Cisco Unified Border Element Features Roadmap

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This roadmap lists the features documented in the *Cisco Unified Border Element Configuration guide* (previously known as the *Cisco Multiservice IP-to-IP Gateway Application Guide*) and maps them to the chapters in which they appear.

Activation Before you can configure the software features described in this guide, you will need a Product Authorization Key (PAK). Before you start the configuration process, please register your products and activate your PAK at the following URL http://www.cisco.com/go/license.

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

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

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

Note Table 1 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

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Table 1 Supported Cisco Unified Border Element Configuration Guide Features

Release	Feature Name	Feature Description	Where Documented
Cisco IOS Releases 12.2T, 12.3, 12.3T, 12.4, and 12.4T			
15.0(1)XA	Support for Interworking Between RSVP Capable and RSVP Incapable Networks	The Support for Interworking Between RSVP Capable and RSVP Incapable Networks feature provides precondition-based RSVP support for basic audio call and supplementary services on the Cisco UBE.	"Configuring Support for Interworking Between RSVP Capable and RSVP Incapable Networks" in the "H.323-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP to SIP Calls	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.	"Configuring Support for Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls Feature" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure	This feature provides option to configure the error response code when a dial peer is busied out because of an Out-of-Dialog OPTIONS ping failure.	"Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Cisco UBE MIB support	Supports CISCO-VOICE-DIAL-CONTROL-MIB objects to obtain call volume and call rate information and CISCO-DSP-MGMT-MIB objects to report transcoding sessions availability information on the Cisco Unified Border Element.	"Related Documents" section of the, "H.323-to-H.323 Connections on a Cisco Unified Border Element", "H.323-to-SIP Connections on a Cisco Unified Border Element" and the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
15.0(1)M	Adjustable Timers for Registration Refresh and Retries	This feature provides the ability for 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.	"Configuring Adjustable Timers for Registration Refresh and Retries" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Cisco Unified Border Element Support for SRTP-RTP Internetworking	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.	"Cisco Unified Border Element Support for SRTP-RTP Internetworking" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Configurable SIP Parameters via DHCP	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP.	"Configurable SIP Parameters via DHCP" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Forced Update to SIP Parameters via DHCP updated in FTS.	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP.	"Enabling Forced Update of SIP Parameters via DHCP" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Out-of-dialog OPTIONS Ping to Monitor Dial-peers to Specified SIP Servers and Endpoints	This feature provides a keepalive mechanism at the SIP level between any number of destinations. The generic heartbeat mechanism allows Cisco UBE to monitor the status of SIP servers or endpoints and provide the option of busying-out associated dial-peer upon total heartbeat failure.	"Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information	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	"Support for Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information"in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers	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. The system also supports appending the phone-context to the tel: URL.	"Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco UBE	This feature provides the ability to: • Translate P-headers from one type to another. • configure the privacy header and the use of the PCPID header to route INVITE messages. • Supports multiple PAURI headers in the response messages (200 OK) it receives to REGISTER messages.	"Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco Unified Border Element" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Unsupported Content Pass-through	The 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.	"Configuring Cisco Unified Border Element for Unsupported Content Pass-through" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
12.4(24)T	SIP Registration Message	Provides the ability to send a SIP Registration Message from Cisco Unified Border Element using the **credentials** command.	"Configuring Cisco Unified Border Element for Unsupported Content Pass-through" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
12.4(22)YB	Adjustable Timers for Registration Refresh and Retries	This feature provides the ability for 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.	"Configuring Adjustable Timers for Registration Refresh and Retries" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Cisco Unified Border Element Support for SRTP-RTP Internetworking	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.	"Cisco Unified Border Element Support for SRTP-RTP Internetworking" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Configurable SIP Parameters via DHCP	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP.	"Configurable SIP Parameters via DHCP" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Forced Update to SIP Parameters via DHCP updated in FTS.	The Configurable SIP Parameters via DHCP feature introduces the configuring of SIP parameters via DHCP. The following sections provide information about this feature:	"Enabling Forced Update of SIP Parameters via DHCP" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Out-of-dialog OPTIONS Ping to Monitor Dial-peers to Specified SIP Servers and Endpoints	This feature provides a keepalive mechanism at the SIP level between any number of destinations. The generic heartbeat mechanism allows Cisco UBE to monitor the status of SIP servers or endpoints and provide the option of busying-out associated dial-peer upon total heartbeat failure.	"Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information	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	"Support for Preloaded Routes in Outgoing INVITE Messages Based on REGISTER Information"in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Element	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.	"Configuring Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Element" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers	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. The system also supports appending the phone-context to the tel: URL.	"Selectively Using sip: URI or tel: URL Formats on Individual SIP Headers" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco UBE	This feature provides the ability to: • Translate P-headers from one type to another. • configure the privacy header and the use of the PCPID header to route INVITE messages. • Supports multiple PAURI headers in the response messages (200 OK) it receives to REGISTER messages.	"Support for PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco Unified Border Ele ment" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.
	Unsupported Content Pass-through	The 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.	"Configuring Cisco Unified Border Element for Unsupported Content Pass-through" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide.

