



Overview of Cisco Unified Border Element

Revised: October 27, 2009,
First Published: June 19, 2006
Last Updated: October 27, 2009

This Cisco Unified Border Element (previously known as the Cisco Multiservice IP-to-IP Gateway) is a special Cisco IOS software image that runs on Cisco multiservice gateway platforms. It provides a network-to-network interface point for billing, security, call admission control, quality of service, and signaling interworking. This chapter describes basic gateway functionality, software images, topology, and summarizes supported features.



Activation

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

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

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

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



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

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Prerequisites for Cisco Unified Border Element Configuration

Cisco Unified Border Element Hardware

- Install the routers that will serve as session border controllers in your VoIP network.

Cisco Unified Border Element Software

- Obtain the appropriate feature license for each router on which you will install an image that supports the Unified Border Element feature. Additional information on obtaining a feature license can be found at:

http://www.cisco.com/en/US/products/sw/voicesw/ps5640/products_data_sheet09186a00801da698.html



Activation

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- Install the appropriate Cisco IOS image on each router and configure a working VoIP network. Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS 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.

Table 1 provides additional information on image and feature sets that support the Cisco Unified Border Element.

Table 1 Cisco IOS Image and Feature Sets for the Cisco Unified Border Element Feature

Platform	Software Image Name	Software Feature Set
Cisco 2601XM	c2600-adventerprisek9_ivs-mz	Cisco 2600 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES
Cisco 2611XM	c2600-ipvoice_ivs-mz	Cisco 2600 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 2620XM		
Cisco 2621XM	c2600-ipvoice_ivs-mz	Cisco 2600 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 2650XM		
Cisco 2651XM		

Table 1 Cisco IOS Image and Feature Sets for the Cisco Unified Border Element Feature (continued)

Platform	Software Image Name	Software Feature Set
Cisco 2691	c2691-adventerprisek9_ivs-mz c2691-ipvoice_ivs-mz	Cisco 2600 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 2600 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 2811 Cisco 2821 Cisco 2851	c2800nm-adventerprisek9_ivs-mz c2800nm-ipvoice_ivs-mz	Cisco 2800 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 2800 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 2801	c2801-adventerprisek9_ivs-mz c2801-ipvoice_ivs-mz	Cisco 2801 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 2801 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 3725	c3725-adventerprisek9_ivs-mz c3725-ipvoice_ivs-mz	Cisco 3725 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 3725 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 3745	c3745-adventerprisek9_ivs-mz c3745-ipvoice_ivs-mz	Cisco 3745 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 3745 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 3825	c3825-adventerprisek9_ivs-mz c3825-ipvoice_ivs-mz	Cisco 3825 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 3825 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco 3845	c3845-adventerprisek9_ivs-mz c3845-ipvoice_ivs-mz	Cisco 3845 INT VOICE/VIDEO, IPIPGW, TDMIP GW AES Cisco 3845 INT VOICE/VIDEO, IPIPGW, TDMIP GW
Cisco AS5350XM	c5350-jk9su2_ivs-mz c5350-js_ivs-mz	Cisco AS5350 Series IOS INT Voice/Video IPIPGW, TDMIP GW LI Cisco AS5350 Series IOS INT Voice/Video IPIPGW, TDMIP GW EPLUS
Cisco AS5400XM	c5400-jk9su2_ivs-mz c5400-js_ivs-mz	Cisco AS5400 Series IOS INT VOICE/VIDEO IPIPGW, TDMIP GW LI Cisco AS5400 Series IOS INT VOICE/VIDEO IPIPGW, TDMIP GW EPLUS
Cisco 7200	c7200-adventerprisek9-mz c7200-adipservicesk9li-mz	Cisco 7200 Series IOS Advanced Enterprise Services Cisco 7200 Series IOS ADV IP Services w/ Lawful Intercept
Cisco 7301	c7301-adventerprisek9-mz c7301-adipservicesk9li-mz	Cisco 7301 Series IOS Advanced Enterprise Services Cisco 7301 Series IOS ADV IP Services w/ Lawful Intercept

Restrictions for Cisco Unified Border Element Configuration

- Cisco Unified Border Elements that require the Registration, Admission, and Status (RAS) protocol must have a via-zone-enabled gatekeeper or equivalent.
- Cisco Unified Border Elements interoperate with Cisco ATA 186, Cisco ATA 188, Cisco Unified Communications Manager, Cisco CallManager Express 3.1, Cisco IOS gateways, NetMeeting, and Polycom ViewStation.
- Cisco fax relay is reported as a voice call on an Cisco Unified Border Element. Fax relay is enabled by default for all systems. No further configuration is needed.
- Fax calls are reported as a modem plus fax call when modem CLI are present.
- Cisco Unified Border Element supports T.38 fax relay (H.323 Annex D). However, endpoints configured with Named Signaling Events (NSE) may result in reduced fax transmission quality and are not supported.
- Codec filtering must be based on codec types; filtering based on byte size is not supported.

- When a Tcl script is running on an Cisco Unified Border Element, the Cisco Unified Border Element does not support ringback tone generation.
- Transcoding is not supported on the Cisco AS5350XM, AS5400XM, Cisco 7200 and the Cisco 7301.

Information About Cisco Unified Border Element

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 H.323 VoIP and videoconferencing calls from one IP network to another. This gateway performs most of the same functions of a PSTN-to-IP gateway, but typically joins two IP call legs, rather than a PSTN and an IP call leg. Media packets can flow either through the gateway (thus hiding the networks from each other) or around the border element, if so configured.

Cisco Unified Border Element is a special Cisco IOS software image that runs on Cisco Unified Border Element platforms. It provides a network-to-network interface point for billing, security, call admission control, quality of service, and signaling interworking.

