered to Cisco Unified CallManager Express and associates it with an SCCP client interface's MAC address. • In the example, DSP farm 1 is associated with the sxcoder device.
Step 6	**em logout** *time1 time2 time3* **Example:** `Router(config-telephony)# em logout 0:0 0:0 0:0`	Configures three time-of-day based timers for automatically logging out all Extension Mobility feature users. • In the example, all users are logged out from Extension Mobility after 00:00.
Step 7	**max-ephones 4** **Example:** `Router(config-telephony)# max-ephones 4`	Sets the maximum number of Cisco IP phones to be supported by a Cisco CallManager Express router. • In the example, a maximum of four phones are supported by the Cisco CallManager Express router.
Step 8	**max-dn** *max-directory-numbers* **Example:** `Router(config-telephony)# max-dn 4`	Sets the maximum number of extensions (ephone-dns) to be supported by a Cisco Unified CallManager Express router. • In the example, a maximum of four extensions is allowed.

	Command or Action	Purpose
Step 9	**ip source-address** *ip-address* **Example:** `Router(config-telephony)# ip source-address 10.13.2.52`	Identifies the IP address and port through which IP phones communicate with a Cisco Unified CallManager Express router. • In the example, 10.13.2.52 is configured as the router IP address.
Step 10	**secure-signaling trustpoint** *label* **Example:** `Router(config-telephony)# secure-signaling trustpoint secdsp`	Specifies the name of the PKI trustpoint with the certificate to use for TLS handshakes with IP phones on TCP port 2443. • In the example, PKI trustpoint secdsp is configured.
Step 11	**tftp-server-credentials trustpoint** *label* **Example:** `Router(config-telephony)# tftp-server-credentials trustpoint scme`	Specifies the PKI trustpoint that signs the phone configuration files. • In the example, PKI trustpoint scme is configured.
Step 12	**create cnf-files** **Example:** `Router(config-telephony)# create cnf-files`	Builds the XML configuration files that are required for IP phones in Cisco Unified CallManager Express.
Step 13	**no sccp** **Example:** `Router(config-telephony)# no sccp`	Disables SCCP and its related applications (transcoding and conferencing) and exits telephony-service configuration mode.
Step 14	**sccp** **Example:** `Router(config)# sccp`	Enables SCCP and its related applications (transcoding and conferencing).
Step 15	**end** **Example:** `Router(config)# end`	Exits global configuration mode.

Configuring SRTP-RTP Internetworking Support

Perform the steps in this section to enable SRTP-RTP internetworking support between one or multiple Cisco Unified Border Elements for SIP-SIP audio calls. In this task, RTP is configured on the incoming call leg and SRTP is configured on the outgoing call leg.

Prerequisites

Before you configure the Cisco Unified Border Element Support for SRTP-RTP Internetworking feature, you should register the secure universal transcoder to the Cisco Unified Border Element, as described in the "Registering the Secure Universal Transcoder to the Cisco Unified Border Element" section on page 266.

Restrictions

The Cisco Unified Border Element Support for SRTP-RTP Internetworking feature is available only on platforms that support transcoding on the Cisco Unified Border Element. The feature is also available only on secure Cisco IOS images on the Cisco Unified Border Element.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **destination-pattern** *string*
5. **session protocol sipv2**
6. **session target ipv4:***destination-address*
7. **incoming called-number** *string*
8. **codec** *codec*
9. **end**
10. **dial-peer voice** *tag* **voip**
11. Repeat Steps 4, 5, 6, and 7 to configure a second dial peer.
12. **srtp**
13. **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 201 voip`	Defines a particular dial peer, to specify the method of voice encapsulation, and enters dial peer voice configuration mode. • In the example, the following parameters are set: – Dial peer 201 is defined. – VoIP is shown as the method of encapsulation.
Step 4	**destination-pattern** *string* **Example:** `Router(config-dial-peer)# destination-pattern 5550111`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer string. • In the example, 5550111 is specified as the pattern for the telephone number.

	Command or Action	Purpose
Step 5	**session protocol sipv2** **Example:** `Router(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the packet network. • In the example, the **sipv2** keyword is configured so that the dial peer uses the IEFTF SIP.
Step 6	**session target ipv4:***destination-address* **Example:** `Router(config-dial-peer)# session target ipv4:10.13.25.102`	Designates a network-specific address to receive calls from a VoIP or VoIPv6 dial peer. • In the example, the IP address of the dial peer to receive calls is configured as 10.13.25.102.
Step 7	**incoming called-number** *string* **Example:** `Router(config-dial-peer)# incoming called-number 5550111`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer. • In the example, 5550111 is specified as the pattern for the E.164 or private dialing plan telephone number.
Step 8	**codec** *codec* **Example:** `Router(config-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for the dial peer. • In the example, G.711 mu-law at 64,000 bps, is specified as the voice coder rate for speech.
Step 9	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial peer voice configuration mode.
Step 10	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 200 voip`	Defines a particular dial peer, to specify the method of voice encapsulation, and enters dial peer voice configuration mode. • In the example, the following parameters are set: – Dial peer 200 is defined. – VoIP is shown as the method of encapsulation.
Step 11	Repeat Steps 4, 5, 6, and 7 to configure a second dial peer.	—
Step 12	**srtp** **Example:** `Router(config-dial-peer)# srtp`	Specifies that SRTP is used to enable secure calls for the dial peer.
Step 13	**codec** *codec* **Example:** `Router(config-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for the dial peer. • In the example, G.711 mu-law at 64,000 bps, is specified as the voice coder rate for speech.
Step 14	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode.

Troubleshooting Tips

The following commands can help troubleshoot Cisco Unified Border Element support for SRTP-RTP internetworking:

- **show crypto pki certificates**
- **show sccp**
- **show sdspfarm**

Configuring Assisted Real-time Transport Control Protocol (RTCP) Report Generation

The assisted Real-time Transport Control Protocol (RTCP) feature adds the ability for Cisco Unified Border Element (Cisco UBE) to generate standard RTCP keepalive reports on behalf of endpoints. RTCP reports determine the liveliness of a media session during prolonged periods of silence, such as call hold or mute. Therefore, it is important for the Cisco UBE to generate RTCP reports irrespective of whether the endpoints send or receive media.

Cisco UBE generates RTCP report only when inbound and outbound call legs are SIP, or SIP to H.323, or H.323 to SIP.

Restrictions

- RTCP report generation over IPv6 is not supported.
- RTCP report generation is not supported for Secure Real-time Transport Protocol (SRTP) or SRT Control Protocol (SRTCP) pass-through as Cisco UBE is not aware of the media encryption or decryption keys.
- RTCP report generation is not supported for loopback calls, T.38 fax, and modem relay calls.
- RTCP or SRTCP report generation is not supported when Cisco UBE inserts a Digital Signal Processor (DSP) for RTP-SRTP interworking on RTP and SRTP call legs.
- RTCP report generation is not supported when there is a call hold with an invalid media address such as 0.0.0.0 in Session Description Protocol (SDP) or Open Logical Channel (OLC).
- RTCP report generation is not supported for RTCP multiplexed with RTP on the same address and port.
- RTCP report generation is not supported on enterprise aggregation services routers (ASR) Cisco UBE.
- RTCP packet generation is not supported on the SIP leg when the H.323 leg puts the SIP leg on hold in a Slow Start to Delayed-Offer call.

Configuring RTCP Report Generation on Cisco UBE

RTCP keepalive packets indicate session liveliness. When configured on Cisco UBE, RTCP keepalive packets are sent on both inbound and outbound SIP or H.323 call legs.

Perform this task to configure RTCP report generation on Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type* **to** *to-type*
5. **rtcp keepalive**
6. **end**

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 voice service configuration mode.
Step 4	**allow-connections** *from-type* **to** *to-type* **Example:** `Router(conf-voi-serv)# allow-connections sip to sip`	Allows connections between SIP endpoints in a VoIP network.
Step 5	**rtcp keepalive** **Example:** `Router(conf-voi-serv)# rtcp keepalive`	Configures RTCP keepalive report generation.
Step 6	**end** **Example:** `Router(conf-voi-serv)# end`	Exits voice service configuration mode and returns to privileged EXEC mode.

Troubleshooting Tips

Use the following debug commands for debugging related to RTCP keepalive packets:

- **debug voip rtcp packet**—Shows details related to RTCP keepalive packets such as RTCP sending and receiving paths, Call ID, Globally Unique Identifier (GUID), packet header, and so on.

```
Router# debug voip rtcp packet
```

```
01:06:27.450: //6/xxxxxxxxxxx/RTP//Event/voip_rtp_send_rtcp_keepalive: Generate RTCP
Keepalive
*Mar 17 01:06:27.450: rtcp_send_report: Attributes
        (src ip=192.168.30.3, src port=17101, dst ip=192.168.30.4, dst port=18619
         bye=0, initial=1, ssrc=0x07111E02, keepalive=1)
*Mar 17 01:06:27.450: rtcp_construct_keepalive_report: Constructed Report
        (rtcp=0x2E5AF214, ssrc=0x07111E02, source->ssrc=0x00001E03, total_len=36)
2E5AF210:          80C90001 07111E02 81CA0006      .I.......J..
2E5AF220: 07111E02 010F302E 302E3040 392E3435  ......0.0.0@9.45
2E5AF230: 2E33302E 33000000 00                 .30.3....
```

Caution Under moderate traffic loads, the **debug voip rtp packet** command produces a high volume of output and the command should be enabled only when the call volume is very low.

- **debug voip rtp packet**—Shows details about VoIP RTP packet debugging trace.

```
Router# debug voip rtp packet

VOIP RTP All Packets debugging is on
```

- **debug voip rtp session**—Shows all RTP session debug information.

```
Router# debug voip rtp session

VOIP RTP All Events debugging is on
```

- **debug voip rtp error**—Shows details about debugging trace for RTP packet error cases.

```
Router# debug voip rtp error

VOIP RTP Errors debugging is on
```

- **debug ip rtp protocol**—Shows details about RTP protocol debugging trace.

```
Router# debug ip rtp protocol

RTP protocol debugging is on
```

- **debug voip rtcp session**—Shows all RTCP session debug information.

```
Router# debug voip rtcp session

VOIP RTCP Events debugging is on
```

- **debug voip rtcp error**— Shows details about debugging trace for RTCP packet error cases.

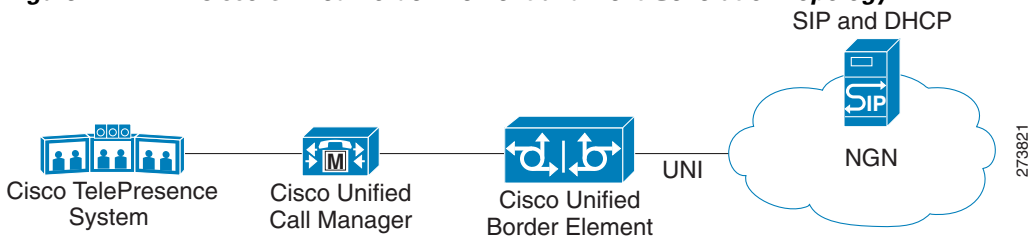
```
Router# debug voip rtcp error

VOIP RTCP Errors debugging is on
```

Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco UBE

Figure 7 shows a typical network topology where the Cisco Unified Border Element is configured to route messages between a call manager system (such as the Cisco Unified Call Manager) and a Next Generation Network (NGN).

Figure 7 Cisco Unified Border Element and Next Generation Topology



Devices that connect to an NGN must comply with the User-Network Interface (UNI) specification. The Cisco Unified Border Element supports the NGN UNI specification and can be configured to interconnect NGN with other call manager systems, such us the Cisco Unified Call Manager.

The Cisco Unified Border Element supports the following:

- the use of P-Preferred Identity (PPID), P-Asserted Identity (PAID), Privacy, P-Called Party Identity (PCPID), in INVITE messages
- the translation of PAID headers to PPID headers and vice versa
- the translation of From: or RPID headers to PAID or PPID headers and vice versa
- the configuration and/or pass through of privacy header values
- the use of the PCPID header to route INVITE messages
- the use of multiple PAURI headers in the response messages (200 OK) it receives to REGISTER messages

P-Preferred Identity and P-Asserted Identity Headers

NGN servers use the PPID header to identify the preferred number that the caller wants to use. The PPID is part of INVITE messages sent to the NGN. When the NGN receives the PPID, it authorizes the value, generates a PAID based on the preferred number, and inserts it into the outgoing INVITE message towards the called party.

However, some call manager systems, such as Cisco Unified Call Manager 5.0, use the Remote-Party Identity (RPID) value to send calling party information. Therefore, the Cisco Unified Border Element must support building the PPID value for an outgoing INVITE message to the NGN, using the RPID value or the From: value received in the incoming INVITE message. Similarly, CUBE supports building the RPID and/or From: header values for an outgoing INVITE message to the call manager, using the PAID value received in the incoming INVITE message from the NGN.

In non-NGN systems, the Cisco Unified Border Element can be configured to translate between PPID and PAID values, and between From: or RPID values and PAID/PPID values, at global and dial-peer levels.

In configurations where all relevant servers support the PPID or PAID headers, the Cisco Unified Border Element can be configured to transparently pass the header.

Note If the NGN sets the From: value to anonymous, the PAID is the only value that identifies the caller.

Table 4 describes the types of INVITE message header translations supported by the Cisco Unified Border Element. It also includes information on the configuration commands to use to configure P-header translations.

Note Table 4 shows the P-header translation configuration settings only. In addition to configuring these settings, you must configure other system settings (such as the session protocol).

Table 4 P-header Configuration Settings

Incoming Header	Outgoing Header	Configuration Notes
From:	PPID	To enable the translation to PPID headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id ppi` To enable the translation to PPID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id ppi`
From:	PAID	To enable the translation to PAID headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id pai` To enable the translation to PAID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id pai`
From:	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# remote-party-id` This is the default system behavior. **Note** If both, **remote-party-id** and **asserted-id** commands are configured, then the **asserted-id** command takes precedence over the **remote-part-id** command.
PPID	PAID	To enable the translation to PAID privacy headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id pai` To enable the translation to PAID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id pai`
PPID	From:	By default, the translation to RPID headers is enabled and the system translates PPID headers in incoming messages to RPID headers in the outgoing messages. To disable the default behavior and enable the translation from PPID to From: headers, use the **no remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# no remote-party-id`
PPID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# remote-party-id` This is the default system behavior.