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
12.4(20)T1	Delayed offer to Early offer for SIP Video Calls	Forces a Cisco Unified Border Element to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL).	"Configuring Delayed-Offer to Early-Offer for SIP Video Calls" in the "Configuring Cisco Unified Border Element Videoconferencing" chapter of this guide
12.4(20)T	Configurable SIP Listening Port	Allows users the ability to configure the port that SIP messages are listened on.	"Configuring SIP Listening Port" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Configuring Bandwidth Parameters for SIP Calls	This features provides the ability to manually configure the bandwidth that is signaled in the outbound SIP invite.	"Configuring Bandwidth Parameters for SIP Calls" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Product Authorization Key (PAK)	Requires users to register products and activate a Product Authorization Key (PAK) before starting the configuration process. **Note** Register Products and activate your PAK at the following URL http://www.cisco.com/go/license	—
	H.323 Calling Name Display	Provides a configurable option on the Cisco Gateway to send and interpret the calling name information received in Q931 Facility messages so that the Cisco Unified Communications Manager can display the calling name on the Cisco IP Phones.	"Configuring H.323 Calling Name Display" in the "Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity" chapter of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Session Border Controller Enhancements for H.323-to-SIP and SIP-to-SIP Supplementary Services, Transcoding Optimization and Firewall Integration.	New H.323-to-SIP features offered in this release include: • Supplementary Services specifically mapping ECS to ReINVITE and ECS to REFER on the Cisco IOS SBC. New SIP-to-SIP features offered in this release include: • Supplementary Services mapping ReINVITE to ReINVITE are on the Cisco IOS SBC. New features offered in this release include: • Also available are enhancements in Transcoding Performance and support for Universal Transcoding Support • RAS message enhancements	"Overview of Cisco Unified Border Element", "H.323-to-SIP Connections on a Cisco Unified Border Element", and "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide
	Session Refresh with reinvites	This feature expands the ability of the Cisco Unified BE to receive a REINVITE that contains either a session refresh parameter or a change in media via a new SDP and ensure the session does not time out.	"Configuring Support for Session Refresh with Reinvites" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	SIP Video Calls with Flow Around Media	Allows SIP video calls where the media flows around the Cisco Unified Border Element.	"SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
12.4(15)XZ	Configurable SIP Parameters	Allows users to change the standard SIP messages sent from the Cisco SIP stack for better interworking with different SIP entities.	"Configuring SIP Parameters" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Product Authorization Key (PAK)	Requires users to register products and activate a Product Authorization Key (PAK) before starting the configuration process. **Note** Register Products and activate your PAK at the following URL http://www.cisco.com/go/license	—
	Configurable SIP Listening Port	Allows users the ability to configure the port that SIP messages are listened on.	"Configuring SIP Listening Port" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Configuring Bandwidth Parameters for SIP Calls	This features provides the ability to manually configure the bandwidth that is signaled in the outbound SIP invite.	"Configuring Bandwidth Parameters for SIP Calls" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Session Refresh with reinvites	This feature expands the ability of the Cisco Unified BE to receive a REINVITE that contains either a session refresh parameter or a change in media via a new SDP and ensure the session does not time out.	"Configuring Support for Session Refresh with Reinvites" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Delayed offer to Early offer for SIP Video Calls	Forces a Cisco Unified Border Element to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL).	"Configuring Delayed-Offer to Early-Offer for SIP Video Calls" in the "Configuring Cisco Unified Border Element Videoconferencing" chapter of this guide
	Session Border Controller Enhancements for H.323-to-SIP and SIP-to-SIP Supplementary Services, Transcoding Optimization and Firewall Integration.	New H.323-to-SIP features offered in this release include: • Supplementary Services specifically mapping ECS to ReINVITE and ECS to REFER on the Cisco IOS SBC. New SIP-to-SIP features offered in this release include: • Supplementary Services mapping ReINVITE to ReINVITE are on the Cisco IOS SBC. New features offered in this release include: • Also available are enhancements in Transcoding Performance and support for Universal Transcoding Support • RAS message enhancements	"Overview of Cisco Unified Border Element", "H.323-to-SIP Connections on a Cisco Unified Border Element", and "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide
	Ability to Send a SIP Registration Message on a Cisco Unified Border Element	This feature introduces a new command that allows the Cisco Unified Border Element to send a REGISTRATION command to a SIP REGISTRAR.	"Configuring SIP Listening Port" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
12.4(15)XY	—	The official marketing name of Cisco Multiservice IP-to-IP Gateway was changed to Cisco Unified Border Element (Cisco UBE).	—