Cisco Unified Border Element is designed to meet the interconnection needs of Internet telephony service providers (ITSPs) and of enterprises. One set of images provides basic interconnection and a second set provides interconnection through an Open Settlement Protocol (OSP) provider, enabling ITSPs to gain the benefits of the Cisco Unified Border Element while making use of the routing, billing, and settlement capabilities offered by OSP-based clearinghouses.

For the most effective and scalable results, use the Cisco Unified Border Element concurrently with a Cisco gatekeeper

Feature benefits include the following:

- Capacity control and improved call routing control using carrier-based routing with the Cisco Unified Border Element feature and routing traffic through the gateways.
- Improved billing and settlement capabilities.
- Provides key services at the edge of the network for scalability.

To configure the Cisco Unified Border Element Feature, you should understand the following concepts:

- [Gateway Functionality, page 21](#)
- [Cisco Unified Border Element Network Topology, page 21](#)
- [Features Supported by the Cisco Unified Border Element, page 23](#)

Gateway Functionality

Gateways are responsible for the following tasks.

- Media stream handling and speech path integrity
- DTMF relay
- Fax relay and passthrough
- Digit translation and call processing
- Dial peers and codec filtering
- Carrier ID handling
- Gateway-based billing
- Termination and re-origination of signaling and media

Cisco Unified Border Element Network Topology

In the current VoIP market, ITSPs who provide wholesale VoIP services use their own IP-to-TDM gateways to exchange calls with the PSTN. Problems occur when a wholesaler receives a call from an originating ITSP and decides to terminate the call to another ITSP. Because it does not own the PSTN gateways, the wholesaler does not receive call setup or release information and therefore cannot bill for the call. Wholesalers are forced either to forbid these connections, thereby foregoing a potential revenue source, or to set up the call through a combination of back-to-back IP-to-TDM gateways. This solution results in reduced quality due to double media coding and decoding, and it wastes TDM port resources.

Cisco Unified Border Element allows the wholesaler to terminate the call from the originating ITSP and then reoriginate it, thereby providing a point at which accurate call detail records (CDRs) can be collected for billing.

The superior interconnect capability provided by the Cisco Unified Border Element enables service providers to conceal their internal network and business relationships while improving call admission control, flexible routing, and protocol interworking capabilities.

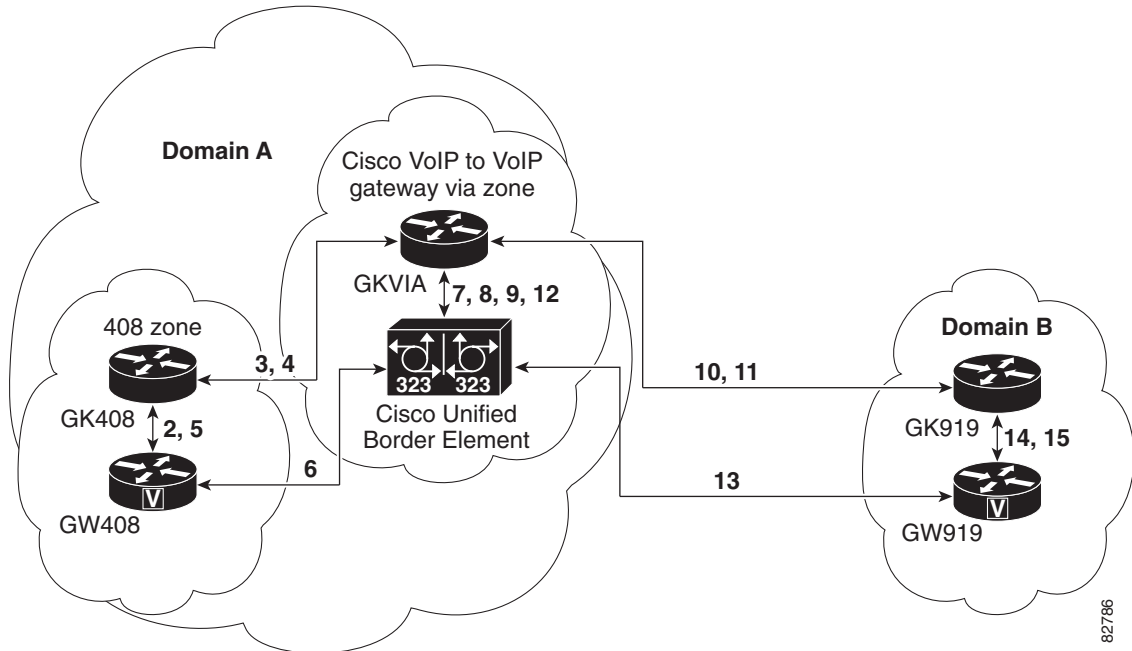
The Cisco Unified Border Element includes the following changes to gateways and gatekeepers to allow Cisco UBE call legs:

- Support for H.323-to-H.323, H.323-to-SIP, and SIP-to-SIP connection types
- Support for transparent codec on H.323-to-H.323 connection types
- Support for H.323 call capacities
- Introduction of gatekeeper via-zones. *Via-zone* is a Cisco term for a zone that contains Cisco Unified Border Elements and via-zone-enabled gatekeepers. A via-zone-enabled gatekeeper is capable of recognizing via-zones and sending traffic to via-zone gateways. Cisco via-zone-enabled gatekeepers include a via-zone command-line interface (CLI) command.

Via-zones are usually located on the edge of an ITSP network and are like a VoIP transfer point, or tandem zone, where traffic passes through on the way to the remote zone destination. Gateways in this zone terminate requested calls and reoriginate traffic to its final destination. Via-zone gatekeepers operate as usual for applications that are not Cisco UBE gatekeepers in via-zones support resource management (for example, gateway selection and load balancing) using the Capacities field in the H.323 Version 4 RAS messages.

