

Preface

This preface describes the objectives and organization of this guide and explains how to find additional information on related products and services. This preface contains the following sections:

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Guide Revision History

The Guide Revision History records technical changes to this guide. The table shows the software release number and guide revision number for the change, the date of the change, and a brief summary of the change.

Cisco IOS Release.	Part Number	Publication Date	Change Summary
Cisco IOS XE Release 3.11S	OL-19820-15	November, 2013	The following features were added: • Blended Transcoding
Cisco IOS XE Release 3.8S	OL-19820-14	November, 2012	The following features were added: • AMR-WB
Cisco IOS XE Release 3.7S	OL-19820-13	July, 2012	The following features were added:H.248 Border Access Controller SupportIMS Rf Billing Interfaces
Cisco IOS XE Release 3.6S	OL-19820-12	March 29, 2012	The following features were added:Common IP Address Media BypassVia Header Passthrough

Cisco IOS	OL-19820-11	November 28, 2011	The following features were added:
XE Release			Alarm-Related Enhancements
5.55			CAC-Related Enhancements
			Call Log Correlation
			• Flexible Media Routing
Cisco IOS	OL-19820-10	July 25, 2011	The following features were added:
XE Release			Limiting Resource Usage
5.45			QoS Demarcation Enhancements
			• SDP Editing Using Script-Based Editors
			• SRTP Support for RTCP Multiplexed with RTP and for SSRC-Based Multiplexing
Cisco IOS	OL-19820-09	March 18, 2011	The following features were added:
XE Release			• SIP Header Manipulation Enhancements
5.50			• Support for H.239
			• Voice Transcoding Per Adjacency Statistics
			• Message, Policy, and Subscriber Statistics Enhancements
			• SPA DSP: Call Recovery
			• Flow Statistics QoS Enhancements
			Selective Radius Billing
			Alternative Contact Rewriting
			BFCP Support
			• Limited H.323 ID Routing and Passthrough Support
			• Support to the Cisco ASR 1006 Series Router and Cisco ASR 1013 Series Router
			Interchassis-Intrachassis Conversion

Cisco IOS	OL-19820-08	November 24, 2010	The following features were added:
XE Release			SPA DSP Services
5.20			• Emergency and Security Enhancements
			- SIP trust model includes H.323 Interface
			 Emergency Call statistics
			• SBC Calls Support using IPSec Tunnels
			ASR1001 Support
			• XML based billing
			• SIP Interworking Enhancements
			- Event Header in Publish Method
			 Source Number Editing during Number analysis
			 Privacy Service
			- Option Ping Enhancements
			 Multiple SBC media bypass
			 Add Expires Header to Register Message
			 Absence of Username Support in Request URI
			• Analysis, Routing, and Policy Enhancements
			- Copy and Swap Procedure
			- Multiple CAC Averaging Periods
			- Administrative Domains
			 Blacklist Alerts
			Media Interworking Enhancements
			- MGX Assisted DTMF Interworking
			 Codec Preference and Re-Ordering
			 Per-Adjacency Codec String Interworking
			- Media Address Pool Support
			• PKI High Availability Support

Cisco IOS	OL-19820-07	July 30, 2010	The following features were added:	
XE Release			• IMS Rx and Diameter	
5.10			• ENUM Client feature	
			Customized System Error Messages	
			• SRTP to RTP Interworking and SRTP Passthrough	
			Media Bandwidth Policy	
			• SDP on 200 Invite	
			Memory Alerting	
			• SIP Destination ID and SIP Source ID	
			• Support for Asymmetric Payload Types	
			• IP IPv6/VRF Feature	
			• DTMF Method Interworking and ACCEPT Header Handling	
			CALEA IRI Interface Support feature	
			Redundant Peer Addresses	
			• Per Subscriber Delete	
Cisco IOS XE Release 2.6.2	OL-19820-06	July 08,2010	Endpoint information in PacketCable billing records was added.	
Cisco IOS XE Release 2.6.1	OL-19820-05	April, 2010	Adjacency information in PacketCable Billing Records was added.	
Cisco IOS XE Release 2.6	OL-19820-04	February 26, 2010	IPv6 support including IPv4 to IPv6 and IPv6 to IPv6 Interworking, Dynamic Codec Configuration, multiple audio and video codec support, H.323 support for Clear Channel calls, SIP-I Support and SIP Non-SDP Body Filtering, Unsignaled (granular-level) Secure Media, Configurable Mutual TLS Authentication per Interface, TLS Transport Parameter in Record-Router Header, Source Number Analysis, and Interoperability for SIP Authentication features were added.	
Cisco IOS XE Release 2.5.1	OL-19820-03	January 27, 2010	H.323 Extra TCS Codecs support was added.	