Table 4 P-header Configuration Settings (continued)

Incoming Header	Outgoing Header	Configuration Notes
PAID	PPID	To enable the translation to PPID privacy headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id ppi` To enable the translation to PPID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id ppi`
PAID	From:	By default, the translation to RPID headers is enabled and the system translates PPID headers in incoming messages to RPID headers in the outgoing messages. To disable the default behavior and enable the translation from PPID to From: headers, use the **no remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# no remote-party-id`
PAID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# remote-party-id` This is the default system behavior.
RPID	PPID	To enable the translation to PPID privacy headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id ppi` To enable the translation to PPID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id ppi`
RPID	PAID	To enable the translation to PAID privacy headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. For example: `Router(conf-serv-sip)# asserted-id pai` To enable the translation to PAID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: `Router(config-dial-peer)# voice-class sip asserted-id pai`
RPID	From:	By default, the translation to RPID headers is enabled and the system translates PPID headers in incoming messages to RPID headers in the outgoing messages. To disable the default behavior and enable the translation from PPID to From: headers, use the **no remote-party-id** command in SIP user-agent configuration mode. For example: `Router(config-sip-ua)# no remote-party-id`

Privacy

If the user is subscribed to a privacy service, the Cisco Unified Border Element can support privacy using one of the following methods:

- Using prefixes

  The NGN dial plan can specify prefixes to enable privacy settings. For example, the dial plan may specify that if the caller dials a prefix of 184, the calling number is not sent to the called party.

The dial plan may also specify that the caller can choose to send the calling number to the called party by dialing a prefix of 186. Here, the Cisco Unified Border Element transparently passes the prefix as part of the called number in the INVITE message.

The actual prefixes for the network are specified in the dial plan for the NGN, and can vary from one NGN to another.

- Using the Privacy header

  If the Privacy header is set to None, the calling number is delivered to the called party. If the Privacy header is set to a Privacy:id value, the calling number is not delivered to the called party.

- Using Privacy values from the peer call leg

  If the incoming INVITE has a Privacy header or a RPID with privacy on, the outgoing INVITE can be set to Privacy: id. This behavior is enabled by configuring **privacy pstn** command globally or **voice-class sip privacy pstn** command on the selected dial-per.

  Incoming INVITE can have multiple privacy header values, id, user, session, and so on. Configure the **privacy-policy passthru** command globally or **voice-class sip privacy-policy passthru** command to transparently pass across these multiple privacy header values.

  Some NGN servers require a Privacy header to be sent even though privacy is not required. In this case the Privacy header must be set to none. The Cisco Unified Border Element can add a privacy header with the value None while forwarding the outgoing INVITE to NGN. Configure the **privacy-policy send-always** globally or **voice-class sip privacy-policy send-always** command in dial-peer to enable this behavior.

If the user is not subscribed to a privacy service, the Cisco Unified Border Element can be configured with no Privacy settings.

P-Called Party Identity

The Cisco Unified Border Element can be configured to use the PCPID header in an incoming INVITE message to route the call, and to use the PCPID value to set the To: value of outgoing INVITE messages.

The PCPID header is part of the INVITE messages sent by the NGN, and is used by Third Generation Partnership Project (3GPP) networks. The Cisco Unified Border Element uses the PCPID from incoming INVITE messages (from the NGN) to route calls to the Cisco Unified Call Manager.

Note The PCPID header supports the use of E.164 numbers only.

P-Associated URI

The Cisco Unified Border Element supports the use of PAURI headers sent as part of the registration process. After the Cisco Unified Border Element sends REGISTER messages using the configured E.164 number, it receives a 200 OK message with one or more PAURIs. The number in the first PAURI (if present) must match the contract number. The Cisco Unified Border Element supports a maximum of six PAURIs for each registration.

Note The Cisco Unified Border Element performs the validation process only when a PAURI is present in the 200 OK response.

The registration validation process works as follows:

- The Cisco Unified Border Element receives a REGISTER response message that includes PAURI headers that include the contract number and up to five secondary numbers.

- The Cisco Unified Border Element validates the contract number against the E.164 number that it is registering:
  - If the values match, the Cisco Unified Border Element completes the registration process and stores the PAURI value. This allows administration tools to view or retrieve the PAURI if needed.
  - If the values do not match, the Cisco Unified Border Element unregisters and then reregisters the contract number. The Cisco Unified Border Element performs this step until the values match.

Random Contact Support

The Cisco Unified Border Element can use random-contact information in REGISTER and INVITE messages so that user information is not revealed in the contact header.

To provide random contact support, the Cisco Unified Border Element performs SIP registration based on the random-contact value. The Cisco Unified Border Element then populates outgoing INVITE requests with the random-contact value and validates the association between the called number and the random value in the Request-URI of the incoming INVITE. The Cisco Unified Border Element routes calls based on the PCPID, instead of the Request-URI which contains the random value used in contact header of the REGISTER message.

The default contact header in REGISTER messages is the calling number. The Cisco Unified Border Element can generate a string of 32 random alphanumeric characters to replace the calling number in the REGISTER contact header. A different random character string is generated for each pilot or contract number being registered. All subsequent registration requests will use the same random character string.

The Cisco Unified Border Element uses the random character string in the contact header for INVITE messages that it forwards to the NGN. The NGN sends INVITE messages to the Cisco Unified Border Element with random-contact information in the Request URI. For example: INVITE sip:FefhH3zIHe9i8ImcGjDD1PEc5XfFy51G@10.12.1.46:5060.

The Cisco Unified Border Element will not use the To: value of the incoming INVITE message to route the call because it might not identify the correct user agent if supplementary services are invoked. Therefore, the Cisco Unified Border Element must use the PCPID to route the call to the Cisco Unified Call Manager. You can configure routing based on the PCPID at global and dial-peer levels.

Configuring P-Header and Random-Contact Support on the Cisco Unified Border Element

To enable random contact support you must configure the Cisco Unified Border Element to support Session Initiation Protocol (SIP) registration with random-contact information, as described in this section.

To enable the Cisco Unified Border Element to use the PCPID header in an incoming INVITE message to route the call, and to use the PCPID value to set the To: value of outgoing INVITE messages, you must configure P-Header support as described in this section.

This section contains the following tasks:

- Configuring P-Header Translation on a Cisco Unified Border Element, page 279
- Configuring P-Header Translation on an Individual Dial Peer, page 281
- Configuring P-Called-Party-Id Support on a Cisco Unified Border Element, page 281
- Configuring P-Called-Party-Id Support on an Individual Dial Peer, page 283
- Configuring Privacy Support on a Cisco Unified Border Element, page 284
- Configuring Privacy Support on an Individual Dial Peer, page 285

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the"Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344

Restrictions

To enable random-contact support, you must configure the Cisco Unified Border Element to support SIP registration with random-contact information. In addition, you must configure random-contact support in VoIP voice-service configuration mode or on the dial peer.

If random-contact support is configured for SIP registration only, the system generates the random-contact information, includes it in the SIP REGISTER message, but does not include it in the SIP INVITE message.

If random-contact support is configured in VoIP voice-service configuration mode or on the dial peer only, no random contact is sent in either the SIP REGISTER or INVITE message.

Configuring P-Header Translation on a Cisco Unified Border Element

To configure P-Header translations on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **asserted-id** *header-type*
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**asserted-id** *header-type* **Example:** `Router(conf-serv-sip)# asserted-id ppi`	Specifies the type of privacy header in the outgoing SIP requests and response messages.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring P-Header Translation on an Individual Dial Peer

To configure P-Header translation on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip asserted-id** *header-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	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip asserted-id** *header-type* **Example:** `Router(config-dial-peer)# voice-class sip asserted-id ppi`	Specifies the type of privacy header in the outgoing SIP requests and response messages, on this dial peer.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring P-Called-Party-Id Support on a Cisco Unified Border Element

To configure P-Called-Party-Id support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. call-route p-called-party-id
6. random-request-uri validate
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**call-route p-called-party-id** **Example:** `Router(conf-serv-sip)# call-route p-called-party-id`	Enables the routing of calls based on the PCPID header.
Step 6	**random-request-uri validate** **Example:** `Router(conf-serv-sip)# random-request-uri validate`	Enables the validation of the random string in the Request URI of the incoming INVITE message.
Step 7	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring P-Called-Party-Id Support on an Individual Dial Peer

To configure P-Called-Party-Id support on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. voice-class sip call-route p-called-party-id
5. voice-class sip random-request-uri validate
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	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip call-route p-called-party-id** **Example:** `Router(config-dial-peer)# voice-class sip call-route p-called-party-id`	Enables the routing of calls based on the PCPID header on this dial peer.
Step 5	**voice-class sip random-request-uri validate** **Example:** `Router(config-dial-peer)# voice-class sip random-request-uri validate`	Enables the validation of the random string in the Request URI of the incoming INVITE message on this dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring Privacy Support on a Cisco Unified Border Element

To configure privacy support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. privacy *privacy-option*
6. privacy-policy *privacy-policy-option*
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**privacy** *privacy-option* **Example:** `Router(conf-serv-sip)# privacy id`	Enables the privacy settings for the header.
Step 6	**privacy-policy** *privacy-policy-option* **Example:** `Router(conf-serv-sip)# privacy-policy passthru`	Specifies the privacy policy to use when passing the privacy header from one SIP leg to the next.
Step 7	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring Privacy Support on an Individual Dial Peer

To configure privacy support on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip privacy** *privacy-option*
5. **voice-class sip privacy-policy** *privacy-policy-option*
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	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip privacy** *privacy-option* **Example:** `Router(config-dial-peer)# voice-class sip privacy id`	Enables the privacy settings for the header on this dial peer.
Step 5	**voice-class sip privacy-policy** *privacy-policy-option* **Example:** `Router(config-dial-peer)# voice-class sip privacy-policy passthru`	Specifies the privacy policy to use when passing the privacy header from one SIP leg to the next, on this dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring Random-Contact Support on a Cisco Unified Border Element

To configure random-contact support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **credentials username** *username* **password** *password* **realm** *domain-name*
5. **registrar ipv4:***destination-address* **random-contact expires** *expiry*
6. **exit**
7. **voice service voip**
8. **sip**
9. **random-contact**
10. **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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**credentials username** *username* **password** *password* **realm** *domain-name* **Example:** `Router(config-sip-ua)# credentials username 123456 password cisco realm cisco`	Sends a SIP registration message from the Cisco Unified Border Element.

	Command or Action	Purpose
Step 5	**registrar ipv4:***destination-address* **random-contact expires** *expiry* **Example:** `Router(config-sip-ua)# registrar ipv4:10.1.2.2 random-contact expires 200`	Enables the SIP gateways to register E.164 numbers on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and Skinny Client Control Protocol (SCCP) phones with an external SIP proxy or SIP registrar. • The **random-contact** keyword configures the Cisco Unified Border Element to send the random string from the REGISTER message to the registrar.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 7	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 8	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 9	**random-contact** **Example:** `Router(conf-serv-sip)# random-contact`	Enables random-contact support on a Cisco Unified Border Element.
Step 10	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring Random-Contact Support for an Individual Dial Peer

To configure random-contact support for an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **credentials username** *username* **password** *password* **realm** *domain-name*
5. **registrar ipv4:***destination-address* **random-contact expires** *expiry*
6. **exit**
7. **dial-peer voice** *tag* **voip**
8. **voice-class sip random-contact**
9. **exit**

DETAILED STEPS

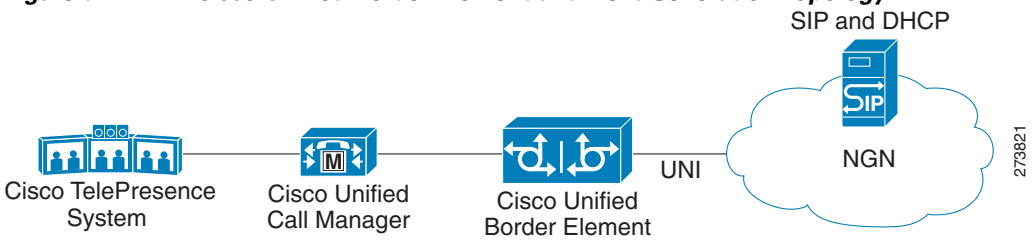
	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**credentials username** *username* **password** *password* **realm** *domain-name* **Example:** `Router(config-sip-ua)# credentials username 123456 password cisco realm cisco`	Sends a SIP registration message from the Cisco Unified Border Element.
Step 5	**registrar ipv4:***destination-address* **random-contact expires** *expiry* **Example:** `Router(config-sip-ua)# registrar ipv4:10.1.2.2 random-contact expires 200`	Enables the SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar. • The **random-contact** keyword configures the Cisco Unified Border Element to send the random string from the REGISTER message to the registrar.

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 7	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 8	**voice-class sip random-contact** **Example:** `Router(config-dial-peer)# voice-class sip random-contact`	Enables random-contact support on this dial peer.
Step 9	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Support for Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information

Figure 8 shows a typical network topology where the Cisco Unified Border Element is configured to route messages between a call manger system (Cisco Unified Call Manager) and a Next-Generation-Network (NGN).

Figure 8 Cisco Unified Border Element and Next Generation Topology



The Cisco Unified Border Element supports the use of preloaded routes for dialog initiating INVITE requests. The system routes INVITE messages based on the information received in REGISTER response, such as information in the Service-Route header.

The Cisco Unified Border Element sends REGISTER messages containing the Supported: path header to the NGN server. The NGN server may accept the REGISTER request sent by the Cisco Unified Border Element and send a 200 Ok response. Apart from the routine information the 200 Ok response may include the Service-Route header. The value of the Service-Router header can be an IP address or Fully Qualified Domain Name (FQDN).