Figure 1 shows a simple topology example of the Cisco Unified Border Element using via-zone gatekeepers.

Figure 1 Cisco Unified Border Element Feature Sample Topology



The gatekeeper in Domain A and the gatekeeper in Domain B are connected to the via-zone gatekeeper. GK408 and the via-zone gatekeeper exchange Registration, Admission, and Status (RAS) messages for the originating side. Then the connection is made between the originating gateway and the Cisco Unified Border Element. The via-zone gatekeeper exchanges RAS messages with GK919 for the terminating side. If the call is accepted, the Cisco Unified Border Element completes the connection from GW408 to GW919, and the media flows through the Cisco Unified Border Element.

In a basic call scenario, on receiving a location request (LRQ) message from the originating gatekeeper (GK408), the via-zone-enabled gatekeeper (GKVIA) processes the message and determines that the call should be set up using the Cisco Unified Border Element. After the originating gateway receives its admission confirmation (ACF) message, it sets up the call.

With the Cisco Unified Border Element feature, instead of the originating gateway signaling the terminating gateway directly, the Cisco Unified Border Element controls the call set-up both the signaling and media channel. The Cisco Unified Border Element is terminating the signaling and media channels, but the information associated with the media is propagated through to the opposite call leg. This process allows the endpoints to determine what media channel capabilities to use for the call. When the call is established, the audio stream flows through the Cisco Unified Border Element, meaning that the gateway terminates the audio channel on one call leg and then reoriginates it to the other leg.

The following scenario illustrates a basic call from the originating gateway to the terminating gateway, using the Cisco Unified Border Element and gatekeepers.

1. GW408 (the originating gateway) calls someone in the 919 area code, which is serviced by GW919 (the terminating gateway).
2. GW408 sends an ARQ with the called number having the 919 area code to a gatekeeper in its zone (GK408).

3. GK408 resolves 919 to belong to a via-zone gatekeeper (GKVIA). GK408 then sends an LRQ to GKVIA.
4. GKVIA receives the LRQ for the 919 number. GKVIA resolves the 919 prefix to belong to the Cisco Unified Border Element. GKVIA is configured to route requests for 919 prefix calls through its Cisco Unified Border Element. GKVIA sends an LCF to GK408.
5. GK408 returns an ACF specifying Cisco Unified Border Element to GW408.
6. GW408 sends a SETUP message to Cisco Unified Border Element for the 919 number.
7. Cisco Unified Border Element consults GKVIA with an ARQ message with the **answerCall=true** parameter to admit the incoming call.
8. GKVIA responds with an ACF to admit the call. From the perspective of the gatekeeper, the first call leg has been established.
9. Cisco Unified Border Element has a dial peer specifying that RAS messages should be sent to GKVIA for all prefixes. Cisco Unified Border Element initiates the resending of the call by sending the ARQ message with the **answerCall** parameter set to, false to GKVIA for 919.
10. GKVIA knows that prefix 919 belongs to GK919, and since the source zone is the via-zone, the GKVIA sends an LRQ to GK919.
11. GK919 sees prefix 919 as a local zone and sends an LCF pointing to GW919.
12. GKVIA returns an ACF specifying GW919.
13. Cisco Unified Border Element sends a SETUP message to GW919 for the 919 call.
14. GW919 sends an ARQ to GK919 to request admission for the call.
15. GK919 sends an ACF with the **answerCall=true** parameter.

All other messages (for example, Proceeding, Alerting, and Connect) are created as two legs between GW408, and GW919, with the Cisco Unified Border Element acting as an intermediate gateway.

Features Supported by the Cisco Unified Border Element

This section contains lists of the following types of supported features:

- [Gateway Call-Signaling Features, page 24](#)
- [Cisco Unified Border Element Protocol Features, page 28](#)
- [Protocol Interworking, page 29](#)
- [Billing Features, page 30](#)
- [Application and Tcl Script Features, page 30](#)
- [Interoperability Features, page 30](#)
- [IVR Features, page 31](#)
- [Lawful Intercept Support, page 31](#)
- [QoS Features, page 32](#)
- [Tcl Objects Supported by the Cisco Unified Border Element, page 32](#)

Gateway Call-Signaling Features

Table 2 is a list of supported gateway call-signaling features.

Table 2 Cisco Unified Border Element Call Signaling Features

Feature	Details	H.323-to-H.323 Support?	H.323-to-SIP Support?	SIP-to-SIP Support?	Additional Information
Accounting					
	Calling/called name and number RADIUS call accounting records	Yes	Yes	Yes	—
	Conference ID for call relating the two call legs	Yes	Yes	Yes	—
Address Hiding					
	Address hiding	Yes	Yes	Yes	—
Call Admission Control					
	Call Admission Control	Yes	Yes	Yes	CPU, memory utilization, total calls, Max connections, RSVP and IP Circuits for all the protocol combinations (H.323-to-H.323, H.323-to-SIP, and SIP-to-SIP).
	RSVP nonsynchronized	Yes	No	No	Video only
	RSVP synchronized	Yes	No	No	Video only
Cause Codes					
	SIP Cause Codes	No	No	Yes	—
Cisco CallManager Connections					
	Interoperability with Cisco Unified Communications Manager 5.0 and Cisco Unified Communications Manager 4.1.3	Yes	Yes	Yes	—
	No MTP for Cisco Unified Communications Manager Trunks to Cisco Unified BE	Yes	No	No	—
Codec Support					
	Audio Codecs	Yes	Yes	Yes	G.711u, G.711a, G.723, G.726, G.729r8, G.728, AMR-NB, iLBC.
	Codec Transparent Support	Yes	Yes	Yes	SIP-to-SIP support in Cisco IOS Release 12.4(22)YB and later.
	Video Codecs	Yes	No	No	H.261, H.263, H.264.
Codec Transcoding					