Cisco IOS XE Release 2.5	OL-19820-02	November 25, 2009	H.323 support, H.323-SIP interworking features, H.323 call routing, Transcoding support, multiple SIP features, 100rel interworking, SIP IP-FQDN URI translation, Contact Username Passthrough, IP Realm support, customized offer for late-to-early media, regular expression based routing, support for external server, call duration monitoring, signaling congestion handling, support for P-visited-network-ID, and other features were added in this release. See the Feature History Table in each chapter for supported features.
Cisco IOS XE Release 2.4	OL-19820-01	June 26, 2009	This guide introduced the unified model and a new unified feature set on the Cisco Unified Border Element (SP Edition). See the Feature History Table in each chapter for supported features. The name Cisco Unified Border Element (SP Edition) replaced the Integrated Session Border Controller name.

Objectives

This guide describes the Integrated Session Border Controller functions, features, restrictions, and configuration tasks for the Cisco ASR 1000 Series Aggregation Services Routers. It is not intended as a comprehensive guide to all of the software features that can be run using the Cisco ASR 1000 Series Routers, but only the Integrated Session Border Controller software specific to these Routers.

For information on general Cisco IOS software features that are also available on the Cisco ASR 1000 Series Routers, see the feature module or the technology guide for that software feature.

Intended Audience

This guide is intended for the following people:

- Experienced service provider administrators
- Cisco telecommunications management engineers
- Customers who use and manage Cisco ASR 1000 Series Routers

Organization

Section	Title	Description
Part 1	Basics	This part contains the following modules:
		• Using the Command-Line Interface in Cisco IOS XE Software
		• Cisco Unified Border Element (SP Edition) Overview
		• Configuring Cisco Unified Border Element (SP Edition)
		Media Address Pools
		• Implementing Multi-VRF on Cisco Unified Border Element (SP Edition)
		• Implementing Adjacencies on Cisco Unified Border Element (SP Edition)
		• Implementing Cisco Unified Border Element (SP Edition) Policies
		Call Duration Monitoring
		• IP Realm Support
		Managing Emergency Calls
Part 2	Service	This part contains the following modules:
		Unexpected Source Address Alerting
		• DoS Prevention and Dynamic Blacklisting
Part 3	Dual Tone Multifrequency	This part contains the following module:
	(DTMF)	• Implementing Interworking DTMF
Part 4	Redundancy-High Availability	This part contains the following modules:
		• Cisco Unified Border Element (SP Edition) Redundancy—High Availability Support
		• Interchassis High Availability
Part 5	Media	This part contains the following modules:
		Fax Support
		• Codec Handling, page 1
		• SDP Bandwidth Field Features
		• SDP Handling
		• Flexible Media Routing

This guide contains the following chapters and appendixes:

Section	Title	Description
Part 6	Session Initiation Protocol	This part contains the following modules:
	(SIP)	• Inherit Profiles for Non-IMS Adjacencies
		• Cisco Unified Border Element (SP Edition) Registration Features
		• SIP Message Manipulation
		Signaling Congestion Handling
		• SIP IP-FQDN URI Translation
		• SIP Tel URI Support
		• SIP Timer
		• SIP Configuration Flexibility
		SIP Renegotiation
		• 100rel Interworking Support
		Customized System Error Messages
		BFCP Support
Part 7	H.323	This part contains the following modules:
		H.323 Support
		• H.323 to SIP Interworking
		• Support for H.239
Part 8	Billing	This part contains the following modules:
		• Implementing Billing on Cisco Unified Border Element (SP Edition)
		Billing Support
Part 9	Secure Real-Time	This part contains the following module:
	Transport Protocol (SKTP)	• Secure Media and SRTP Passthrough
Part 10	Quality of Service (QoS)	This part contains the following module:
		• Implementing QoS (Marking)
Part 11	Transcoding	This part contains the following modules:
		Implementing Transcoding
		Cisco Unified Border Element (SP Edition)—SPA DSP Services
Part 12	Management and	This part contains the following modules:
	Operations	Tracking Policy Failure Statistics
		• Implementing SNMP
		Logging Support

Section	Title	Description
Part 13	Service	This part contains the following modules:
		• SIP 3xx Redirect Responses
		• SIP Call Hold
		• SIP Call Transfer
		• SIP Authentication
		• Late-to-Early Media Interworking
		• Early Media
		SIP Instant Messaging
		• Integration of Resource Management and SIP
		• ENUM Client
Part 14	IPv6	This part contains the following module:
		IPv6 Support
Part 15	IP Multimedia Subsystem	This part contains the following modules:
	(IMS)	P-CSCF Support
		IBCF Processing Support
		• IMS Rx, Diameter, and IMS Rf
Part 16	CALEA IRI Interface	This part contains the following module:
	Support	CALEA IRI Interface Support
Appendix	Appendix A	End-to-End Cisco Unified Border Element (SP
		Edition) Configuration Example
	Appendix B	SIP Compliance and Interoperability
	Appendix C	XML Billing Schema

Related Documentation

This section refers you to other documentation that might also be useful as you configure your Cisco ASR 1000 Series Routers. The documentation listed below is available on Cisco.com.

For information on Cisco Unified Border Element (SP Edition) commands, see the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at: http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information on the Cisco Unified Border Element (SP Edition) distributed model, see the:

- Cisco Unified Border Element (SP Edition) Configuration Guide: Distributed Model at: http://www.cisco.com/en/US/docs/routers/asr1000/configuration/guide/sbc/2_xe/sbc_2_xe_book.h tml
- *Cisco Unified Border Element (SP Edition) Command Reference: Distributed Model* at: http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbc_book.html

For information on the Cisco Unified Border Element (SP Edition) examples, see the Cisco Unified Border Element (SP Edition) Configuration Profile Examples at:

http://www.cisco.com/en/US/docs/routers/asr1000/profiles/SBC_Config_Examplebook.html

For other related command documentation, see the:

• Cisco IOS command reference books for the new Cisco ASR 1000 Series Router commands and commands in existing Cisco IOS features for this release at the following link:

http://www.cisco.com/en/US/products/ps9587/prod_command_reference_list.html

• Command Lookup Tool for information about Cisco IOS commands in general or a Cisco IOS master commands list at the following link:

http://tools.cisco.com/Support/CLILookup

For Quick Start guides and installation documentation for the Cisco ASR 1000 Series Router, see the hardware documentation that was provided as a part of this release at:

http://www.cisco.com/en/US/products/ps9343/prod_installation_guides_list.html

For information on new software features, see the:

• Cisco ASR 1000 Series Aggregation Services Routers Software Configuration Guide

http://www.cisco.com/en/US/docs/routers/asr1000/configuration/guide/chassis/asrswcfg.html