Depending on the configuration you specify, the Cisco Unified Border Element can send the information of Service-Route header and SIP server values in the Route header of outgoing INVITE messages. The **preloaded-route** command can be used to configure the content of Route: header in outgoing INVITE messages.

If the Cisco Unified Border Element is configured to include Service-Route information only, then the Route: header in the outgoing INVITE message contains the Service-Route value from the Service-Route header of the 200 OK response for REGISTER request.

If the Cisco Unified Border Element is configured to include Service-Route and SIP server information, then the Route: header in the outgoing INVITE message contains the Service-Route and SIP server values. The Service-Route values are taken from the Service-Route header of the 200 OK Register message. The SIP server information, is taken from the outbound-proxy if present, else it is taken from Session target.

If the Cisco Unified Border Element is configured to include Service-Route and SIP server information, but no Service-Route is received in the 200 OK Register response, then the Route: header in the outgoing INVITE message contains the SIP server value only.

If the Cisco Unified Border Element receives a response message other than 100 that includes the Record-Route header, then it adds the Record-Route value to the Route: header for subsequent requests in the same dialog.

The INVITE message also contains random-contact user information in the Request-Line URI. Therefore, the Cisco Unified Border Element can use the P-Called Party Identify value to route the call to Cisco Unified Call Manager.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the"Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344

Configuring Support for SIP UPDATE Message per RFC 3311

The Support for SIP UPDATE Message per RFC 3311 feature provides Session Description Protocol (SDP) support for Session Initiation Protocol (SIP)-to-SIP calls. The SIP Service Provider Interface (SPI) is modified to support the following media changes using the UPDATE message:

- Early dialog SIP-to-SIP media changes.
- Mid dialog SIP-to-SIP media changes.

The Support for SIP UPDATE Message per RFC 3311 feature is enabled by default on the Cisco Unified Border Element (UBE) and no configuration is required.

Prerequisites

- At least one offer or answer negotiation must be completed for Cisco UBE to handle the UPDATE message with SDP.
- An early dialog UPDATE message with SDP is processed only when both endpoints support the UPDATE message.

Restrictions

- An UPDATE message with SDP is not supported for SIP-to-H323 calls.
- An UPDATE message with SDP with a fully qualified domain name (FQDN) is not supported.

- Contact information in the UPDATE message is not supported.
- A retransmitted UPDATE message with SDP is ignored by the SIP stack. No response is sent for retransmitted UPDATE messages.

Configuring Preloaded Route Support on the Cisco Unified Border Element

To configure preloaded route support on the Cisco Unified Border Element by enabling support for the Service-Route values in the Route header of outgoing INVITE message, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **preloaded-route** [**sip-server**] **service-route**
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters SIP configuration mode.

	Command or Action	Purpose
Step 5	**preloaded-route** [**sip-server**] **service-route** **Example:** `Router(conf-serv-sip)# preloaded-route sip-server service-route`	Configures the system to include the SIP server and Service-Route information in the Route header of outgoing INVITE message.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring Preloaded Route Support on the Cisco Unified Border Element on an Individual Dial Peer

To configure preloaded route support for an individual dial peer on the Cisco Unified Border Element, by enabling support for the Service-Route in the Route header of outgoing INVITE message, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip preloaded-route** [**sip-server**] **service-route**
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 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.

	Command or Action	Purpose
Step 4	**voice-class sip preloaded-route** [**sip-server**] **service-route** **Example:** `Router(config-dial-peer)# voice-class sip preloaded-route sip-server service-route`	Enables preloaded route support for SIP calls for this dial-peer, and enables the system to add SIP server and Service-Route information to the Route header in outgoing INVITE messages.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers

The Cisco Unified Border Element supports the construction of request URIs in tel: format. The system supports this format for both the To: header and the Request-Line. The system also supports appending the phone-context parameter to the tel: URL.

Phone-context

If the system is configured to use the tel: URL format in the Request-Line or the To: header, then the phone-context is appended to the tel: URL.

The system populates the phone-context parameter with the session target hostname and domain. The system identifies the session target hostname and domain in one of the following ways:

- The session target hostname and domain is manually configured using the **session target** command at the dial-peer level.
- The session target DHCP is configured and the system dynamically retrieves the values from the DHCP server.

The system must populate the phone-context parameter with a domain name. Therefore, if the configured session target is an IP address, the system does not append a phone-context parameter to the tel: URL.

Request-Line URIs

The Cisco Unified Call Manager uses the sip: format in the Request-Line URIs when it sends INVITE messages to the Cisco Unified Border Element server. However, some servers require the tel: format in Request-Line URIs. Therefore, the Cisco Unified Border Element must use the tel: format in the Request-Line URI of INVITE messages sent to the these servers. The tel: format must include the phone-context value when applicable.

Some servers use the sip: format in the Request-Line URIs of the INVITE messages that it sends to the Cisco Unified Border Element. The Cisco Unified Call Manager also supports the use of the sip: format.

To: Header

The Cisco Unified Call Manager uses sip: format in the To: header, when it sends INVITE messages to the Cisco Unified Border Element. However, some servers require the tel: format in the To: headers. Therefore, the Cisco Unified Border Element must use the tel: format in the To: header of INVITE messages sent to these servers. The tel: format must include the phone-context value.

Some servers require the tel: format in the To: header in the INVITE messages that it sends to the Cisco Unified Border Element. However, the Cisco Unified Call Manager supports the use of the sip: format. Therefore, the Cisco Unified Border Element must use the sip: format in the To: header of INVITE messages sent to the Cisco Unified Call Manager.

Note Some servers requires the tel: format in the To: header only for the initial INVITE. The regular dialog processing rules apply for header construction for the subsequent requests.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the"Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344

Configuring tel: URL Formats and Phone-Context Parameter

The tasks in this section describe how to send URIs in the Request-Line and the To: header as telephone (TEL) URIs and how to include the phone-context parameter in the headers, at both a system level and on an individual dial peer.

This section contains the following tasks:

Configuring tel: URI Formats and Phone-Context Parameter on Individual SIP Headers

To enable the URIs in the Request-Line and the To: header to be sent as telephone (TEL) URIs and to include the phone-context parameter in the headers, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **url tel phone-context**
6. **tel-config to-hdr phone-context**
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**url tel phone-context** **Example:** `Router(conf-serv-sip)# url tel phone-context`	Configures the Request-Line URI to tel: format.
Step 6	**tel-config to-hdr phone-context** **Example:** `Router(conf-serv-sip)# tel-config to-hdr phone-context`	Configures the To: header Request-URI to tel: format.
Step 7	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring tel: URI Formats and Phone-Context Parameter on Individual SIP Headers on an Individual Dial Peer

To enable the URIs in the Request-Line and the To: header to be sent as telephone (TEL) URIs on an individual dial peer and to include the phone-context parameter in the headers, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip url tel phone-context**
5. voice-class sip tel-config to-hdr phone-context
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	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip url tel phone-context** **Example:** `Router(config-dial-peer)# voice-class sip url tel phone-context`	Configures the Request-Line URI to tel: format and appends the phone-context parameter to the header, on the initial outgoing INVITE associated with this dial peer.
Step 5	**voice-class sip tel-config to-hdr phone-context** **Example:** `Router(config-dial-peer)# voice-class sip tel-config to-hdr phone-context`	Configures the To: header Request-URI to tel: format and appends the phone-context parameter to the header, on the initial outgoing INVITE associated with this dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring tel: URI Formats on the To: Header

To enable the URIs in the To: header to be sent as telephone (TEL) URIs, without including the phone-context parameter in the header, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **tel-config to-hdr**
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	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**tel-config to-hdr** **Example:** `Router(conf-serv-sip)# tel-config to-hdr`	Configures the To: header Request-URI to tel: format.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring tel: URI Formats on the To: Header on an Individual Dial Peer

To enable the URIs in the To: header to be sent as telephone (TEL) URIs, without including the phone-context parameter in the header, on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. voice-class sip tel-config to-hdr
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 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • No phone-context parameter is appended here.
Step 4	**voice-class sip tel-config to-hdr** **Example:** `Router(config-dial-peer)# voice-class sip tel-config to-hdr`	Configures the To: header Request-URI to tel: format on the initial outgoing INVITE associated with this dial peer.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configuring Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Element

This feature adds support on Cisco UBE for selective filtering of outgoing provisional responses, including 180 - Alerting, and 183-Session In Progress responses. Selective filtering can be further based on the availability of media information in the received provisional response.

Next Generation Network (NGN) restricts the UNI from sending 183 response with SDP towards the NGN network. Cisco Unified CM always sends 183 response with SDP responses. It is necessary for the Cisco UBE to block these responses to allow Cisco Unified CM to interwork within the Next Generation network.

Prerequisites

To enable this feature, you must have Cisco IOS Release 12.4(22)YB or a later release installed and running on your Cisco gateway. For detailed information on platform availability and subsequent releases, see the"Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element" section on page 344

Restrictions

Blocking 180 and183 responses with or without SDP requirement is to block 183 with SDP only.

Configuring Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Element

To enable Selective Filtering of Outgoing Provisional Response on the Cisco UBE perform the steps in this section. This section contains the following subsections:

- Configuring Cisco UBE for Unsupported Content Pass-through at the Global Level, page 198
- Configuring Cisco UBE for Unsupported Content Pass-through at the Dial Peer Level, page 199

Configuring Selective Filtering of Outgoing Provisional Response on the Cisco UBE at the Global Level

To configure Selective Filtering of Outgoing Provisional Response on the Cisco UBE at the global level, perform the steps in this section:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **block 183 sdp absent**
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	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**block 183 sdp absent** **Example:** `Router(conf-serv-sip)# block 183 sdp absent}`	Filters outgoing provisional responses, including 180 - Alerting, and 183-Session In Progress responses.
Step 6	**exit** **Example:** `Router(conf-voi-serv)# exit`	Exits the current mode.

Configuring Selective Filtering of Outgoing Provisional Response on the Cisco UBE at the Dial Peer Level

To configure Selective Filtering of Outgoing Provisional Response on the Cisco UBE at the dial-peer level, configure the outgoing dial-peer as follows the steps in this section:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **voice-class sip block 183 sdp present**
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 22 voip`	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	**voice-class sip block 183 sdp present** **Example:** `Router (conf-dial-peer)# voice-class sip block 183 sdp present`	Filters outgoing provisional responses, including 180 - Alerting, and 183-Session In Progress responses.
Step 5	**exit** **Example:** `Router(conf-voi-serv)# exit`	Exits the current mode.

Configuring Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element

The Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature supports negotiation of an audio codec using the Voice Class Codec and Codec Transparent infrastructure on the Cisco Unified Border Element (Cisco UBE).

Benefits

Following are the benefits of the Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature:

- You can configure dissimilar Voice Class Codec configurations on the incoming and outgoing dial peers.
- Both normal transcoding and high-density transcoding are supported with the Voice Class Codec configuration.
- Mid-call codec changes for supplementary services are supported with the Voice Class Codec configuration. Transcoder resources are dynamically inserted or deleted when required.
- Reinvite-based supplementary services invoked from the Cisco Unified Communications Manager (CUCM), like call hold, call resume, music on hold (MOH), call transfer, and call forward are supported with the Voice Class Codec configuration.

- T.38 fax and fax passthru switchover with Voice Class Codec configuration are supported.
- Reinvite-based call hold and call resume for Secure Real-Time Transfer protocol (SRTP) and Real-Time Protocol (RTP) interworking on Cisco UBE are supported with the Voice Class Codec configuration.

Prerequisites

To the configure Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature you must know the following:

- Transcoding configuration on the Cisco UBE.
- The digital signal processor (DSP) requirements to support the transcoding feature on the Cisco UBE.
- The existing Voice Class Codec configuration on the dial peers.

Restrictions

The Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature has the following limitations:

- Mid-call insertion or deletion of the transcoder with voice class codec for H323-H323 and H323-SIP is not supported.
- Voice class codec is not supported for video calls.

Disabling Codec Filtering

Cisco UBE is configured to filter common codecs for the subsets, by default. The filtered codecs are sent in the outgoing offer. You can configure the Cisco UBE to offer all the codecs configured on an outbound leg instead of offering only the filtered codecs.

Note This configuration is applicable only for early offer calls from the Cisco UBE. For delayed offer calls, by default all codecs are offered irrespective of this configuration.

Perform this task to disable codec filtering and allow all the codecs configured on an outbound leg.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class codec** *tag* [**offer-all**]
5. **end**

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 10 voip`	Enters dial peer voice configuration mode.
Step 4	**voice-class codec** *tag* [**offer-all**] **Example:** `Router(config-dial-peer)# voice-class codec 10 offer-all`	Adds all the configured voice class codec to the outgoing offer from the Cisco UBE.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	Exits the dial peer voice configuration mode.

Troubleshooting Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element

Use the following commands to debug any errors that you may encounter when you configure the Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature:

- **debug ccsip all**
- **debug voip ccapi inout**
- **debug sccp messages**
- **debug voip rtp session**

Verifying Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element

Perform this task to display information to verify Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element configuration. These **show** commands need not be entered in any specific order.

SUMMARY STEPS

1. **enable**

2. **show call active voice brief**
3. **show voip rtp connections**
4. **show sccp connections**
5. **show dspfarm dsp active**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

Step 2 **show call active voice brief**

Displays a truncated version of call information for voice calls in progress.

```
Router# show call active voice brief

<ID>: <CallID> <start>ms.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>

 media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>

 long duration call detected:<y/n> long duration call duration :<sec> timestamp:<time>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
   last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l>
i/o:<l>/<l> dBm
  MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
         speeds(bps): local <rx>/<tx> remote <rx>/<tx>
 Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
 bw: <req>/<act> codec: <audio>/<video>
  tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
 rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>


Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4
1243 : 11 971490ms.1 +-1 pid:1 Answer 1230000 connecting
 dur 00:00:00 tx:415/66400 rx:17/2561
 IP 192.0.2.1:19304 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw
TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a

1243 : 12 971500ms.1 +-1 pid:2 Originate 3210000 connected
 dur 00:00:00 tx:5/10 rx:4/8
 IP 9.44.26.4:16512 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729br8 TextRelay:
off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
```

```
0    : 13 971560ms.1 +0 pid:0 Originate  connecting
 dur 00:00:08 tx:415/66400 rx:17/2561
 IP 192.0.2.2:2000 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay:
off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a

0    : 15 971570ms.1 +0 pid:0 Originate  connecting
 dur 00:00:08 tx:5/10 rx:3/6
 IP 192.0.2.3:2000 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729br8 TextRelay:
off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a


Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4
```

Step 3 **show voip rtp connections**

Displays Real-Time Transport Protocol (RTP) connections.