Table 2 Cisco Unified Border Element Call Signaling Features

Feature	Details	H.323-to-H.323 Support?	H.323-to-SIP Support?	SIP-to-SIP Support?	Additional Information
	Codec bytes payload value negotiation	Yes	No	No	Payload size is passed transparently and negotiated between the endpoints. Codec bytes configuration on the Cisco Unified Border Element is ignored.
	Codec transcoding (G.711-G.729)	Yes	Yes	No	—
	DTMF Transcoding with the Cisco AS5xxx platforms	Yes	Yes	Yes	—
DTMF					
	DTMF	Yes	Yes	Yes	Configuration must be consistent between the originating and terminating gateways. DTMF configuration is needed at the Cisco Unified Border Element.
	KPML	No	Yes	Yes	—
	DTMF relay and hookflash relay	Yes	No	No	H.245 alphanumeric, H.245 signal, RFC 2833, and Cisco RTP DTMF relay types supported. Configuration not needed on Cisco Unified Border Element.
	G.711 Inband DTMF to RFC 2833	Yes	Yes	Yes	—
ENUM Support- RFC-2916					
	ENUM support	Yes	Yes	Yes	—
Fax/Modem					
	Cisco-proprietary fax relay	Yes	Yes	Yes	Fax relay is enabled by default for all systems. No further configuration is needed.
	Fax pass-through	Yes	Yes	Yes	—
	Modem passthrough	Yes	Yes	Yes	The Cisco Unified Border Element display may not display the codec upshift (G.729 to G.711).
	Modem relay	No	No	No	—
	Fax with Transcoding	Yes	Yes	No	—

Table 2 Cisco Unified Border Element Call Signaling Features

Feature	Details	H.323-to-H.323 Support?	H.323-to-SIP Support?	SIP-to-SIP Support?	Additional Information
	T.38 fax relay (flow-through)	Yes	Yes	Yes	<ul style="list-style-type: none"> TCP/UDP like-to-like transport: Yes Standards OLC: Yes Cisco Proprietary NSE: No
Lawful Intercept					
	Lawful intercept	Yes	Yes	Yes	See Table 9 and Table 10 in this chapter for a list of supported platforms.
Media Inactivity Timer					
	RTCP media inactivity timer	Yes	Yes	Yes	—
Media Modes					
	Media Flow Through	Yes	Yes	Yes	—
	Media Flow Around	Yes	No	Yes	Note SIP-to-SIP support is limited to basic audio calls.
Other Features					
	IP address bind	Yes	Yes	Yes	Interface can be bound to only one protocol type.
	Session refresh with OPTIONS	No	No	Yes	—
	Media Statistics on an Cisco UBE	Yes	Yes	Yes	—
	SIP Error Message Pass Through	No	No	Yes	—
Protocol Compliance					
	H.323 v4	Yes	Yes	No	—
	SIP v2	No	Yes	Yes	—
Quality of Service					
	ToS/DSCP marking support	Yes	Yes	Yes	—
Rotary Support					
	Call Failure Recovery (Rotary)	Yes	No	Yes	SIP-to-SIP calls must have same codec.
	EmptyCapability (TCS=0)	Yes	No	No	TCS=0 message is transparently transferred from leg to leg.
Security					
	CryptoToken - IRR	Yes	No	No	—
	H235CallSecurity	Yes	No	No	Tokens are not transferred from leg to leg. A security token cannot be generated for only one leg (for example, only on the outgoing leg).

Table 2 Cisco Unified Border Element Call Signaling Features

Feature	Details	H.323-to-H.323 Support?	H.323-to-SIP Support?	SIP-to-SIP Support?	Additional Information
	IPSEC	Yes	Yes	Yes	—
	Secure RTP with IPSEC for Signaling	Yes	No	No	—
	SRTP	Yes	No	No	—
	Transport Layer Security (TLS)	No	Yes	Yes	—
Signaling Interworking					
	Delayed Media to Delayed Media	No	No	Yes	—
	Delayed Media to Slow Start	No	No	No	—
	Early Media to Early Media	n/a	n/a	Yes	Invite with SDP parameters.
	Fast Start to Delay Media	No	No	No	—
	Fast Start to Fast Start	Yes	n/a	n/a	Fast start elements are sent in PROG or ALERT and not in CALLPROC.
	Slow Start to Delayed Media	No	Yes	No	—
	Slow Start to Early Media	No	No	No	—
	Slow Start to Fast Start	Yes	No	No	Support for basic calls
	Slow Start to Slow Start	Yes	No	No	—
	Progress indicator interworking for media cut-through	Yes	No	No	—
	Tunneled H.245 traffic	Yes	No	No	—
Supplementary Services (Including Cisco Unified Communications Manager)					
	Call Forward	Yes	No	Yes	H323:H450.3, SIP:302
	Call Hold/Resume	Yes	No	Yes	SIP: Reinvite
	ECS to ReINVITE on the Cisco IOS SBC.	No	Yes	Yes	—
	ECS to REFER on the Cisco IOS SBC.	No	Yes	Yes	—
	Call Transfer	Yes	No	Yes	H323:H450.2, SIP:Refer
	Call Waiting	No	No	Yes	—
	Distinctive Ringing	No	No	Yes	—
	Message Waiting Indication (MWI)	Yes	No	Yes	—
	Music on Hold	Yes	Yes	Yes	Not locally generated on Cisco Unified Border Element.