• Cisco IOS XE release notes

http://www.cisco.com/en/US/docs/ios/ios_xe/2/release/notes/rnasr21.html

For further information, see the Cisco ASR 1000 Series Aggregation Services Routers Documentation Roadmap at:

http://www.cisco.com/en/US/docs/routers/asr1000/roadmap/asr1000rm.html

Documentation for the Cisco IOS XE configuration guides and feature modules can be found at:

http://www.cisco.com/en/US/products/ps9587/tsd_products_support_configure.html

Conventions

This document uses	the	following	conventions:
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Convention	Indication	
bold font	Commands and keywords and user-entered text appear in bold font.	
<i>italic</i> font	Document titles, new or emphasized terms, and arguments for which you supply values are in <i>italic</i> font.	
[]	Elements in square brackets are optional.	
$\{x \mid y \mid z\}$	Required alternative keywords are grouped in braces and separated by vertical bars.	
$[x \mid y \mid z]$	Optional alternative keywords are grouped in brackets and separated by vertical bars.	
string	A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.	
courier font	Terminal sessions and information the system displays appear in courier font.	
< >	Nonprinting characters such as passwords are in angle brackets.	
[]	Default responses to system prompts are in square brackets.	
!, #	An exclamation point (!) or a pound sign (#) at the beginning of a line of code indicates a comment line.	



Means reader take note.

Tin

Means the following information will help you solve a problem.



Means *reader be careful*. In this situation, you might perform an action that could result in equipment damage or loss of data.



Means *the described action saves time*. You can save time by performing the action described in the paragraph.



Means *reader be warned*. In this situation, you might perform an action that could result in bodily injury.

Configuration Guides, Command References, and Supplementary Resources

Table 1 lists, in alphabetical order, Cisco IOS XE software configuration guides and command references, including brief descriptions of the contents of the documents. The command references contain commands for both Cisco IOS software and Cisco IOS XE software, for all releases. The command references support many different software releases and platforms. Your Cisco IOS XE software release or platform may not support all these technologies.

Table 2 lists documents and resources that supplement the Cisco IOS XE software configuration guides and command references. These supplementary resources include release notes and caveats; master command lists; new, modified, removed, and replaced command lists; system messages; and the debug command reference.

For additional information about configuring and operating specific networking devices, and to access Cisco IOS documentation, go to the Product/Technologies Support area of Cisco.com at the following location:

http://www.cisco.com/go/techdocs

Configuration Guide and Command Reference Titles		Features/Protocols/Technologies	
•	Cisco ASR 1000 Series Aggregation Services Routers SIP and SPA Software Configuration Guide	Configuration and troubleshooting of SPA interface processors (SIPs) and shared port adapters (SPAs) that are supported on the Cisco ASR 1000 Series Router.	
•	Cisco ASR 1000 Series Aggregation Services Routers Software Configuration Guide	Overview of software functionality that is specific to the Cisco ASR 1000 Series Aggregation Services Routers.	