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP     LocalIP
RemoteIP
1     11         12         16662    19304    192.0.2.1
192.0.2.2
2     12         11         17404    16512    192.0.2.2
192.0.2.3
3     13         14         18422    2000     192.0.2.4
9.44.26.3
4     15         14         16576    2000     192.0.2.6
192.0.2.5
Found 4 active RTP connections
```

Step 4 **show sccp connections**

Displays information about the connections controlled by the Skinny Client Control Protocol (SCCP) transcoding and conferencing applications.

```
Router# show sccp connections

sess_id    conn_id      stype mode     codec   sport rport ripaddr

5          5            xcode sendrecv g729b   16576 2000  192.0.2.3
5          6            xcode sendrecv g711u   18422 2000  192.0.2.4

Total number of active session(s) 1, and connection(s) 2
```

Step 5 **show dspfarm dsp active**

Displays active DSP information about the DSP farm service.

```
Router# show dspfarm dsp active

SLOT DSP VERSION  STATUS CHNL USE   TYPE    RSC_ID BRIDGE_ID PKTS_TXED PKTS_RXED
```

```
0    1   27.0.201 UP     1    USED  xcode   1      0x9         5          8
0    1   27.0.201 UP     1    USED  xcode   1      0x8         2558       17

Total number of DSPFARM DSP channel(s) 1
```

Configuring Support for SIP Registration Proxy on Cisco UBE

The Support for SIP Registration Proxy on Cisco UBE feature provides support for sending outbound registrations from Cisco Unified Border Element (UBE) based on incoming registrations. This feature enables direct registration of Session Initiation Protocol (SIP) endpoints with the SIP registrar in hosted unified communication (UC) deployments. This feature also provides various benefits for handling Cisco UBE deployments with no IP private branch exchange (PBX) support.

In certain Cisco UBE deployments, managed services are offered without an IPPBX installed locally at the branch office. A PBX located at the service provider (SP) offers managed services to IP phones. A Cisco UBE device located at the branch office provides address translation services. However, the registration back-to-back functionality is required to get the phone registered, so that calls can be routed to the branch or the phones.

In such deployment scenarios, enabling the Support for SIP Registration Proxy on Cisco UBE feature provides the following benefits:

- Support for back-to-back user agent (B2BUA) functionality.
- Options to configure rate-limiting values such as expiry time, fail-count value, and a list of registrars to be used for the registration.
- Registration overload protection facility.
- Option to route calls to the registering endpoint (user or phone).
- Option to send the 401 or 407 message to request for user credentials (this process is known as challenge) from an incoming registration.

Registration Pass-Through Modes

Cisco UBE uses the following two modes for registration pass-through:

- End-to-End Mode, page 306
- Peer-to-Peer Mode, page 308

End-to-End Mode

In the end-to-end mode, Cisco UBE collects the registrar details from the Uniform Resource Identifier (URI) and passes the registration messages to the registrar. The registration information contains the expiry time for rate-limiting, the challenge information from the registrar, and the challenge response from the user.

Cisco UBE also passes the challenge to the user if the register request is challenged by the registrar. The registrar sends the 401 or 407 message to the user requesting for user credentials. This process is known as challenge.

Cisco UBE ignores the local registrar and authentication configuration in the end-to-end mode. It passes the authorization headers to the registrar without the header configuration.

End-to-End Mode—Call Flows

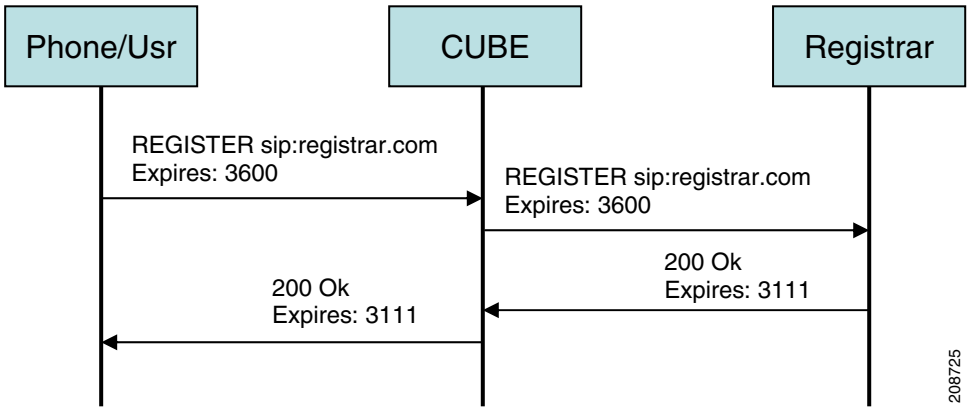
This section explains the following end-to-end pass-through mode call flows:

- Register Success Scenario, page 307
- Registrar Challenging the Register Request Scenario, page 307

Register Success Scenario

Figure 9 shows an end-to-end registration pass-through scenario where the registration request is successful.

Figure 9 End-to-End Registration Pass-through Mode—Register Success Scenario



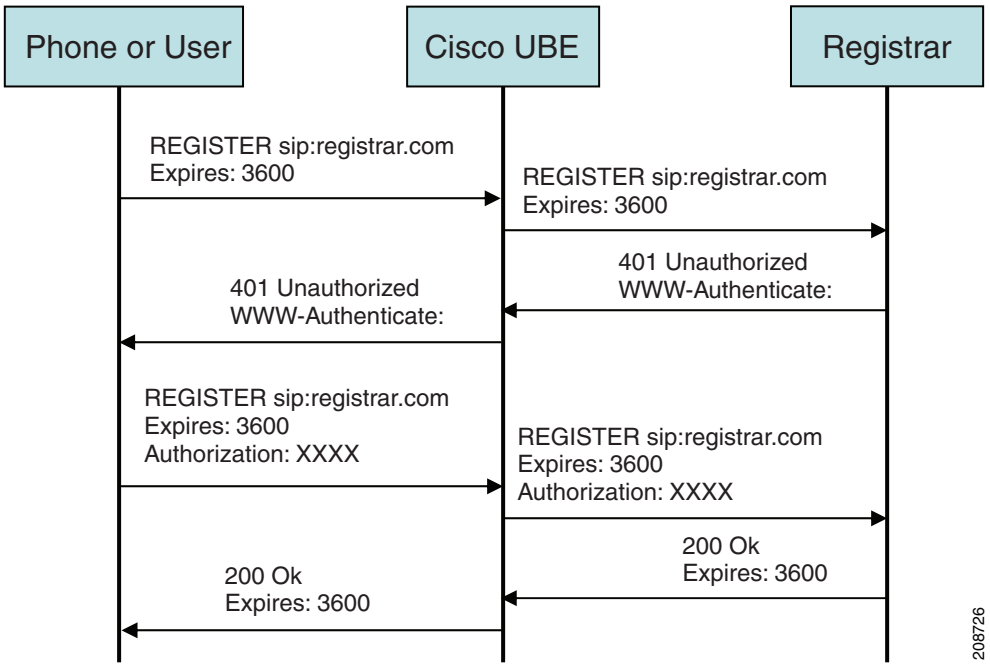
The register success scenario for the end-to end registration pass-through mode is as follows:

1. The user sends the register request to Cisco UBE.
2. Cisco UBE matches the request with a dial peer and forwards the request to the registrar.
3. Cisco UBE receives a success response message (200 OK message) from the registrar and forwards the message to the endpoint (user).
4. The registrar details and expiry value are passed to the user.

Registrar Challenging the Register Request Scenario

Figure 10 shows an end-to end registration pass-through scenario where the registrar challenges the register request.

Figure 10 End-to-End Registration Pass-through Mode—Registrar Challenging the Register Request Scenario



The following scenario explains how the registrar challenges the register request:

1. The user sends the register request to Cisco UBE.
2. Cisco UBE matches the register request with a dial peer and forwards it to the registrar.
3. The registrar challenges the register request.
4. Cisco UBE passes the registrar response and the challenge request, only if the registrar challenges the request to the user.
5. The user sends the register request and the challenge response to the Cisco UBE.
6. Cisco UBE forwards the response to the registrar.
7. Cisco UBE receives success message (200 OK message) from the registrar and forwards it to the user.

Peer-to-Peer Mode

In the peer-to-peer registration pass-through mode, the outgoing register request uses the registrar details from the local Cisco UBE configuration. Cisco UBE answers the challenges received from the registrar using the configurable authentication information. Cisco UBE can also challenge the incoming register requests and authenticate the requests before forwarding them to the network.

In this mode, Cisco UBE sends a register request to the registrar and also handles register request challenges. That is, if the registration request is challenged by the registrar (registrar sends 401 or 407 message), Cisco UBE forwards the challenge to the user and then passes the challenge response sent by the user to the registrar. In the peer-to-peer mode, Cisco UBE can use the **authentication** command to calculate the authorization header and then challenge the user depending on the configuration.

Note The **registrar** command must be configured in peer-to-peer mode. Otherwise, the register request is rejected with the 503 response message.

Peer-to-Peer Mode—Call Flows

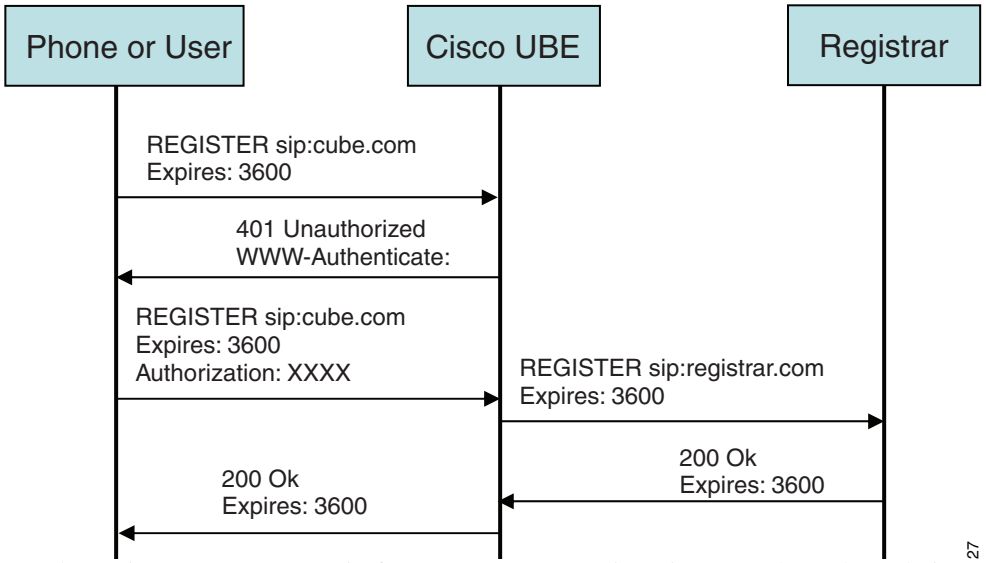
This section explains the following peer-to-peer pass-through mode call flows:

- Register Success Scenario, page 309
- Registrar Challenging the Register Request Scenario, page 309

Register Success Scenario

Figure 11 shows a peer-to-peer registration pass-through scenario where the registration request is successful.

Figure 11 Peer-to-Peer Registration Pass-through Mode—Register Success Scenario



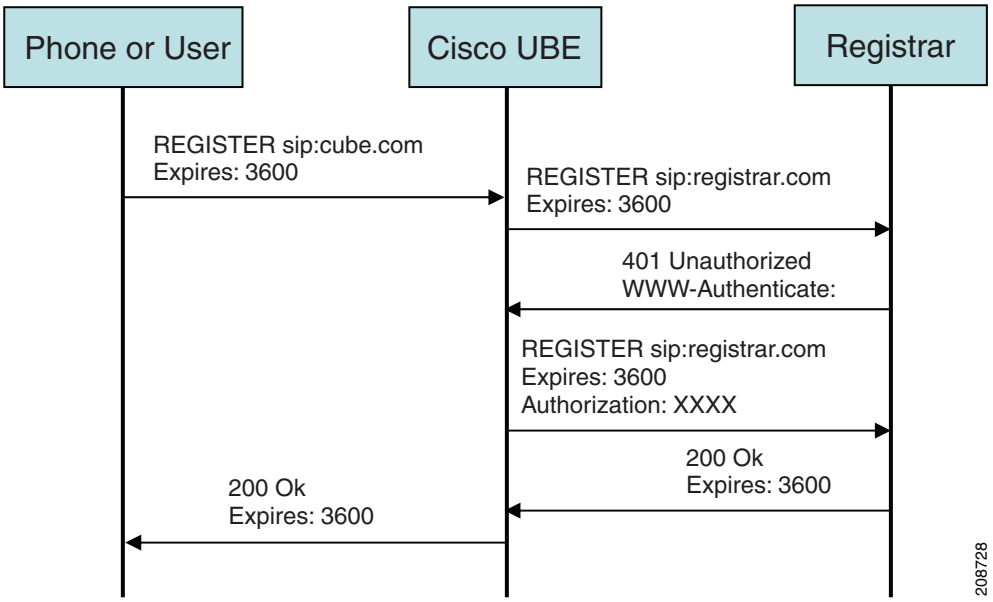
The register success scenario for a peer-to-peer registration pass-through mode is as follows:

1. The user sends the register request to Cisco UBE.
2. Cisco UBE matches the register request with a dial peer and forwards the register request to the registrar.
3. Cisco UBE receives a success message (200 OK message) from the registrar and forwards it to the endpoint (user). The following functions are performed:
   - Cisco UBE picks up the details about the registrar from the configuration.
   - Cisco UBE passes the registrar details and expiry value to the user.

Registrar Challenging the Register Request Scenario

Figure 12 shows a peer-to-peer registration pass-through scenario where the registration request is challenged by the registrar.

Figure 12 Peer-to-Peer Registration Pass-through Mode—Registrar Challenging the Register Request Scenario



The following scenario explains how the registrar challenges the register request:

1. The user sends the register request to Cisco UBE.
2. Cisco UBE matches the register request with a dial peer and forwards the register request to the registrar.
3. The user responds to the challenge request.
4. Cisco UBE validates the challenge response and forwards the register request to the registrar.
5. Cisco UBE receives a success message from the registrar and forwards it to the endpoint (user).

Note You can configure Cisco UBE to challenge the register request and validate the challenge response.

Registration in Different Registrar Modes

This section explains SIP registration pass-through in the following registrar modes:

- Primary-Secondary Mode, page 310
- DHCP Mode, page 311
- Multiple Register Mode, page 311

Primary-Secondary Mode

In the primary-secondary mode the register message is sent to both the primary and the secondary registrar servers simultaneously.

The register message is processed as follows:

- The first successful response is passed to the phone as a SUCCESS message.
- All challenges to the request are handled by Cisco UBE.

- If the final response received from the primary and the secondary servers is an error response, the error response that arrives later from the primary or the secondary server is passed to the phone.
- If only one registrar is configured, a direct mapping is performed between the primary and the secondary server.
- If no registrar is configured, or if there is a Domain Name System (DNS) failure, the "503 service not available" message is sent to the phone.

DHCP Mode

In the DHCP mode the register message is sent to the registrar server using DHCP.

Multiple Register Mode

In the multiple register mode, you can configure a dial peer to select and enable the indexed registrars. Register messages must be sent only to the specified index registrars.

The response from the registrar is mapped the same way as in the primary-secondary mode. See the "Primary-Secondary Mode" section on page 310.

Registration Overload Protection

The registration overload protection functionality enables Cisco UBE to reject the registration requests that exceed the configured threshold value.

To support the registration overload protection functionality, Cisco UBE maintains a global counter to count all the pending outgoing registrations and prevents the overload of the registration requests as follows:

- The registration count is decremented if the registration transaction is terminated.
- The outgoing registrations are rejected if the count goes beyond a configured threshold.