TCL IVR

Table 2 Cisco Unified Border Element Call Signaling Features

Feature	Details	H.323-to-H.323 Support?	H.323-to-SIP Support?	SIP-to-SIP Support?	Additional Information
	IVR with DTMF SIP NOTIFY, RFC 2833	No	Yes	Yes	—
	IVR with H.245 alphanumeric, H.245 signal, RFC 2833	Yes	Yes	n/a	—
Timeouts					
	H.225 configurable timeout	Yes	No	No	—
Transport Protocols					
	UDP	Yes	Yes	Yes	H.323-to-H.323 and H.323-to-SIP connections require a GK.
	TCP	Yes	Yes	Yes	—
	Interworking UDP and TCP Transport	No	No	Yes	—
Voice and Video Calls					
	Voice	Yes	Yes	Yes	—
	Video	Yes	No	No	—
VoiceXML					
	VXML standard 3.x support	No	Yes	No	—
	VXML with DTMF SIP NOTIFY, RFC 2833	n/a	Yes	Yes	—
	VXML with H.245 alphanumeric H.245 signal, RFC 2833	Yes	Yes	n/a	—

Cisco Unified Border Element Protocol Features

Table 3 provides an at-a-glance guide to Cisco Unified Border Element Protocol features and the Cisco Border Element Software version the features are introduced.

Table 3 Cisco UBE Feature List and Software Release Version

Protocol	Feature	Cisco Unified Border Element Software Version
SIP	Session Refresh via REINVITE	Cisco Unified BE 1.1
	SIP Trunk Register	Cisco Unified BE 1.1
	Configurable SIP Listen Port	Cisco Unified BE 1.1
	Delayed Offer to Early Offer — Audio	Cisco Unified BE 1.1
	Delayed Offer to Early Offer — Video	Cisco Unified BE 1.2
	SIP Video support in SDP— Flow-through	Cisco Unified BE 1.1
	SIP Video support in SDP— Flow-around	Cisco Unified BE 1.2

Table 3 Cisco UBE Feature List and Software Release Version (continued)

Protocol	Feature	Cisco Unified Border Element Software Version
SIP continued	SIP-SIP Call Parameter Changes via REINVITES	Cisco Unified BE 1.2
	Bandwidth per dial-peer (b=AS:xx) support in SDP	Cisco Unified BE 1.2
	SIP Profiles (Header Manipulation)	Cisco Unified BE 1.2
	Map ISDN Facility to SIP INFO	Cisco Unified BE 1.2
	Adjustable Timers for Registration Refresh and Retries	Cisco Unified BE 1.3
	Configurable SIP Parameters via DHCP	Cisco Unified BE 1.3
	Forced Update of SIP Parameters via DHCP	Cisco Unified BE 1.3
	Out-of-dialog OPTIONS Ping to Monitor Dial-peers to Specified SIP Servers and Endpoints	Cisco Unified BE 1.3
	PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco Unified Border Element	Cisco Unified BE 1.3
	Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information	Cisco Unified BE 1.3
	Selective Filtering of Outgoing Provisional Response	Cisco Unified BE 1.3
	Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers	Cisco Unified BE 1.3
	SRTP-RTP Internetworking	Cisco Unified BE 1.3
	Unsupported Content Pass-through	Cisco Unified BE 1.3
H.323	H.235 CryptoToken Passthrough	Cisco Unified BE 1.1
	H.239 Passthrough (PIP)	Cisco Unified BE 1.1
	GK ARQ, RRQ and AAA Security Enhancements	Cisco Unified BE 1.1
	GK LRQ Multicast	Cisco Unified BE 1.1
	GK IZCT Enhancement	Cisco Unified BE 1.2

Protocol Interworking

Table 4 shows a list of protocol interworking support.

Table 4 Supported protocol Interworking

Protocol	In Leg	Out Leg	Support
H.323-to-H.323	Fast Start	Fast Start	Bi-Directional
	Slow Start	Slow Start	Bi-Directional
	Fast Start	Slow Start	Bi-Directional
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

Table 4 Supported protocol Interworking (continued)

Protocol	In Leg	Out Leg	Support
	Delayed Offer	Delayed Offer	Bi-Directional
	Delayed Offer	Early Offer	Uni-Directional

Billing Features

Table 5 shows a list of supported billing features.

Table 5 Supported Billing Features

Feature	Supported?	Additional Information
AAA accounting on gateway	Yes	AAA accounting supported on Cisco Unified Border Element. Authentication and authorization supported using available call information (ANI or DNIS) or fixed passwords only. Digit collection for prepaid calling card applications is not supported.
Billing token in unsolicited IRR	Yes	—
Call start time in IRR	Yes	—
Open Settlement Protocol (OSP)	Yes	Cisco Unified Border Element with OSP requires a separate feature license and a separate Cisco IOS image with encryption capabilities.
Per-interface billing	Yes	—

Application and Tcl Script Features

Table 6 shows a list of supported application and Tcl script features.

Table 6 Supported Application and Tcl Script Features

Feature	Supported?	Additional Information
IP call leg IVR	Yes	—
Tcl scripts	Yes	—
VXML session application	Yes	—

Interoperability Features

Table 7 shows a list of supported interoperability features.

Table 7 Supported Interoperability Features

Feature	Supported?	Additional Information
BroadSoft	Yes	First supported in 12.4(6)T images.
Cisco ATA 186	Yes	—

Table 7 Supported Interoperability Features

Feature	Supported?	Additional Information
Cisco ATA 188	Yes	First supported in 12.3(7)T images.
Cisco Unified Communications Manager	Yes	—
Cisco CallManager Express	Yes	—
Cisco gateways	Yes	Compatible with H.323 version 2 and above.
Cisco MCM Proxy	Yes	Cannot register proxy in the same zone as an Cisco Unified Border Element.
Third-party gatekeepers	Yes	Third-party gatekeepers must support the equivalent of via-zone functionality.
Third-party gateways	Partially	First supported in 12.3(7)T images.