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Interworking of Secure RTP calls for SIP and H323	New features offered in this release include: • CUCM SIP Trunks	"Overview of Cisco Unified Border Element"
	H.323 Video Calls Support for H.239 Signalling	New features offered in this release include: • Business to Business Telepresence calls	"Overview of Cisco Unified Border Element"
	H323 Video Calls Support for H.235 Security	New features offered in this release include: • Enhanced Security for France Telecom and Video calls to 3rd party endpoints.	"Overview of Cisco Unified Border Element"
	Delayed offer to Early offer for SIP Audio Calls	Forces a Cisco Unified Border Element to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL).	"Configuring Delayed-Offer to Early-Offer for SIP Audio Calls" in the "SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Lawful Intercept for Cisco 3825	Adds Lawful Intercept support on the Cisco 3825.	"Overview of Cisco Unified Border Element" chapter of this guide.
12.4(11)XW	H.323 Calling Name Display	Provides a configurable option on the Cisco Gateway to send and interpret the calling name information received in Q931 Facility messages so that the Cisco Unified Communications Manager can display the calling name on the Cisco IP Phones.	"Configuring H.323 Calling Name Display" in the "Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity" chapter of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
12.4(11)XJ2	Enhanced Hosted NAT Traversal and IP Call Leg Statistics 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 features offered in this release include: • Lawful Intercept for 2851 and 3845 • DTMF Transcoding and Interworking: – Transcoding with AS5xxx platforms New H.323-to-H.323 features offered in this release on the Cisco 28xx, 38xx, 5350XM and 5400XM include: • Media Statistics on an Cisco Unified Border Element New H.323-to-SIP features offered in this release on the Cisco 28xx, 38xx, 5350XM and 5400XM include: • Media Statistics on a Cisco UBE • H323 to SIP Codec Transparent Support • SIP to H323 Interworking • DTMF Transcoding and Interworking: – H245 to KPML New SIP-to-SIP features offered in this release on the Cisco 28xx, 38xx, 5350XM and 5400XM include: • Media Statistics on an Cisco UBE • SIP Error Message Pass Through • DTMF Transcoding and Interworking: – SIP to KPML	"Overview of Cisco Unified Border Element", "H.323-to-H.323 Connections on a Cisco Unified Border Element"", "H.323-to-SIP Connections on a Cisco Unified Border Element", "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide
12.4(11)T	H.323-to-SIP Supplementary Feature Interworking for Session Border Controller (SBC)	New H.323-to-H.323 features offered in this release include: • G.711 Inband DTMF to RFC 2833 • iLBC Codec Support New H.323-to-SIP features offered in this release include: • H.323 RFC 2833 to SIP NOTIFY • iLBC Codec Support • VXML support with SIP NOTIFY DTMF • TCL IVR support with SIP NOTIFY DTMF New SIP-to-SIP features offered in this release include: • iLBC Codec • Session refresh	"Fundamental Cisco Unified Border Element Configuration", "H.323-to-H.323 Connections on a Cisco Unified Border Element", "H.323-to-SIP Connections on a Cisco Unified Border Element", "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	DTMF Relay Digit-Drop on an Cisco Unified Border Element with Cisco Unified Communications Manager	This feature passes DTMF tones out-of-band and drops in-band digits to avoid sending both tones to the outgoing leg on an H.323-to-SIP Cisco UBE.	"Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity" chapter of this guide.