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
• Cisco IOS XE Access Node Control Protocol Configuration Guide	Communication protocol between digital subscriber line access multiplexers (DSLAMs) and a broadband remote access server
• Cisco IOS Access Node Control Protocol Command Reference	(BRAS).
• Cisco IOS XE Asynchronous Transfer Mode Configuration Guide	LAN ATM, multiprotocol over ATM (MPoA), and WAN ATM.
• Cisco IOS Asynchronous Transfer Mode Command Reference	
• Cisco IOS XE Broadband Access Aggregation and DSL Configuration Guide	PPP over Ethernet (PPPoE).
• Cisco IOS Broadband Access Aggregation and DSL Command Reference	
 Cisco IOS XE Carrier Ethernet Configuration Guide Cisco IOS Carrier Ethernet Command Reference 	IEEE 802.3ad Link Bundling; Link Aggregation Control Protocol (LACP) support for Ethernet and Gigabit Ethernet links and EtherChannel bundles; LACP support for stateful switchover (SSO), in service software upgrade (ISSU), Cisco nonstop forwarding (NSF), and nonstop routing (NSR) on Gigabit EtherChannel bundles; and IEEE 802.3ad Link Aggregation MIB.
 Cisco IOS XE Configuration Fundamentals Configuration Guide Cisco IOS Configuration Fundamentals Command Reference 	Autoinstall, Setup, Cisco IOS command-line interface (CLI), Cisco IOS file system (IFS), Cisco IOS web browser user interface (UI), basic file transfer services, and file management.
Cisco IOS XE DECnet Configuration Guide	DECnet protocol.
Cisco IOS DECnet Command Reference	
Cisco IOS XE Dial Technologies Configuration Guide	Asynchronous communications, dial backup, dialer technology,
Cisco IOS Dial Technologies Command Reference	Multilink PPP (MLP), PPP, and virtual private dialup network (VPDN).
 Easy Virtual Network Configuration Guide Easy Virtual Network Command Reference 	Easy Virtual Network (EVN) is an IP-based virtualization technology that provides end-to-end virtualization of the network. With EVN, you can use a single IP infrastructure to provide separate virtual networks whose traffic paths remain isolated from each other.
 Cisco IOS XE High Availability Configuration Guide Cisco IOS High Availability Command Reference 	A variety of high availability (HA) features and technologies that are available for different network segments (from enterprise access to service provider core) to facilitate creation of end-to-end highly available networks. Cisco IOS HA features and technologies can be categorized in three key areas: system-level resiliency, network-level resiliency, and embedded management for resiliency.
 Cisco IOS XE Intelligent Services Gateway Configuration Guide Cisco IOS Intelligent Services Gateway Command Reference 	Subscriber identification, service and policy determination, session creation, session policy enforcement, session life-cycle management, accounting for access and service usage, and session state monitoring.

Cor	nfiguration Guide and Command Reference Titles	Features/Protocols/Technologies
•	Cisco IOS XE Interface and Hardware Component Configuration Guide	LAN interfaces, logical interfaces, serial interfaces, virtual interfaces, and interface configuration.
•	Cisco IOS Interface and Hardware Component Command Reference	
•	Cisco IOS XE IP Addressing Services Configuration Guide	IP addressing, Address Resolution Protocol (ARP), Network Address Translation (NAT), Domain Name System (DNS),
•	Cisco IOS IP Addressing Services Command Reference	Address Resolution Protocol (NHRP).
•	Cisco IOS XE IP Application Services Configuration Guide	Enhanced Object Tracking (EOT), Gateway Load Balancing Protocol (GLBP), Hot Standby Router Protocol (HSRP), IP
•	Cisco IOS IP Application Services Command Reference	User Datagram Protocol (UDP), and Virtual Router Redundancy Protocol (VRRP).
•	Cisco IOS XE IP Multicast Configuration Guide Cisco IOS IP Multicast Command Reference	Protocol Independent Multicast (PIM) sparse mode (PIM-SM), bidirectional PIM (bidir-PIM), Source Specific Multicast (SSM), Multicast Source Discovery Protocol (MSDP), Internet Group Management Protocol (IGMP), and Multicast VPN (MVPN).
٠	Cisco IOS XE IP Routing: BFD Configuration Guide	Bidirectional forwarding detection (BFD).
٠	Cisco IOS XE IP Routing: BGP Configuration Guide	Border Gateway Protocol (BGP), multiprotocol BGP,
•	Cisco IOS IP Routing: BGP Command Reference	multiprotocol BGP extensions for IP multicast.
•	Cisco IOS XE IP Routing: EIGRP Configuration Guide	Enhanced Interior Gateway Routing Protocol (EIGRP).
•	Cisco IOS IP Routing: EIGRP Command Reference	
٠	Cisco IOS XE IP Routing: ISIS Configuration Guide	Intermediate System-to-Intermediate System (IS-IS).
•	Cisco IOS IP Routing: ISIS Command Reference	
٠	Cisco IOS XE IP Routing: ODR Configuration Guide	On-Demand Routing (ODR).
•	Cisco IOS IP Routing: ODR Command Reference	
٠	Cisco IOS XE IP Routing: OSPF Configuration Guide	Open Shortest Path First (OSPF).
•	Cisco IOS IP Routing: OSPF Command Reference	
•	Cisco IOS XE IP Routing: Protocol-Independent Configuration Guide	IP routing protocol-independent features and commands. Generic policy-based routing (PBR) features and commands are
•	Cisco IOS IP Routing: Protocol-Independent Command Reference	included.
٠	Cisco IOS XE IP Routing: RIP Configuration Guide	Routing Information Protocol (RIP).
•	Cisco IOS IP Routing: RIP Command Reference	
•	Cisco IOS XE IP SLAs Configuration Guide	Cisco IOS IP Service Level Agreements (IP SLAs).
•	Cisco IOS IP SLAs Command Reference	
٠	Cisco IOS XE IP Switching Configuration Guide	Cisco Express Forwarding.
•	Cisco IOS IP Switching Command Reference	