- The incoming register request is rejected with the 503 response if the outgoing registration is activated by the incoming register request.
- A retry timer set for a random value is used for attempting the registration again if the registrations are originated from Cisco UBE or a gateway.

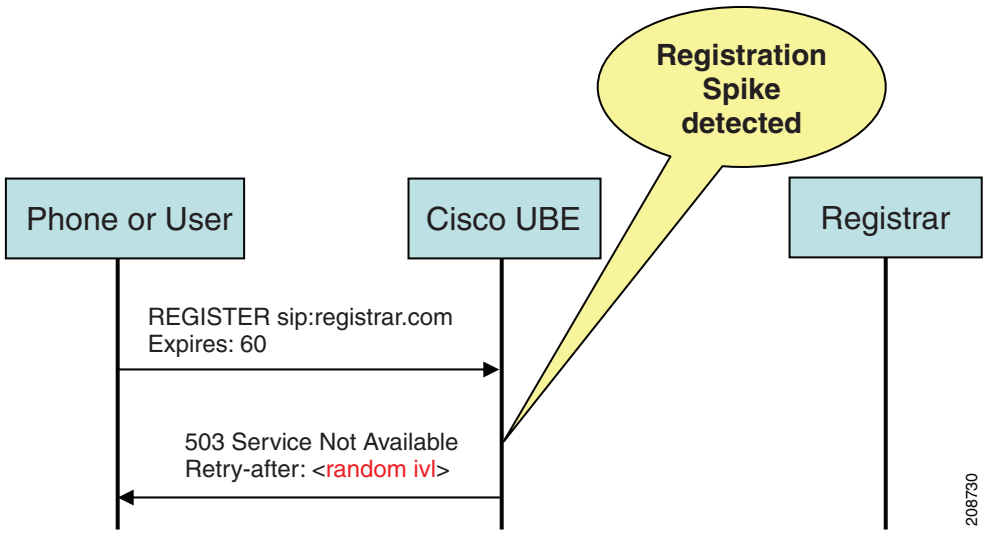
The registration overload protection functionality protects the network from the following:

- Avalanche Restart—All the devices in the network restart at the same time.
- Component Failures—Sudden burst of load is routed through the device due to a device failure.

Registration Overload Protection—Call Flow

Figure 13 shows the call flow when the register overload protection functionality is configured on Cisco UBE:

Figure 13 Register Overload Protection



The following steps explain the register overload protection scenario:

1. The user sends a register request to Cisco UBE.
2. Cisco UBE matches the request with a dial peer and forwards the register request to the registrar.
3. The registration is rejected with a random retry value when the registration threshold value is reached.

Note The call flow for the DNS query on the Out Leg is the same for the end-to-end and peer-to-peer mode.

Registration Rate-limiting

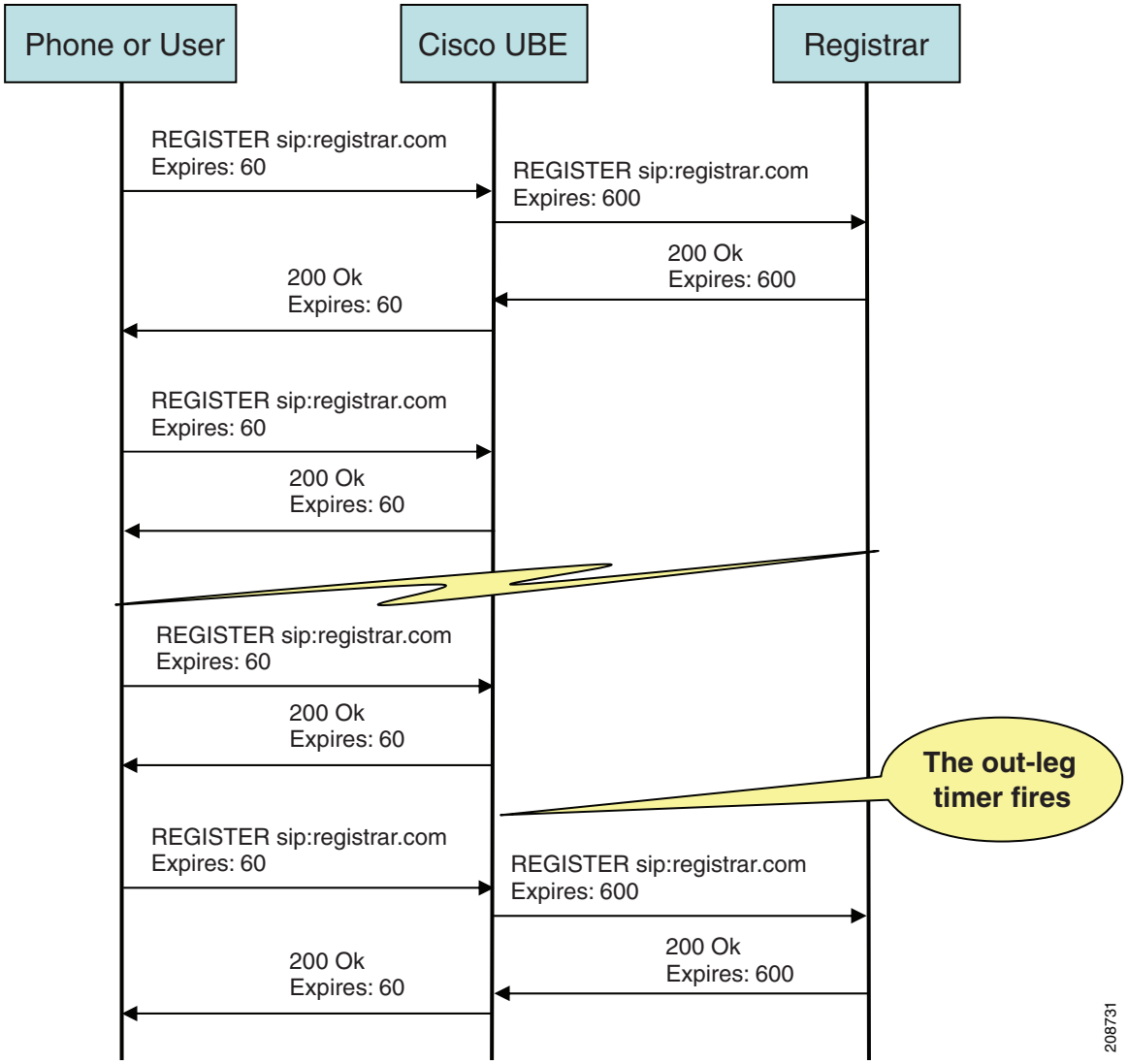
The registration rate-limiting functionality enables you to configure different SIP registration pass-through rate-limiting options. The rate-limiting options include setting the expiry time and the fail count value for a Cisco UBE. You can configure the expiry time to reduce the load on the registrar and the network. Cisco UBE limits the reregistration rate by maintaining two different timers—in-registration timer and out-registration timer.

The initial registration is triggered based on the incoming register request. The expiry value for the outgoing register is selected based on the Cisco UBE configuration. On receiving the 200 OK message (response to the BYE message) from the registrar, a timer is started using the expiry value available in the 200 OK message. The timer value in the 200 OK message is called the out-registration timer. The success response is forwarded to the user. The expiry value is taken from the register request and the timer is started accordingly. This timer is called the in-registration timer. There must be a significant difference between the in-registration timer and the out-registration timer values for effective rate-limiting.

Registration Rate-limiting Success—Call Flow

Figure 14 shows the call flow when the rate-limiting functionality is successful:

Figure 14 Rate-limiting Success Scenario



The following steps explain a scenario where the rate-limiting functionality is successful:

1. The user sends the register request to Cisco UBE.
2. Cisco UBE matches the registration request with a dial peer and forwards it to the registrar. The outgoing register request contains the maximum expiry value if the rate-limiting functionality is configured.
3. The registrar accepts the registration.
4. Cisco UBE forwards the success response with the proposed expiry timer value.
5. The user sends the reregistration requests based on the negotiated value. Cisco UBE resends the register requests until the out-leg expiry timer value is sent.
6. Cisco UBE forwards the subsequent register request to the registrar, if the reregister request is received after the out-leg timer is reached.

Prerequisites

- You must enable the local SIP registrar. See "Enabling Local SIP Registrar" section on page 314.
- You must configure dial peers manually for call routing and pattern matching.

Restrictions

IPv6 support is not provided.

Configuring Support for SIP Registration Proxy on Cisco UBE

Enabling Local SIP Registrar

Perform this task to enable the local SIP registrar.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registrar server** [**expires** [**max** *value*] [**min** *value*]]
6. **end**

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 voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters service SIP configuration mode.
Step 5	**registrar server** [**expires** [**max** *value*] [**min** *value*]] **Example:** `Router(conf-serv-sip)# registrar server`	Enables the local SIP registrar. • Optionally you can configure the expiry time of the registrar using the following keywords: – **expires**—Configures the registration expiry time. – **max**—Configures the maximum registration expiry time. – **min**—Configures the minimum registration expiry time. **Note** The **registrar** command must be configured in peer-to-peer mode. Otherwise, the register request is rejected with the 503 response message.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Exits service SIP configuration mode and returns to privileged EXEC mode.

Configuring SIP Registration at the Global Level

Perform this task to configure the support for the SIP registration proxy on the Cisco UBE at the global level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registration passthrough** [**static**] [**rate-limit** [**expires** *value*] [**fail-count** *value*]] [**registrar-index** [*index*]]
6. **end**

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 voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters service SIP configuration mode.
Step 5	**registration passthrough** [**static**] [**rate-limit** [**expires** *value*] [**fail-count** *value*]] [**registrar-index** [*index*]] **Example:** `Router(conf-serv-sip)# registration passthrough`	Configures the SIP registration pass-through options. • You can specify different SIP registration pass-through options using the following keywords: – **rate-limit**—Enables rate-limiting. – **expires**—Configures expiry value for rate-limiting. – **fail-count**—Configures fail count during rate-limiting. – **registrar-index**—Configures a list of registrars to be used for registration.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Exits service SIP configuration mode and returns to privileged EXEC mode.

Configuring SIP Registration at the Dial Peer Level

Perform this task to configure SIP registration at the dial peer level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
4. **voice-class sip registration passthrough static** [**rate-limit** [**expires** *value*] [**fail-count** *value*] [**registrar-index** [*index*]] | **registrar-index** [*index*]]

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* {**pots** | **voatm** | **vofr** | **voip**} **Example:** `Router(config)# dial-peer voice 444 voip`	Enters dial peer voice configuration mode.
Step 4	**voice-class sip registration passthrough static** [**rate-limit** [**expires** *value*] [**fail-count** *value*] [**registrar-index** [*index*]] | **registrar-index** [*index*]] **Example:** `Router(config-dial-peer)# voice-class sip registration passthrough static`	Configure SIP registration pass-through options on a dial peer on a dial peer. • You can specify different SIP registration pass-through options using the following keywords: – **rate-limit**—Enables rate-limiting. – **expires**—Configures expiry value for rate-limiting. – **fail-count**—Configures fail count during rate-limiting. – **registrar-index**—Configures a list of registrars to be used for registration.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode and returns to global configuration mode.

Configuring Registration Overload Protection Functionality

Perform this task to configure registration overload protection functionality on Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registration spike** *max-number*
5. **end**

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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**registration spike** *max-number* **Example:** `Router(config-sip-ua)# registration spike 100`	Configures registration overload protection functionality on Cisco UBE.
Step 5	**end** **Example:** `Router(config-sip-ua)# end`	Exits SIP user-agent configuration mode and returns to privileged EXEC mode.

Configuring Cisco UBE to Route a Call to the Registrar Endpoint

Perform this task to configure Cisco UBE to route a call to the registrar endpoint.

Note You must perform this configuration on a dial peer that is pointing towards the endpoint.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
4. **session target registrar**
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* {**pots** | **voatm** | **vofr** | **voip**} **Example:** `Router(config)# dial-peer voice 444 voip`	Enters dial peer voice configuration mode.
Step 4	**session target registrar** **Example:** `Router(config-dial-peer)# session target registrar`	Configures Cisco UBE to route the call to the registrar endpoint.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode and returns to global configuration mode.

Configuring Cisco UBE to Challenge Incoming Requests

Perform this task to configure Cisco UBE to challenge incoming requests.

You can configure Cisco UBE to challenge an incoming request. That is, you can configure Cisco UBE to send the 401 or 407 message to the caller requesting for credentials. Based on the information received, Cisco UBE authenticates the request. The configuration also enables Cisco UBE to pass the credentials provided by the user to the registrar if the registrar has challenged the request.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
4. **authentication username** *username* **password** [**0** | **7**] *password* [**realm** *realm* [**challenge**]]
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* {**pots** | **voatm** | **vofr** | **voip**} **Example:** `Router(config)# dial-peer voice 444 voip`	Enters dial peer voice configuration mode.
Step 4	**authentication username** *username* **password** [**0** | **7**] *password* [**realm** *realm* [**challenge**]] **Example:** `Router(config-dial-peer)# authentication username user1 password 7 password1 realm MyRealm.example.com challenge`	Configures Cisco UBE to challenge the incoming registration request.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode and returns to global configuration mode.

Verifying the SIP Registration on Cisco UBE

Perform this task to verify the configuration for SIP registration on Cisco UBE. The **show** commands need not be entered in any specific order.

SUMMARY STEPS

1. **enable**
2. **show sip-ua registration passthrough status**
3. **show sip-ua registration passthrough status detail**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

```
Router> enable
```

Step 2 **show sip-ua registration passthrough status**

Displays the SIP user agent (UA) registration pass-through status information.