12.4(6)XE	H.323-to-SIP Supplementary Feature Interworking for Session Border Controller (SBC)	New H.323-to-SIP features available include: • Support H.323-to-SIP Supplementary services for Cisco Unified Communications Manager with MTP on the H.323 Trunk. • G.711 Inband DTMF to RFC 2833 • VXML 3.x support • VXML support with SIP Notify New SIP-to-SIP features offered in this release include: • G.711 Inband DTMF to RFC 2833	"H.323-to-SIP Connections on a Cisco Unified Border Element" and "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide
12.4(9)T	GSMAMR-NB Codec support on a Cisco Unified Border Element	Support for the complexity multimode codec that supports eight narrowband speech encoding modes with bit rates between 4.75 and 12.2 kbps.	"Fundamental Cisco Unified Border Element Configuration", chapter of this guide
	SIP-to-SIP Supplementary Services for Session Border Controller (SBC)	New SIP-to-SIP features available include: • SIP-to-SIP supplementary services using Refer • Hosted NAT Traversal for SIP • Provides integrated voice and video services on the Cisco AS5350XM and Cisco AS5400XM.	"SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	—	The Gatekeeper content from the Cisco Multiservice IP-to-IP Gateway Application guide was moved to a separate book located at the following http://cisco.com/en/US/docs/ios/voice/cubegk/configuration/guide/ve_book/ve_book.html	Cisco Unified Border Element with Gatekeeper Configuration Guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
12.4(6)T	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 • Media Inactivity • Modem passthrough • TCP and UDP interworking • Tcl scripts with SIP NOTIFY VoiceXML with SIP-to-SIP • Transport Layer Security (TLS) • ENUM support • Lawful Intercept • Interoperability with Cisco Unified Communications Manager 5.0 and BroadSoft	"SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	H.323-to-SIP Extended Call Interworking for Session Border Controller (SBC)	New H.323-to-SIP features available include: • Call Admission Control (based on CPU, memory, total calls)	"H.323-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	H.323-to-H.323 Extended Call Interworking for Session Border Controller (SBC)	New H.323-to-H.323 features available include: • Secure RTP with IPSEC for Signaling • No MTP for Cisco Unified Communications Manager Trunks to Cisco UBE • Call Admission Control (based on CPU, memory, total calls)	"H.323-to-H.323 Connections on a Cisco Unified Border Element" chapter of this guide
12.4(4)T	Interoperability Enhancements to the Cisco Unified Border Element	Enables operation of Cisco Unified Border Element features concurrently on the same router with H.323 gatekeeper and TDM-IP voice-gateway features.	"Fundamental Cisco Unified Border Element Configuration"chapter of this guide
	SIP-to-H.323 Extended Call Interworking for Session Border Controller (SBC)	Enables the Cisco UBE to bridge calls between networks that support different VoIP call-signaling protocols (SIP and H.323).	"H.323-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	SIP-to-SIP Basic Functionality for Session Border Controller (SBC)	Enables the Cisco UBE to bridge calls between SIP networks.	"SIP-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Support for Cisco UBE and Gatekeeper Features on the Cisco 2801	Provides integrated voice and video services on the Cisco 2801.	"Fundamental Cisco Unified Border Element Configuration" chapter of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
12.3(11)T	Scalability Enhancements for the Cisco 7301 + 7200 NPE-G1 (Extended Socket Boundary)	Increases the call capacity of the Cisco UBE by extending the total number of sockets supported on the Cisco 7301 and Cisco 7200 NPE-G1 routers.	Integrated into Cisco IOS software. No configuration is required.
	Support for Cisco Unified Border Element and Gatekeeper Features on the Cisco 2800 Series and Cisco 3800 Series	Provides integrated voice and video services on the Cisco 2800 series and Cisco 3800 series.	"Fundamental Cisco Unified Border Element Configuration" and chapter of this guide (see also Table 1 in the Overview of Cisco Unified Border Element chapter in this guide)