Cor	nfiguration Guide and Command Reference Titles	Features/Protocols/Technologies
•	Cisco IOS XE IPv6 Configuration Guide Cisco IOS IPv6 Command Reference	For a list of IPv6 features, protocols, and technologies, go to the IPv6 "Start Here" document at the following URL:
		http://www.cisco.com/en/US/docs/ios/ios_xe/ipv6/configuratio n/guide/ip6-roadmap_xe.html
•	Cisco IOS XE ISO CLNS Configuration Guide	ISO Connectionless Network Service (CLNS).
•	Cisco IOS ISO CLNS Command Reference	
٠	Cisco IOS XE LAN Switching Configuration Guide	VLANs and multilayer switching (MLS).
•	Cisco IOS LAN Switching Command Reference	
•	Cisco IOS XE Multiprotocol Label Switching Configuration Guide	MPLS Label Distribution Protocol (LDP), MPLS Layer 2 VPNs, MPLS Layer 3 VPNs, MPLS Traffic Engineering (TE), and
•	Cisco IOS Multiprotocol Label Switching Command Reference	MPLS Embedded Management (EM) and MIBs.
٠	Cisco IOS XE NetFlow Configuration Guide	Network traffic data analysis, aggregation caches, and export
•	Cisco IOS NetFlow Command Reference	features.
٠	Cisco IOS XE Network Management	Basic system management, system monitoring and logging,
	Configuration Guide	Cisco IOS Scripting with Tool Control Language (Tcl), Cisco networking services (CNS) Embedded Event Manager
•	Cisco IOS Network Management Command Reference	(EEM), Embedded Syslog Manager (ESM), HTTP, Remote Monitoring (RMON), and SNMP.
٠	Cisco IOS XE Novell IPX Configuration Guide	Novell Internetwork Packet Exchange (IPX) protocol.
•	Cisco IOS Novell IPX Command Reference	
•	Cisco IOS XE Optimized Edge Routing Configuration Guide	Optimized edge routing (OER) monitoring and automatic route optimization and load distribution for multiple connections
•	Cisco IOS Optimized Edge Routing Command Reference	between networks.
•	Cisco IOS XE Performance Routing Configuration Guide	Performance Routing (PfR) provides additional intelligence to classic routing technologies to track the performance of, or
•	Cisco IOS Performance Routing Command Reference	verify the quality of, a path between two devices over a WAN infrastructure in order to determine the best egress or ingress path for application traffic.
•	Cisco IOS XE Quality of Service Solutions Configuration Guide	Class-based weighted fair queueing (CBWFQ), low latency queueing (LLQ), Modular Quality of Service (QoS)
•	Cisco IOS Quality of Service Solutions Command Reference	Command-Line Interface (CLI) (MQC), Network-Based Application Recognition (NBAR), priority queueing, Multilink PPP (MLP) for QoS, header compression, Resource Reservation Protocol (RSVP), weighted fair queueing (WFQ), and weighted random early detection (WRED).
•	Cisco IOS Security Command Reference	Access control lists (ACLs); authentication, authorization, and accounting (AAA); firewalls; IP security and encryption; neighbor router authentication; network access security; public key infrastructure (PKI); RADIUS; and TACACS+.