```
Router# show sip-ua registration passthrough status

CallId       Line         peer         mode In-Exp       reg-I Out-Exp

============ ============ ============ ==== ============ ===== ============

771          5500550055   1             p2p  64           1     64

================================================================================
```

Step 3 **show sip-ua registration passthrough status detail**

Displays the SIP UA registration pass-through status information in detail.

```
Router# show sip-ua registration passthrough status detail


============================================================

Configured Reg Spike Value: 0

Number of Pending Registrations: 0

============================================================

Call-Id: 763

Registering Number: 5500550055

Dial-peer tag: 601

Pass-through Mode: p2p

Negotiated In-Expires: 64 Seconds

Next In-Register Due in: 59 Seconds

In-Register Contact: 9.45.36.5

--------------------------------------

 Registrar Index: 1

 Registrar URL: ipv4:9.45.36.4

 Negotiated Out-Expires: 64 Seconds

 Next Out-Register After: 0 Seconds

============================================================
```

Configuring Support for Conditional Header Manipulation of SIP Headers

The Support for Conditional Header Manipulation of SIP Headers feature provides the following enhancements to Cisco Unified Border Element (Cisco UBE):

- The ability to pass unsupported parameters present in a mandatory Session Initiation Protocol (SIP) header from one call leg to another of Cisco UBE.
- The ability to copy contents from one header to another in an outgoing SIP message.

Restrictions

- You cannot configure more than 99 variables for the SIP profiles copy option.
- This feature does not support any header other than SIP.

Passing an Unsupported Parameter Present in a Mandatory Header from One Call Leg to Another of Cisco UBE

Perform this task to pass an unsupported parameter present in a mandatory header from one call leg to another of Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-copylist** *tag*
4. **sip-header** {**sip-req-uri** | *header-name*}
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 class sip-copylist** *tag* **Example:** `Router(config)# voice class sip-copylist 100`	Configures a list of entities to be sent to a peer call leg and enters voice class configuration mode.
Step 4	**sip-header** {**sip-req-uri** | *header-name*} **Example:** `Router(config-class)# sip-header From`	Specifies the SIP header to be sent to the peer call leg.
Step 5	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.

Copying Contents from One Header to Another in an Outgoing SIP Message

Perform the following tasks to copy contents from one header to another in an outgoing SIP message:

- Copying Contents from One SIP Header to Another in an Outgoing Message, page 323 (required)
- Copying Contents from Peer Header to a SIP Header in an Outgoing Message, page 324 (required)

Copying Contents from One SIP Header to Another in an Outgoing Message

Perform this task to copy contents from one SIP to another in an outgoing message.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *tag*
4. **request** *method* **sip-header** *field* {**add** | **copy** | **modify** | **remove**} *string*
5. **response** *option* **sip-header** *field* {**add** | **copy** | **modify** | **remove**} *string*
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 class sip-profiles** *tag* **Example:** `Router(config)# voice class sip-profiles 10`	Enables dial peer-based VoIP SIP profile configurations and enters voice class configuration mode.
Step 4	**request** *method* **sip-header** *field* {**add** | **copy** | **modify** | **remove**} *string* **Example:** `Router(config-class)# request INVITE sip-header contact copy "(.*)" u01`	Modifies SIP profiles to copy the contents from one SIP header to another in a SIP request message.
Step 5	**response** *option* **sip-header field** {**add** | **copy** | **modify** | **remove**} *string* **Example:** `Router(config-class)# response 200 sip-header contact copy "(.*)" u01`	Modifies SIP profiles to copy contents from one SIP header to another in a SIP response message.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.

Copying Contents from Peer Header to a SIP Header in an Outgoing Message

Perform this task to copy contents from peer header to a SIP header in an outgoing message.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *tag*
4. **request** *method* **peer-header sip** {**sip-req-uri** | *header-name*} **copy** *match-pattern variable*
5. **response** *option* **peer-header sip** {**sip-req-uri** | *header-name*} **copy** *match-pattern variable*
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 class sip-profiles** *tag* **Example:** `Router(config)# voice class sip-profiles 10`	Enables dial peer-based VoIP SIP profile configurations and enters class configuration mode.
Step 4	**request** *method* **peer-header sip** {**sip-req-uri** | *header-name*} **copy** *match-pattern variable* **Example:** `Router(config-class)# request invite peer-header contact copy "(.*)" u01`	Copies contents from a peer header to a SIP header in an outgoing SIP request message.
Step 5	**response** *option* **peer-header sip** {**sip-req-uri** | *header-name*} **copy** *match-pattern variable* **Example:** `Router(config-class)# response 200 peer-header contact copy "(.*)" u01`	Copies contents from a peer header to a SIP header in an outgoing SIP response message.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.

Configuring Support for Reporting End-of-Call Statistics in SIP BYE Message

The Support for Reporting End-of-Call Statistics in Session Initiation Protocol (SIP) BYE Message feature enables you to send call statistics to a remote end when a call terminates. The call statistics are sent as a new header in the BYE message or in the 200 OK message (response to BYE message). The statistics include Real-time Transport Protocol (RTP) packets sent or received, total bytes sent or received, total number of packets that are lost, delay jitter, round-trip delay, and the call duration.

This feature enables Cisco Unified Border Element (Cisco UBE) to use the call statistics to update the call data records in Cisco Unified Communications Manager (Cisco UCM) or Cisco Unified Communications Manager Express (Cisco UCME).

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled bu default on Cisco UBE.

A new header P-RTP-Stat is added to the BYE and 200 OK messages. The format of P-RTP-Stat is as follows:

P-RTP-Stat: PS=<Packets Sent>, OS=<Octets Sent>, PR=<Packets Recd>, OR=<Octets Recd>, PL=<Packets Lost>, JI=<Jitter>, LA=<Round Trip Delay in ms>, DU=<Call Duration in seconds>

Table 4 describes the P-RTP-Stat header field description.

Figure 15 P-RTP-Stat Header Fields

Field	Description	Range of Values
PS	Packets Sent	0 to 4294967295
OS	Octets Sent	0 to 4294967295
PR	Packets Received	0 to 4294967295
OR	Octets Received	0 to 4294967295
PL	Packets Lost	0 to 4294967295
JI	Jitter	0 to 4294967295
LA	Round Trip Delay, in milliseconds (ms)	-2147483648 to +2147483647
DU	Call Duration, in seconds	0 to 4294967295

Restrictions

- If the **media flow-around** command is configured, the call statistics are not sent for a 200 OK message.
- If the **media flow-around** command is configured, the call statistics are passed through the Cisco UBE for a BYE message.
- The values are not validated when the incoming statistics are passed to the endpoints. Hence, in some cases the values may be invalid.
- The value of round-trip delay is valid only if the remote end supports Real-Time Control Protocol (RTCP).

Disabling Support for Reporting End-of-Call Statistics in SIP BYE Message feature

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default on the Cisco UBE. That is, the P-RTP-Stat header is added to the list of headers that can be processed through the SIP profiles. You must apply SIP profile rules to remove the header from the mandatory header list.

This section contains the following tasks:

- Defining SIP Profile Rules to Remove a Header, page 326 (required)
- Disabling Support for Reporting End-of-Call Statistics in SIP BYE Message at the Global Level, page 327 (optional)
- Disabling Support for Reporting End-of-Call Statistics in SIP BYE Message at the Dial Peer Level, page 328 (optional)

Defining SIP Profile Rules to Remove a Header

Perform this task to define SIP profile rules to remove a header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *tag*
4. **request bye sip-header p-rtp-stat remove**
5. **response 200 sip-header p-rtp-stat remove**
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 class sip-profiles** *tag* **Example:** `Router(config)# voice class sip-profiles 100`	Configures SIP profiles for a voice class and enters voice class configuration mode.
Step 4	**request BYE sip-header p-rtp-stat remove** **Example:** `Router(config-class)# request bye sip-header p-rtp-stat remove`	Removes the P-RTP-Stat SIP header from the BYE message.

	Command or Action	Purpose
Step 5	**response 200 sip-header p-rtp-stat remove** **Example:** `Router(config-class)# response 200 sip-header p-rtp-stat remove`	Removes the P-RTP-Stat SIP header from the 200 OK message.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.

Disabling Support for Reporting End-of-Call Statistics in SIP BYE Message at the Global Level

Perform this task to disable the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the global level.

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default on Cisco UBE. Hence, to disable the feature, you must modify the SIP profiles to remove the P-RTP-Stat SIP header from the request and the response messages and then configure the modified SIP profile on the Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **sip-profiles** *tag*
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`	Specifies VoIP as the voice encapsulation method and enters voice-service configuration mode.

	Command or Action	Purpose
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters service SIP configuration mode.
Step 5	**sip-profiles** *tag* **Example:** `Router(conf-serv-sip)# sip-profiles 100`	Disables the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the global level. • Here, the Cisco UBE is configured to use the modify SIP profiles as defined in "Defining SIP Profile Rules to Remove a Header" section on page 326 to disable the configuration.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits service SIP configuration mode.

Disabling Support for Reporting End-of-Call Statistics in SIP BYE Message at the Dial Peer Level

Perform this task to disable the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the dial peer level.

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default. Hence to disable the feature, you must modify the SIP profiles to remove the P-RTP-Stat SIP header from the request and the response messages and then configure the modified SIP profile on the Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip profiles** *tag*
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.

	Command or Action	Purpose
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a dial peer to specify the method of voice encapsulation and enters dial peer configuration mode.
Step 4	**voice-class sip profiles** *tag* **Example:** `Router(config-dial-peer)# voice-class sip profiles 100`	Disables the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the dial peer level. • Here, the Cisco UBE is configured to use the modify SIP profiles as defined in "Defining SIP Profile Rules to Remove a Header" section on page 326 to disable the configuration.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer configuration mode.

Configuring RTP Media Loopback for SIP Calls

RTP packets are looped back toward the source device when the RTP Media Loopback for SIP Calls feature is configured on a dial peer. The SIP RTP media loopback can be used during Cisco UBE deployments to make test calls to verify the media path between the endpoints and Cisco UBE. In a voice loopback call, an echo is heard at the device originating the call. In a video loopback call, the locally captured video and the audio echo must be rendered at the source device.

Prerequisites for Configuring RTP Media Loopback for SIP Calls

Media packets must be enabled to pass through the gateway. Use the **media flow-through** command in dial peer voice or voice service configuration mode to enable the media packets.

Restrictions for Configuring RTP Media Loopback for SIP Calls

- SRTP, DTLS, and STUN are not supported in loopback mode.
- Fax (midcall transmit function change) is not supported.
- RSVP is not supported.
- Call transfer is not supported.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **destination-pattern** *string*
5. **session protocol sipv2**
6. **session target loopback:rtp**

7. **incoming called-number** *string*
8. **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 77 voip`	Specifies that the dial peer is a VoIP peer and enters dial peer voice configuration mode.
Step 4	**destination-pattern** *string* **Example:** `Router(config-dial-peer)# destination-pattern 77`	Specifies the prefix or the full E.164 number for the dial peer.
Step 5	**session protocol sipv2** **Example:** `Router(config-dial-peer)# session protocol sipv2`	Specifies the session protocol for calls with the SIP option.
Step 6	**session target loopback:rtp** **Example:** `Router(config-dial-peer)# session target loopback:rtp`	Designates a network-specific address to receive calls from a VoIP dial peer and configures all voice data to loop back to the source.
Step 7	**incoming called-number** *string* **Example:** `Router(config-dial-peer)# incoming called-number 77`	Specifies a digit string that can be matched by an incoming call to associate the call with the dial peer.
Step 8	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial peer voice configuration mode and enters global configuration mode.

Configuration Examples for RTP Media Loopback

This section provides the following examples:

Example: Configuring Video Loopback with Cisco Telepresence System

The following sample output shows Media Loopback for SIP Calls configured on a Cisco Telepresence System (CTS).