	SIP-to-H.323 Basic Call Interworking for Session Border Controller (SBC)	Enables the Cisco Unified Border Element to bridge calls between networks that support different VoIP call-signaling protocols (SIP and H.323).	"H.323-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	SIP-to-H.323 Dual Tone Multifrequency Relay Digit-Drop	Passes DTMF tones out-of-band and drops in-band digits to avoid sending both tones to the outgoing leg on H.323-to-SIP Cisco UBE.	"H.323-to-SIP Connections on a Cisco Unified Border Element" chapter of this guide
	Call Failure Recovery (Rotary) on the Cisco Unified Border Element	Eliminates codec restrictions and enables the Cisco UBE to restart codec negotiation with the originating endpoint based on the codec capabilities of the next dial peer in the rotary group for H.323-to-H.323 interconnections.	"H.323-to-H.323 Connections on a Cisco Unified Border Element" chapter of this guide
	H.323-to-H.323 Interworking Between FastStart and Normal H.245 Signaling	Enables the Cisco UBE to bridge calls between VoIP endpoints that support only H.323 FastStart procedures and endpoints that support only normal H.245 signaling (SlowStart).	"H.323-to-H.323 Connections on a Cisco Unified Border Element" chapter of this guide
	Transcoding G.711-G.729	Supports transcoding (compression and decompression of voice streams to match endpoint-device capabilities) between G.711 and G.729 codecs when the router chassis is equipped with DSP resources (H.323-H.323 and H.323-SIP).	"Fundamental Cisco Unified Border Element Configuration" chapter of this guide
12.3(7)T	Interoperability Enhancements to the Cisco Unified Border Element	Enables operation of Cisco UBE features concurrently on the same router with H.323 gatekeeper and TDM-IP voice-gateway features. Supports interoperability with the Cisco ATA-188 and with Microsoft NetMeeting.	"Fundamental Cisco Unified Border Element Configuration" chapter of this guide
12.3(4)T	Videoconferencing for the Cisco Unified Border Element Feature	Adds video capabilities and improved QoS, allowing increased scalability and control for IP telephony and IP videoconferencing networks.	"Configuring Cisco Unified Border Element Videoconferencing" chapter of this guide
12.3(1)	Cisco Unified Border Element for H.323 Cisco Unified Communications Manager to H.323 Service Provider Connectivity	Provides interoperability with Cisco Unified Communications Manager for basic calls, caller-ID services, supplementary services, and RSVP synchronization with audio.	"H.323-to-H.323 Connections on a Cisco Unified Border Element" chapter of this guide

Table 1 Supported Cisco Unified Border Element Configuration Guide Features (continued)

Release	Feature Name	Feature Description	Where Documented
	Cisco Unified Border Element with Media Flow-Around	Adds media flow-around capability on the Cisco UBE by supporting the processing of call setup and teardown requests (VoIP call signaling) and for media streams (flow-through and flow-around). Improves scalability and performance when network-topology hiding and bearer-level interworking features are not required.	How to Configuring Media Flow-Around sections of the, "H.323-to-H.323 Connections on a Cisco Unified Border Element", "H.323-to-SIP Connections on a Cisco Unified Border Element", and "SIP-to-SIP Connections on a Cisco Unified Border Element" chapters of this guide.
12.2(13)T3	H.323 Cisco Unified Border Element	Provides a network-to-network demarcation point between independent VoIP and video networks by for billing, security, call-admission control, QoS, and signaling interworking. Performs most of the functions of a PSTN-to-IP gateway but joins two H.323 VoIP call legs.	"Fundamental Cisco Unified Border Element Configuration" chapter of this guide