Configuration Guide and Command Reference Titles		Features/Protocols/Technologies	
•	Cisco IOS XE Security Configuration Guide: Secure Connectivity	Internet Key Exchange (IKE) for IPsec VPNs; security for VPNs with IPsec; VPN availability features (reverse route injection, IPsec preferred peer, and real-time resolution for the IPsec tunnel peer); IPsec data plane features; IPsec management plane features; Public Key Infrastructure (PKI); Dynamic Multipoint VPN (DMVPN); Easy VPN; and Cisco Group Encrypted Transport VPN (GET VPN).	
•	Cisco IOS XE Security Configuration Guide: Securing the Control Plane	Control Plane Policing, Neighborhood Router Authentication.	
•	<i>Cisco IOS XE Security Configuration Guide: Securing the Data Plane</i>	Access Control Lists (ACLs); Firewalls: Context-Based Access Control (CBAC) and Zone-Based Firewall; Cisco IOS Intrusion Prevention System (IPS); Flexible Packet Matching; Unicast Reverse Path Forwarding (uRPF); Threat Information Distribution Protocol (TIDP) and TMS.	
•	Cisco IOS XE Security Configuration Guide: Securing User Services	AAA (includes Network Admission Control [NAC]); Security Server Protocols (RADIUS and TACACS+); Secure Shell (SSH); Secure Access for Networking Devices (includes Autosecure and Role-Based CLI access); Lawful Intercept.	
•	Cisco IOS XE Service Advertisement Framework Configuration Guide	Cisco Service Advertisement Framework.	
•	Cisco IOS Service Advertisement Framework Command Reference		
•	Cisco IOS XE VPDN Configuration Guide Cisco IOS VPDN Command Reference	Multihop by Dialed Number Identification Service (DNIS), timer and retry enhancements for L2TP and Layer 2 Forwarding (L2F), RADIUS Attribute 82 (tunnel assignment ID), shell-based authentication of VPDN users, and tunnel authentication via RADIUS on tunnel terminator.	
•	Cisco IOS XE Wide-Area Networking Configuration Guide	Frame Relay; L2VPN Pseudowire Redundancy; and Media-Independent PPP and Multilink PPP.	
•	Cisco IOS Wide-Area Networking Command Reference		

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies	
 Cisco Unified Border Element (Enterprise) Configuration Guide Cisco IOS Voice Command Reference 	The Cisco Unified Border Element (Enterprise) on the Cisco ASR 1000 brings a scalable option for enterprise customers. Running as a process on the Cisco ASR 1000 and utilizing the high-speed RTP packet processing path, the Cisco Unified Border Element (Enterprise) is used as an IP-to-IP gateway by enterprises and commercial customers to interconnect SIP and H.323 voice and video networks. The Cisco UBE (Enterprise) provides a network-to-network demarcation interface for signaling interworking, media interworking, address and port translations, billing, security, quality of service (QoS), and bandwidth management.	
 Cisco Unified Border Element (SP Edition) Configuration Guide: Distributed Model Cisco Unified Border Element (SP Edition) Command Reference: Distributed Model 	The Cisco Unified Border Element (SP Edition) is a session border controller (SBC) that is VoIP-enabled and deployed at the edge of networks. For Cisco IOS XE Release 2.3 and earlier releases, Cisco Unified Border Element (SP Edition) is supported only in the distributed mode. Operating in the distributed mode, the SBC is a toolkit of functions that can be used to deploy and manage VoIP services, such as signaling interworking, network hiding, security, and quality of service.	
 Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model Cisco Unified Border Element (SP Edition) Command Reference: Unified Model 	 The Cisco Unified Border Element (SP Edition) is a highly scalable, carrier-grade session border controller (SBC) that is designed for service providers and that is generally deployed at the border of the enterprise or SP networks to enable the easy deployment and management of VoIP services. Cisco Unified Border Element (SP Edition) is integrated into Cisco routing platforms and can use a large number of router functions to provide a very feature-rich and intelligent SBC application. Formerly known as Integrated Session Border Controller, Cisco Unified Border Element (SP Edition) provides a network-to-network demarcation interface for signaling interworking, media interworking, address and port translations, billing, security, quality of service, call admission control, and bandwidth management. For Cisco IOS XE Release 2.4 and later releases, Cisco Unified Border Element (SP Edition) can operate in two modes or deployment models; unified and distributed. The configuration 	

Table 2 lists documents and resources that supplement the Cisco IOS XE software configuration guides and command references.