IVR Features

Table 8 shows a list of supported IVR features.

Table 8 Supported IVR Features

Feature	Supported?	Additional Information
TCL IP-IP	Partially	<ul style="list-style-type: none"> TCL Verbs: Yes TDM related: No
VXML IP-IP	Partially	<ul style="list-style-type: none"> TCL Verbs: Yes TDM related: No

Lawful Intercept Support

Lawful Intercept (LI) is the term used to describe the process by which law enforcement agencies conduct electronic surveillance of circuit communications as authorized by judicial or administrative order. Cisco Service Independent Intercept (SII) supports voice and data intercept and intercept requests are initiated by MD using SNMPv3.

Table 9 and Table 10 provide quick reference to platforms and images that support lawful intercept

Table 9 TDM Gateway Lawful Intercept Support and Related Images

Platform	H.323	SIP	Dial	First Cisco IOS Release	Image
AS5350	Yes	Yes	Yes	12.3(14)T	c5350-ik9su2-mz
AS5400	Yes	Yes	Yes	12.3(14)T	c5400-jk9su2-mz
C2851	Yes	Yes	No	12.4(11)XJ2	c2800nm-adventerprisek9_ivs_li-mz
C3845	Yes	Yes	No	12.4(11)XJ2	c3845-adventerprisek9_ivs_li-mz
C72xx	Yes	Yes	Yes	12.4(6)T	c7200-advipservicesk9_li-mz
C73xx	Yes	Yes	Yes	12.4(6)T	c7300-advipservicesk9_li-mz

Table 10 Cisco Unified Border Element Gateway Lawful Intercept Support and Related Images

Platform	H.323	SIP	Dial	First Cisco IOS Release	Image
AS5350	Yes	Yes	Yes	12.3(14)T	c5350-ik9su2_ivs-mz
AS5400	Yes	Yes	No	12.3(14)T	c5400_jk9su2_ivs-mz
C2851	Yes	Yes	Yes	12.4(11)XJ2	c2800nm-adventerprisek9_ivs_li-mz
C3825	Yes	Yes	Yes	12.4(15)XY	c3825-adventerprisek9_ivs_li-mz
C3845	Yes	Yes	Yes	12.4(11)XJ2	c3845-adventerprisek9_ivs_li-mz
C72xx	Yes	Yes	Yes	12.4(6)T	c7200-advipservicesk9_li-mz
C73xx	Yes	Yes	Yes	12.4(6)T	c7300-advipservicesk9_li-mz

QoS Features

Table 11 shows a list of supported quality-of-service (QoS) features.

Table 11 Supported Quality of Service Features

Feature	Supported?
Class-based weighted fair queueing (LLQ)	Yes
Custom queueing	Yes
Differentiated services code point (DSCP)	Yes
IP precedence	Yes
Link fragmentation and interleaving (LFI)	Yes
Priority-queue weighted fair queueing (PQWFQ)	Yes
RTP header compression	Yes

Tcl Objects Supported by the Cisco Unified Border Element

The Cisco Unified Border Element supports all current Cisco IOS Tcl functions except those that are required to support IVR as defined: Tone generation

Table 12 through Table 15 list the Tcl commands, information tags, events, and status codes, respectively, that are supported by the Cisco Unified Border Element. Those listed as unsupported may function partially or incorrectly, and therefore their use is not recommended.



Note

For a complete list of Tcl commands, see the *Tcl IVR API Version 2.0 Programming Guide* at http://www.cisco.com/en/US/products/sw/voicew/ps2192/products_programming_reference_guide_book09186a00800de1de.html

Table 12 *Tcl Commands Supported by the Cisco Unified Border Element*

Command	Supported?	Command	Supported?
aaa accounting	Yes	aaa authenticate	Yes
aaa authorize	Yes	call close	Yes
clock	Yes	command terminate	Yes
connection create	Yes	connection destroy	Yes
fsm define	Yes	fsm setstate	Yes
handoff appl	Yes	handoff callappl	Yes
handoff return	Yes	infotag get	Yes
infotag set	Yes	leg collectdigits	Yes
leg connect	Yes	leg disconnect	Yes
leg proceeding	Yes	leg setup	Yes
leg setupack	Yes	leg vxmldialog	No
leg vxmlsend	No	media pause	No
media play	Yes	media resume	No
media seek	No	media stop	No
playtone	Yes	puts	Yes
requiredversion	Yes	set avsend	Yes
set callinfo	Yes	timer left	Yes
timer start	Yes	timer stop	Yes

Table 13 *Tcl Events Supported by the Cisco Unified Border Element*

Event	Supported?	Event	Supported?
ev_any_event	Yes	ev_authorize_done	Yes
ev_authenticate_done	Yes	ev_call_timer0	Yes
ev_collectdigits_done	Yes	ev_create_done	Yes
ev_destroy_done	Yes	ev_digit_end	Yes
ev_disconnect_done	Yes	ev_disconnected	Yes
ev_grab	Yes	ev_hookflash	No
ev_handoff	Yes	ev_leg_timer	Yes
ev_media_done	Yes	ev_returned	Yes
ev_setup_done	Yes	ev_setup_indication	Yes
ev_vxmldialog_done	No	ev_vxmlsend_event	No