```
!
codec profile 1 aacld
 fmtp "fmtp:96
profile-level-id=16;streamtype=5;mode=AAChbr;config=B98C00;sizeLength=13;indexLength=3;ind
exDeltaLength=3;constantDura
tion=480"
!
codec profile 2 h264
 fmtp "fmtp:112 profile-level-id=4D0028;sprop-parametersets=
R00AKAmWUgDwBDyA,SGE7jyA=;packetization-mode=1"
!
voice class codec 4
 codec preference 1 aacld profile 1
 video codec h264 profile 2
!
dial-peer voice 2000 voip
 destination-pattern 2000
 rtp payload-type cisco-codec-fax-ind 110
 rtp payload-type cisco-codec-aacld 96
 rtp payload-type cisco-codec-video-h264 112
 session protocol sipv2
 session target loopback:rtp
 incoming called-number 2000
 voice-class codec 4
 voice-class sip bandwidth audio tias-modifier 64000
 voice-class sip bandwidth video tias-modifier 4500000
!
```

Example: Configuring Video Loopback with Cisco Unified Video Advantage (CUVA)

The following sample output shows Media Loopback for SIP Calls configured on a Cisco Unified Video Advantage (CUVA).

```
!
codec profile 3 h264
 fmtp "fmtp:98 profile-level-id=420015"
 !
voice class codec 6
 codec preference 1 g711ulaw
 video codec h264 profile 3
 !
dial-peer voice 5000 voip
 description CUVA
 destination-pattern 5000
 rtp payload-type cisco-codec-video-h264 98
 session protocol sipv2
 session target loopback:rtp
 incoming called-number 5000
 voice-class codec 6
 voice-class sip bandwidth video tias-modifier 384000
```

Verifying and Troubleshooting SIP-to-SIP Connections on a Cisco Unified Border Element

To troubleshoot or verify connections in an Cisco UBE, perform the following task:

- Troubleshooting Tips, page 332
- Verifying SIP-to-SIP Connections in an Cisco Unified Border Element, page 332

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 commands on the gateway as appropriate to troubleshoot SIP-to-SIP call scenarios:
  - **debug ccsip all**
  - **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*
- *VoIP Debug Commands*

Verifying SIP-to-SIP Connections in an Cisco Unified Border Element

To verify SIP-to-SIP feature configuration and operation, perform the following steps (listed alphabetically) as appropriate.

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.

Configuration Examples for SIP-to-SIP Connections in a Cisco Unified Border Element

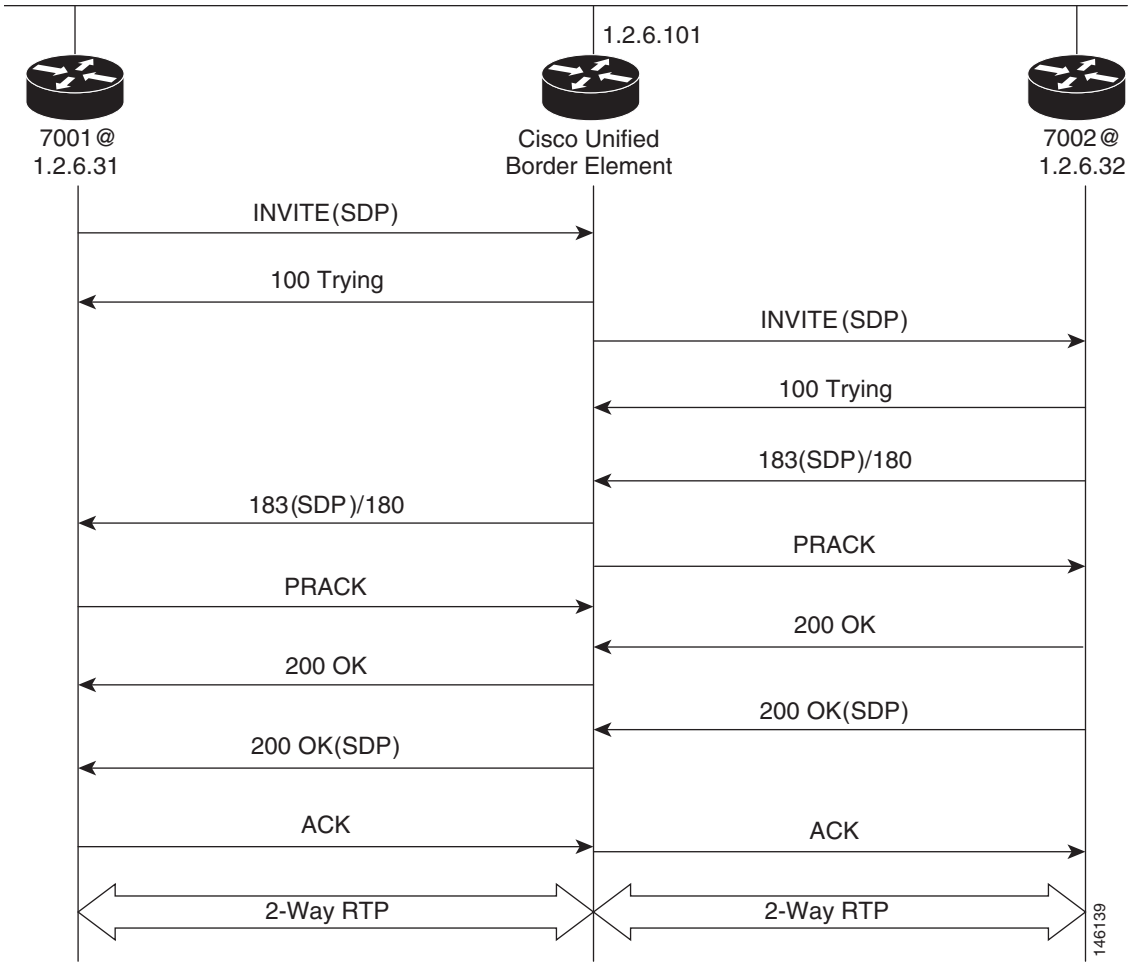
This section contains the following examples:

- Basic SIP-to-SIP Call Flow: Example, page 334
- SRTP-RTP Internetworking: Example, page 336
- Example: Configuring Support for SIP Registration Proxy on Cisco UBE, page 338

Basic SIP-to-SIP Call Flow: Example

The following scenario illustrates a basic SIP-to-SIP call flow, using the Cisco Unified Border Element. Figure 16 shows a simple topology example of the SIP-to-SIP gateway topology.

Figure 16 Cisco Unified Border Element Feature Sample Topology



Call Flow

- The Cisco UBE receives INVITE with Session Description Protocol (SDP) from the OGW. The SDP contains information about the capabilities the endpoint supports for this call like the Audio Codec's, DTMF etc.
- The Codec and DTMF type received from OGW is matched with the incoming configured or default dial-peer.
- The SGW responds to the INVITE message from the OGW by sending a 100 Trying message to OGW.

- The Matched Capabilities are sent to the application which forwards the Matched Capabilities to the outbound SPI.
- The application receives the Matched Capabilities.
- The Codec Type and DTMF type is selected and SDP is formed based on the outgoing dial-peer configured capabilities, and the capabilities received from the application.
- The Cisco UBE sends an Invite with SDP to the TGW.
- The TGW responds to the Cisco UBE with a 100 Trying message.
- The TGW sends 183 (SDP) if the Phone type on TGW is POTS or 180 if the Phone type on TGW is SIP /SCCP Phone to the Cisco UBE.
- Cisco UBE sends a PRACK Message to the TGW.
- OGW receives 183(SDP)/180 from the Cisco UBE.
- TGW sends 200 Ok to the Cisco UBE.
- Cisco UBE receives a PRACK message from the OGW.
- OGW receives 200 Ok from the Cisco UBE.
- The TGW sends 200 Ok with SDP to the Cisco UBE.
- Cisco UBE sends 200 Ok with SDP to the OGW.
- OGW sends ACK to the Cisco UBE.
- Cisco UBE sends ACK towards TGW only after it receives ACK from the OGW.
- Two-phase exchange provides negotiation capabilities based on simple offer/answer model of SDP exchange
- The Contact header field should always carry the address of Cisco UBE and in none of the messages the IP address on one service provider should be sent to other.

  INVITE message received from OGW has Contact: <sip:70005@1.2.6.31:5060>

  INVITE message sent from Cisco UBE has Contact: <sip:70005@1.2.6.101:5060>

  The same applies to the Contact address when sending response messages towards OGW.

Table 5 shows support for Early Media and their supported Codec and packetization values.

Table 5 Early Media Codec packetization values

Incoming Leg	Outgoing Leg
711 A/U	711 A/U
723 r53 723 r63 723 ar53 723 ar63	723 r53 723 r63 723 ar53 723 ar63
726 r16 726 r24 726 ar32	726 r16 726 r24 726 ar32
728	728
729r8/729/729ar8 729br8/729abr8	729r8/729/729ar8 729br8/729abr8
gsmfr/gsmefr	gsmfr/gsmefr

Transparent Codec

Most video endpoints have proprietary codecs for both audio and video. This makes transparent codecs are most important when handling video calls. If Codec T is configured under the dial-peer all the audio capabilities are transparently passed from one leg to another. Codecs that are not supported by the platform are also passed from incoming leg to outgoing leg.

Table 6 Transport Codec

Incoming Leg	Outgoing Leg	Support
H.323	H.323	Yes

Note If both g79r8 and g729br8 is configured using voice class Codec then g729br8 is only codec sent in INVITE.

INVITE message contains

m=audio 19078 RTP/AVP 18 101 19

c=IN IP4 1.5.5.2

a=rtpmap:18 G729/8000

a=fmtp:18 annexb=yes

- if TGW sends in session progress G729r8 then G.729r8 is the negotiated codec.
- if TGW sends in session progress G729br8 then G.729br8 is the negotiated codec.

Packetization

The packetization values with different codecs are sent to Cisco UBE with attribute “ptime”. The Cisco UBE should ensure that the packetization value received from OGW is sent to TGW.

For example: ptime = 10 is sent when g711ulaw is configured 80 bytes.

SRTP-RTP Internetworking: Example

The following example shows how to configure Cisco Unified Border Element support for SRTP-RTP internetworking. In this example, the incoming call leg is RTP and the outgoing call leg is SRTP.

```
enable
 configure terminal
 ip http server
 crypto pki server 3845-cube
  database level complete
  grant auto
  no shutdown
%PKI-6-CS_GRANT_AUTO: All enrollment requests will be automatically granted.
% Some server settings cannot be changed after CA certificate generation.
% Please enter a passphrase to protect the private key or type Return to exit
Password:
Re-enter password:
% Generating 1024 bit RSA keys, keys will be non-exportable...[OK]
% SSH-5-ENABLED: SSH 1.99 has been enabled
% Exporting Certificate Server signing certificate and keys...
% Certificate Server enabled.
%PKI-6-CS_ENABLED: Certificate server now enabled.
```

```
!
crypto pki trustpoint secdsp
 enrollment url http://10.13.2.52:80
 serial-number
 revocation-check crl
 rsakeypair 3845-cube
 exit
!
crypto pki authenticate secdsp
Certificate has the following attributes:
 Fingerprint MD5: CCC82E9E 4382CCFE ADA0EB8C 524E2FC1
 Fingerprint SHA1: 34B9C4BF 4841AB31 7B0810AD 80084475 3965F140
% Do you accept this certificate? [yes/no]: yes
Trustpoint CA certificate accepted.
crypto pki enroll secdsp
% Start certificate enrollment ..
% Create a challenge password. You will need to verbally provide this password to the CA
Administrator in order to revoke your certificate. For security reasons your password will
not be saved in the configuration. Please make a note of it.
Password:
Re-enter password:
% The subject name in the certificate will include: 3845-CUBE
% The serial number in the certificate will be: FHK1212F4MU
% Include an IP address in the subject name? [no]:
Request certificate from CA? [yes/no]: yes
% Certificate request sent to Certificate Authority
% The 'show crypto pki certificate secdsp verbose' command will show the fingerprint.
CRYPTO_PKI:  Certificate Request Fingerprint MD5: 56CE5FC3 B8411CF3 93A343DA 785C2360
CRYPTO_PKI:  Certificate Request Fingerprint SHA1: EE029629 55F5CA10 21E50F08 F56440A2
DDC7469D
%PKI-6-CERTRET: Certificate received from Certificate Authority
!
voice-card 0
 dspfarm
 dsp services dspfarm
 voice-card 1
 dspfarm
 dsp services dspfarm
 exit
!
sccp local GigabitEthernet 0/0
sccp ccm 10.13.2.52 identifier 1 version 5.0.1
sccp
SCCP operational state bring up is successful.sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register sxcoder
 dspfarm profile 1 transcode universal security
  trustpoint secdsp
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec ilbc
  codec g729br8
  maximum sessions 84
  associate application sccp
  no shutdown
  exit
!
telephony-service
%LINEPROTO-5-UPDOWN: Line protocol on Interface EDSP0, changed state to upsdspfarm units 1
 sdspfarm transcode sessions 84
 sdspfarm tag 1 sxcoder
```

```
 em logout 0:0 0:0 0:0
 max-ephones 4
 max-dn 4
 ip source-address 10.13.2.52
Updating CNF files
CNF-FILES: Clock is not set or synchronized, retaining old versionStamps
CNF files updating complete
 secure-signaling trustpoint secdsp
 tftp-server-credentials trustpoint scme
CNF-FILES: Clock is not set or synchronized, retaining old versionStamps
CNF files update complete (post init)
 create cnf-files
CNF-FILES: Clock is not set or synchronized, retaining old versionStamps
 no sccp
!
sccp
SCCP operational state bring up is successful.
end
%SDSPFARM-6-REGISTER: mtp-1:sxcoder IP:10.13.2.52 Socket:1 DeviceType:MTP has registered.
%SYS-5-CONFIG_I: Configured from console by console
dial-peer voice 201 voip
 destination-pattern 5550111
 session protocol sipv2
 session target ipv4:10.13.25.102
 incoming called-number 5550112
 codec g711ulaw
!
dial-peer voice 200 voip
 destination-pattern 5550112
 session protocol sipv2
 session target ipv4:10.13.2.51
 incoming called-number 5550111
 srtp
 codec g711ulaw
```

Example: Configuring Support for SIP Registration Proxy on Cisco UBE

The following example shows how to configure support for the SIP registration proxy on the Cisco UBE.

```
!
!
voice service voip
sip
  registrar server expires max 121 min 61
  registration passthrough static challenge rate-limit expires 9000 fail-count 5
registrar-index 1 3 5
!
dial-peer voice 1111 voip
 destination-pattern 1234
 voice-class sip pass-thru content unsupp
 session protocol sipv2
 session target registrar
!
dial-peer voice 1111 voip
 destination-pattern 1234
 voice-class sip pass-thru content unsupp
 voice-class sip registration passthrough static rate-limit expires 9000 fail-count 5
registrar-index 1 3 5
 authentication username 1234 password 7 075E731F1A realm cisco.com challenge
 session protocol sipv2
 session target registrar
```

```
!

sip-ua
 registration spike 1000
!
!
```

Troubleshooting Tips

The following commands can help troubleshoot SIP-to-SIP calls on the Cisco UBE:

- **debug ccsip all**
- **debug voip ccapi**

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
- 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 SIP-to-SIP Cisco Unified Border Element 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

Related Topic	Document Title
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
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 Topic	Document Title
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 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 SIP-to-SIP Connections on a Cisco Unified Border Element

Table 7 lists the features in this module and provides links to specific configuration information.

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

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 7 lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element

Feature Name	Releases	Feature Information
Address Hiding	12.4(9)T	Address hiding in all SIP messages. The following section provides information about this feature: • Configuring IP Address-Hiding, page 189 The following command was modified: **address-hiding**.
Adjustable Timers for Registration Refresh and Retries	12.4(22)Y 15.0(1)M	This feature provides the ability for Cisco IOS software to: • Refresh the REGISTER at a configurable fraction of the expiry timer. • Retransmit REGISTER upon failure responses per the min-expires value in a "423 interval too brief" response, or retry-after if present and terminal re-registration interval if retry-after value is absent in 4xx/5xx/6xx responses. • Retransmit REGISTER per Timer E up to 32 seconds, and at a user-defined random interval thereafter. The following section provides information about this feature: • Configuring Adjustable Timers for Registration Refresh and Retries, page 253 The following commands were introduced or modified: **registrar** and **retry register**.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Allow 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 for billing, security, call-admission control, QoS, and signaling interworking. The following section provides information about this feature: • Configuring SIP-to-SIP Connections on a Cisco Unified Border Element, page 190 The following command was modified: **allow-connections**.
Assisted Real-time Transport Control Protocol (RTCP) Report Generation	15.1(2)T	This feature adds the ability for the Cisco UBE to generate standard RTCP keepalive reports on behalf of endpoints and ensures the liveliness of a media session during prolonged periods of silence, such as call hold. The following section provides information about this feature: • Configuring Assisted Real-time Transport Control Protocol (RTCP) Report Generation, page 271 The following commands were introduced or modified: **debug ip rtp protocol**, **debug voip rtcp**, **debug voip rtp**, **ip rtcp report interval**, and **rtcp keepalive**.
Call Admission Control	12.4(6)T	This feature enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP UAs based on CPU, memory, and total calls.
Call Escalation from Voice to Video	15.1(4)M	This feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. The following section provides information about this feature: • Configuring Call Escalation from Voice to Video, page 194
Calls for SIP and H.323	12.4(15)XY	This feature was introduced.
Cisco UBE MIB Support	15.0(1)XA 15.1(1)T	This feature was introduced.
Cisco Unified Communications Manager Connections	12.4(6)T	Interoperability with Cisco Unified Communications Manager 5.0 and BroadSoft.
Codec Support	12.4(4)T 12.4(11)T	In Cisco IOS Release 12.4(4)T, support for the SIP-to-SIP basic functionality for SBC for Cisco UBE was introduced. This functionality provides termination and reorigination of both signaling and media between VoIP and video networks using SIP signaling. In Cisco IOS Release 12.4(11)T, support for the iLBC codec was introduced.
Configurable Bandwidth Parameters for SIP Calls	12.4(15)XZ	This feature provides the ability to manually configure the bandwidth that is signaled in the outbound SIP invite.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Configurable SIP Parameters via DHCP	12.4(22)YB 15.0(1)M	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP. The following commands were introduced or modified: **credentials** (sip-ua), **debug ccsip dhcp**, **dhcp interface**, **ip dhcp-client forcerenew**, **outbound-proxy**, **registrar**, **session target** (VoIP dial peer), **show sip dhcp**, **voice-class sip outbound-proxy**.
DTMF Relay	12.4(4)T 12.4(6)XE 12.4(11)T	In Cisco IOS Release 12.4(4)T, support for the DTMF Relay Digit-Drop for SIP Calls Using NTE feature was introduced. In Cisco IOS Release 12.4(11)T, support for passing DTMF tones out-of-band and dropping in-band digits to avoid sending both tones to the outgoing leg on an H.323-to-SIP Cisco Unified Border Element was introduced. In Cisco IOS Release 12.4(6)XE, support for G.711 Inband DTMF to RFC 2833 was introduced.
ENUM Support	12.4(6)T	This feature enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP UAs.
Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure	15.0(1)XA 15.1(1)T	This feature provides the option to configure the error response code when a dial peer is busied out because of an Out-of-Dialog OPTIONS ping failure. The following section provides information about this feature: • Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure, page 229 The following commands were introduced or modified: **error-code-override options-keepalive failure**, **voice-class sip error-code-override options-keepalive failure**.
Fax/Modem	12.4(6)T	In Cisco IOS Release 12.4(6)T, support for modem passthrough was introduced.
Forced Update of SIP Parameters via DHCP	12.4(22)YB 15.0(1)M	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP. The following section provides information about this feature: • Enabling Forced Update of SIP Parameters via DHCP, page 245
Hosted NAT Traversal for SIP	12.4(9)T	This feature was introduced.
Media Antitrombone	15.1(3)T	The Media Antitrombone feature is a media signaling service in SIP entity to overcome media loops. The following section provides information about this feature: • Configuring Media Antitrombone, page 212

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Media Modes	12.3(1) 12.4(9)T 15.1(3)T	In Cisco IOS Release 12.3(1), support for media flow-through and flow-around improving scalability and performance when network-topology hiding and bearer-level interworking features are not required was introduced. In Cisco IOS Release 12.4(9)T, support for media flow-around was introduced. In Cisco IOS Release 15.1(3)T, support for the Configured Delayed-Offer to Early-Offer Media Flow-Around feature at the global and dial-peer level was introduced.
Out-of-dialog OPTIONS Ping to Monitor Dial Peers to Specified SIP Servers and Endpoints	12.4(22)YB 15.0(1)M	This feature provides a keepalive mechanism at the SIP level between any number of destinations. The generic heartbeat mechanism allows the Cisco UBE to monitor the status of SIP servers or endpoints and provide the option of busying-out an associated dial peer upon total heartbeat failure. The following section provides information about this feature: • Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints, page 226 The following command was introduced: **voice-class sip options-keepalive**.
Pass-through of STUN and DTLS Packets	15.1(2)T	This feature enables and supports Cisco TelePresence System (CTS) endpoints to send and receive STUN and DTLS packets to open and refresh firewall pinholes and establish the Secure Real-Time Transport Protocol (SRTP) security parameters. The following section provides information about this feature: • Configuring Assisted Real-time Transport Control Protocol (RTCP) Report Generation, page 271
Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information	12.4(22)YB 15.0(1)M	This feature supports the use of preloaded routes for outgoing INVITE messages. The system routes INVITE messages based on REGISTER message information, such as the path: and Service-Route values. The following commands were modified: **url** (SIP) and **voice-class sip url**.
RTP Media Loopback for SIP Calls	15.1(4)M	RTP packets are looped back toward the source when the RTP Media Loopback for SIP Calls feature is configured on a dial peer. SIP RTP media loopback helps in verifying the media path between the device originating the call and the intermediate device.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Element	12.4(22)YB 15.0(1)M	This feature supports selective filtering of outgoing provisional responses, including 180-Alerting, and 183-Session In Progress responses. Selective filtering can be further based on the availability of media information in the received provisional response.
Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers	12.4(22)YB 15.0(1)M	This feature supports the construction of request URIs in tel: format. The system supports this format for both the To: header and the Request-Line and the system supports appending the phone-context parameter to the tel: URL. The following command was introduced: **tel-config to-hdr**. The following commands were modified: **url**, **voice-class sip url**.
Session Refresh	12.4(11)T 12.4(15)XZ 12.4(20)T	In Cisco IOS Release 12.4(11)T, this feature was introduced. In Cisco IOS Release 12.4(15)XZ, support for SIP-to-SIP session refresh call flows using reINVITEs was introduced.
Session Refresh with Reinvites	12.5(15)XZ	This feature expands the ability of the Cisco UBE to control the session refresh parameters and ensure that the session does not time out.
Signal Interworking	12.4(6)T	Delayed Media Call, Media Inactivity. Enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP UAs.
SIP Error Message Pass Through	12.4(11)XJ2	This feature allows a received error response from one SIP leg to pass transparently over to another SIP leg. The following section provides information about this feature: • Configuring SIP Error Message Pass Through, page 196
SIP Listening Port	12.4(15)XZ 12.4(20)T	This feature allows users to configure the port that SIP messages are listened on.
SIP Parameter Modification	12.4(15)XZ 12.4(20)T	This feature allows users to change the standard SIP messages sent from the Cisco SIP stack for better interworking with different SIP entities.
SIP Registration Message	12.4(24)T	This feature provides the ability to send a SIP Registration Message from the Cisco Unified Border Element using the **credentials** command.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
SRTP-RTP Internetworking	12.4(22)YB 15.0(1)M	This feature allows secure enterprise-to-enterprise calls. Support for SRTP-RTP internetworking between one or multiple Cisco Unified Border Elements is enabled for SIP-SIP audio calls. The following sections provide information about this feature: • Information About Cisco Unified Border Element Support for SRTP-RTP Internetworking, page 257 • How to Configure Cisco Unified Border Element Support for SRTP-RTP Internetworking, page 259 The following command was introduced: **tls**.
Supplementary Services	12.4(9)T	• Message waiting indication • Call waiting • Call transfer • Call forward • Distinctive ringing • Call hold/resume • Music on hold
Support for Conditional Header Manipulation of SIP Headers	15.1(3)T	The Support for Conditional Header Manipulation of SIP Headers feature provides the following enhancements to Cisco UBE: • The ability to pass unsupported parameters present in a mandatory header from one call leg to another. • The ability to copy contents from one header to another header in an outgoing SIP message. The following section provides information about this feature: • Configuring Support for Conditional Header Manipulation of SIP Headers, page 321 The following commands were introduced or modified: **response**, **response peer-header**, **request**, **request peer-header**, **sip-header**, **voice-class sip copy-list**, **voice class sip-copylist**.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls	15.0(1)XA 15.1(1)T	The Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls feature provides dynamic payload type interworking for DTMF and codec packets for SIP to SIP calls. The following section provides information about this feature: • Configuring Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls Feature, page 218 The following commands were introduced or modified: **asymmetric payload** and **voice-class sip asymmetric payload**.
Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element	15.1(2)T	The Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element feature supports negotiation of an audio codec using the Voice Class Codec and Codec Transparent infrastructure on the Cisco UBE. The following section provides information about this feature: • Configuring Support for Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element, page 301 The following command was introduced or modified: **voice-class codec**.
Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco UBE	12.4(22)YB 15.0(1)M	The following commands were introduced: **call-route p-called-party-id**, **privacy-policy, random-contact**, **random-request-uri validate**, **voice-class sip call-route p-called-party-id**, **voice-class sip privacy-policy**, **voice-class sip random-contact**, **voice-class sip random-request-uri validate**.
Support for Reporting End-of-Call Statistics in SIP BYE Message	15.1(3)T	The Support for Reporting End-of-Call Statistics in SIP BYE Message feature enables you to send call statistics to remote ends when a call terminates. These statistics are sent as a new header in a BYE message or in the 200 OK message. The following section provides information about this feature: • Configuring Support for Reporting End-of-Call Statistics in SIP BYE Message, page 325
Support for Session Refresh with Reinvites	12.4(15)XZ	This feature expands the ability of the Cisco Unified Border Element to control the session refresh parameters and ensure that the session does not time out.

Table 7 Feature Information for SIP-to-SIP Connections on a Cisco Unified Border Element (continued)

Feature Name	Releases	Feature Information
Support for SIP Registration Proxy on Cisco UBE	15.1(3)T	The Support for SIP Registration Proxy on Cisco UBE feature provides support for sending outbound registrations from the Cisco UBE based on incoming registrations. This feature enables direct registration of SIP endpoints with the SIP registrar in hosted UC deployments. This feature also provides various benefits for handling Cisco UBE deployments with no IPPBX support. The following section provides information about this feature: • Configuring Support for SIP Registration Proxy on Cisco UBE, page 1 The following commands were introduced or modified: **authentication** (dial peer), **registrar server**, **registration passthrough**, **registration spike**, **show sip-ua registration passthrough status**, **voice-class sip registration passthrough static rate-limit**.
Support for SIP UPDATE Message per RFC 3311	15.1(3)T	The Support for SIP UPDATE Message per RFC 3311 feature provides SDP support for SIP-to-SIP calls. The SIP SPI is modified to support the following media changes using the UPDATE message: • Early dialog SIP-to-SIP media changes • Mid dialog SIP-to-SIP media changes The following section provides information about this feature: • Support for SIP UPDATE Message per RFC 3311, page 1
Tcl IVR	12.4(6)T	Tcl scripts with SIP NOTIFY VoiceXML with SIP-to-SIP.
Transport Protocols	12.4(6)T	TCP and UDP interworking.
Unsupported Content Pass-through	12.4(22)YB 15.0(1)M	This feature supports the ability to pass through end to end headers at a global or dial-peer level that are not processed or understood in a SIP-trunk-to-SIP-trunk scenario. The pass-through functionality includes all or only a configured list of unsupported or non-mandatory SIP headers, and all unsupported content/MIME types. The following commands were introduced or modified: **pass-thru** and **voice-class sip pass-thru**.