Table 14 Tcl Information Tags Supported by the Cisco Unified Border Element

Information Tag	Supported?	Information Tag	Supported?
aaa_avpair	Yes	aaa_avpair_exists	Yes
aaa_new_guid	Yes	cfg_avpair	Yes
cfg_avpair_exists	Yes	con_all	Yes
con_ofleg	Yes	evt_connections	Yes
evt_dcdigits	Yes	evt_digit	Yes
evt_digit_duration	Yes	evt_event	Yes
evt_handoff_string	Yes	evt_iscommand_done	Yes
evt_legs	Yes	evt_status	Yes
evt_vxml_params	No	evt_vxmlelement	No
last_command_handle	Yes	leg_all	Yes
leg_ani	Yes	leg_ani_pi	Yes
leg_ani_si	Yes	leg_cdi_nso	Yes
leg_cdi_rr	Yes	leg_chn_noa	Yes
leg_chn_npi	Yes	leg_chn_num	Yes
leg_cid_cid	Yes	leg_cid_ton	Yes
leg_cnn_noa	Yes	leg_cnn_npi	Yes
leg_cnn_num	Yes	leg_cnn_pi	Yes
leg_cnn_si	Yes	leg_cpc	Yes
leg_dnis	Yes	leg_fdc_dat	Yes
leg_fdc_fname	Yes	leg_fdc_instr	Yes
leg_fdc_param	Yes	leg_gea_cni	Yes
leg_gea_noa	Yes	leg_gea_npi	Yes
leg_gea_num	Yes	leg_gea_pi	Yes
leg_gea_si	Yes	leg_gea_type	Yes
leg_guid	Yes	leg_incoming	Yes
leg_incomming_guid	Yes	leg_inconnection	Yes
leg_isdid	Yes	leg_ocn_noa	Yes
leg_ocn_npi	Yes	leg_ocn_pi	Yes
leg_oli	Yes	leg_outgoing	Yes
leg_password	Yes	leg_pci_dat	Yes
leg_pci_instr	Yes	leg_pci_tri	Yes
leg_rdn_pi	Yes	leg_rdn_si	Yes
leg_redirect_cnt	Yes	leg_redirect_cnt	Yes
leg_remoteipaddress	Yes	leg_rgn_noa	Yes
leg_rgn_npi	Yes	leg_rgn_num	Yes
leg_rgn_pi	Yes	leg_rgn_si	Yes

Table 14 *Tcl Information Tags Supported by the Cisco Unified Border Element (continued)*

Information Tag	Supported?	Information Tag	Supported?
leg_rni_orr	Yes	leg_rni_rc	Yes
leg_rni_ri	Yes	leg_rni_rr	Yes
leg_rnn_inn	Yes	leg_rnn_noa	Yes
leg_rnn_npi	Yes	leg_rnn_num	Yes
leg_rnr	Yes	leg_settlement_time	Yes
leg_suppress_outgoing_auto_acct	Yes	leg_tns_cc	Yes
leg_tns_ton	Yes	leg_username	Yes
med_backup_server	No	med_language	No
med_language_map	No	med_location	No
med_total_languages	No	sys_version	Yes

Table 15 *Tcl Status Codes Supported by the Cisco Unified Border Element*

Status Code	Supported?	Status Code	Supported?
Authentication Status	Yes	Authorization Status	Yes
Media Status	Yes	Leg Setup Status	Yes
Digit Collection Status	Yes	Disconnect Cause	Yes
VoiceXML Dialog Completion	No		

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. These features include, but are not limited to, the following:

- Disable secondary dial tone on voice ports—By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.
- Cisco router access control lists (ACLs)—Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized Session Initiation Protocol (SIP) or H.323 calls from unknown parties to be processed and connected by the router or gateway.
- Close unused SIP and H.323 ports—If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplex (TDM) trunks or IP, close the unused H.323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.

- Change SIP port 5060—If SIP is actively used, consider changing the port to something other than well-known port 5060.
- SIP registration—If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.
- SIP Digest Authentication—If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.
- Explicit incoming and outgoing dial peers—Use explicit dial peers to control the types and parameters of calls allowed by the router, especially in IP-to-IP connections used on CME, SRST, and Cisco UBE. Incoming dial peers offer additional control on the sources of calls, and outgoing dial peers on the destinations. Incoming dial peers are always used for calls. If a dial peer is not explicitly defined, the implicit dial peer 0 is used to allow all calls.
- Explicit destination patterns—Use dial peers with more granularity than T for destination patterns to block disallowed off-net call destinations. Use class of restriction (COR) on dial peers with specific destination patterns to allow even more granular control of calls to different destinations on the PSTN.
- Translation rules—Use translation rules to manipulate dialed digits before calls connect to the PSTN to provide better control over who may dial PSTN destinations. Legitimate users dial an access code and an augmented number for PSTN for certain PSTN (for example, international) locations.
- Tcl and VoiceXML scripts—Attach a Tcl/VoiceXML script to dial peers to do database lookups or additional off-router authorization checks to allow or deny call flows based on origination or destination numbers. Tcl/VoiceXML scripts can also be used to add a prefix to inbound DID calls. If the prefix plus DID matches internal extensions, then the call is completed. Otherwise, a prompt can be played to the caller that an invalid number has been dialed.
- Host name validation—Use the “permit hostname” feature to validate initial SIP Invites that contain a fully qualified domain name (FQDN) host name in the Request Uniform Resource Identifier (Request URI) against a configured list of legitimate source hostnames.
- Dynamic Domain Name Service (DNS)—If you are using DNS as the “session target” on dial peers, the actual IP address destination of call connections can vary from one call to the next. Use voice source groups and ACLs to restrict the valid address ranges expected in DNS responses (which are used subsequently for call setup destinations).

For more configuration guidance, see the [“Cisco IOS Unified Communications Toll Fraud Prevention”](#) paper.

Where to Go Next

- [Fundamental Cisco Unified Border Element Configuration](#)

Additional References

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



Note

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

Related Documents

Related Topic	Document Title
Cisco IOS commands	Cisco IOS Master Commands List, All Releases
Cisco IOS Voice commands	Cisco IOS Voice Command Reference
Cisco IOS Voice Configuration Library	For more information about Cisco IOS voice features, including feature documents, and troubleshooting information—at http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm
Cisco IOS Release 15.0	Cisco IOS Release 15.0 Configuration Guides
Cisco IOS Release 12.4	<ul style="list-style-type: none"> • Cisco IOS Release 12.4 Configuration Guides • Cisco IOS Release 12.4T Configuration Guides
Cisco IOS Release 12.3	<ul style="list-style-type: none"> • Cisco IOS Release 12.3 documentation • Cisco IOS Voice Troubleshooting and Monitoring Guide • Tcl IVR Version 2.0 Programming Guide
Cisco IOS Release 12.2	Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2
DSP documentation	High-Density Packet Voice Feature Card for Cisco AS5350XM and AS5400XM Universal Gateways http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/vfc_dsp.html
GKTMP (GK API) Documents	<ul style="list-style-type: none"> • <i>GKTMP Command Reference:</i> http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_cli.htm • <i>GKTMP Messages:</i> http://www.cisco.com/en/US/docs/ios/12_2/gktmp/gktmpv4_2/gk_tmp.html

Related Topic	Document Title
internet Low Bitrate Codec (iLBC) Documents	<ul style="list-style-type: none"> • Codecs section of the Dial Peer Configuration on Voice Gateway Routers Guide http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_ovrvw.html • Dial Peer Features and Configuration section of the Dial Peer Configuration on Voice Gateway Routers Guide http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_config.html
Cisco Unified Border Element Configuration Examples	<ul style="list-style-type: none"> • Local-to-remote network using the IPIPGW http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00801b0803.shtml • Remote-to-local network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edc.shtml • Remote-to-remote network using the IPIPGW: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edd.shtml • Remote-to-remote network using two IPIPGWs: http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a0080203edb.shtml
Related Application Guides	<ul style="list-style-type: none"> • Cisco Unified Communications Manager and Cisco IOS Interoperability Guide • Cisco IOS Fax, Modem, and Text Support over IP Configuration Guide • “Configuring T.38 Fax Relay” chapter • Cisco IOS SIP Configuration Guide • Cisco Unified Communications Manager (CallManager) Programming Guides • Quality of Service for Voice over IP
Related Platform Documents	<ul style="list-style-type: none"> • Cisco 2600 Series Multiservice Platforms • Cisco 2800 Series Integrated Services Routers • Cisco 3600 Series Multiservice Platforms • Cisco 3700 Series Multiservice Access Routers • Cisco 3800 Series Integrated Services Routers • Cisco 7200 Series Routers • Cisco 7301
Related gateway configuration documentation	<p>Media and Signaling Authentication and Encryption Feature for Cisco IOS H.323 Gateways.</p> <p>http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/htsecure.htm</p>

Related Topic	Document Title
Cisco IOS NAT Configuration Guide, Release 12.4T	<p><i>Configuring Cisco IOS Hosted NAT Traversal for Session Border Controller</i></p> <p>http://www.cisco.com/en/US/docs/ios/12_4t/ip_addr/configuration/guide/htnatsbc.html</p>
Troubleshooting and Debugging guides	<ul style="list-style-type: none"> • Cisco IOS Debug Command Reference, Release 12.4 at http://www.cisco.com/en/US/docs/ios/debug/command/reference/db_book.html • <i>Troubleshooting and Debugging VoIP Call Basics</i> at http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml • <i>VoIP Debug Commands</i> at http://www.cisco.com/en/US/docs/routers/access/1700/1750/software/configuration/guide/debug.html

Standards

Standard	Title
H.323 Version 4 and earlier	<i>H.323 (ITU-T VOIP protocols)</i>
H.323 - H.245 Version 12, Annex R	<i>H.323 (ITU-T VOIP protocols)</i>

MIBs

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

RFCs

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

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

Technical Assistance

Description	Link
<p>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</p> <p>To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds.</p> <p>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</p>	http://www.cisco.com/cisco/web/support/index.html

Feature Information for Cisco Unified Border Element Configuration Guide

Table 16 lists the features in this module and provides links to specific configuration information. Only features that were introduced or modified in Cisco IOS Release 12.2(13)T3 or a later release appear in the table.


Note

Table 16 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 16 Feature Information for Cisco Unified Border Element Overview

Feature Name	Release	Feature Information
ATA-188 Interoperability	12.3(7)T	This feature was introduced.
Cisco UBE Image Consolidation	12.3(7)T	This feature was introduced.
Cisco UBE MIB support	15.0(1)XA	This feature was introduced.
H.323 Video Calls Support for H.235 Security	12.4(15)XY	This feature was introduced.
H.323 Video Calls Support for H.239 Signaling	12.4(15)XY	This feature was introduced.
Interworking of Secure RTP calls for SIP and H.323	12.4(15)XY	This feature was introduced.
Lawful Intercept	12.4(6)T 12.4(11)XJ2 12.4(15)XY	12.3(14)T—Support for lawful intercept was introduced on the Cisco AS5350 and Cisco AS5400 12.4(6)T—Support was added for the Cisco 7200 and Cisco 7300. 12.4(11)XJ2—Support was added for the Cisco 2851 and Cisco 3845. 12.4(15)XY—Support was added for the Cisco 3825.
Support for Cisco 7200 and Cisco 7301	12.3(8)T	This feature was introduced.
Support for the Cisco 2801	12.4(4)T	This feature was introduced.
Support for the Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3825, and Cisco 3845	12.3(11)T	This feature was introduced.

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