Document Title or Resource	Description
Cisco IOS Master Command List, All Releases	Alphabetical list of all the commands documented in all Cisco IOS XE software releases.
Cisco IOS Debug Command Reference	Alphabetical list of debug commands including brief descriptions of use, command syntax, and usage guidelines.
Cisco IOS XE system messages	List of Cisco IOS XE system messages and descriptions. System messages may indicate problems with your system, may be informational only, or may help diagnose problems with communications lines, internal hardware, or the system software.
Release notes and caveats	Information about new and changed features, system requirements, and other useful information about specific software releases; information about defects in specific Cisco IOS XE software releases.
MIBs	Files used for network monitoring. To locate and download MIBs for selected platforms, Cisco IOS XE software releases, and feature sets, use Cisco MIB Locator at the following URL: http://www.cisco.com/go/mibs
RFCs	Standards documents maintained by the Internet Engineering Task Force (IETF) that Cisco IOS XE documentation references where applicable. The full text of referenced RFCs may be obtained at the following URL:
	http://www.rfc-editor.org/

 Table 2
 Cisco IOS XE Software Supplementary Documents and Resources

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS version 2.0.



Using the Command-Line Interface in Cisco IOS XE Software

This chapter provides basic information about the command-line interface (CLI) in Cisco IOS XE software and how you can use some of the CLI features. This document contains the following sections:

- Initially Configuring a Device, page 1-17
- Using the CLI, page 1-18
- Saving Changes to a Configuration, page 1-27
- Additional Information, page 1-28

For more information about using the CLI, see "Part 1: Using the Cisco IOS Command-Line Interface (CLI)" of *Cisco IOS XE Configuration Fundamentals Configuration Guide*.

For information about the software documentation set, see the "About Cisco IOS XE Software Documentation" document.

Initially Configuring a Device

Initially configuring a device varies by platform. For information about performing an initial configuration, see the hardware installation documentation that is provided with the original packaging of the product or go to the Product Support area of Cisco.com at http://www.cisco.com/go/techdocs.

After you have performed the initial configuration and connected the device to your network, you can configure the device by using the console port or a remote access method, such as Telnet or Secure Shell (SSH), to access the CLI or by using the configuration method provided on the device, such as Security Device Manager.

Changing the Default Settings for a Console or AUX Port

There are only two settings that you can change on a console port or an AUX port:

- Change the port speed with the **config-register 0x** command. Changing the port speed is not recommended. The well-known default speed is 9600.
- Change the behavior of the port; for example, by adding a password or changing the timeout value.



The AUX port on the Route Processor (RP) installed in a Cisco ASR 1000 series router does not serve any useful customer purpose and should be accessed only under the advisement of a customer support representative.

Using the CLI

This section describes the following topics:

- Understanding Command Modes, page 1-18
- Using the Interactive Help Feature, page 1-21
- Understanding Command Syntax, page 1-22
- Understanding Enable and Enable Secret Passwords, page 1-23
- Using the Command History Feature, page 1-24
- Abbreviating Commands, page 1-25
- Using Aliases for CLI Commands, page 1-25
- Using the no and default Forms of Commands, page 1-26
- Using the debug Command, page 1-26
- Filtering Output Using Output Modifiers, page 1-26
- Understanding CLI Error Messages, page 1-27

Understanding Command Modes

The CLI command mode structure is hierarchical, and each mode supports a set of specific commands. This section describes the most common of the many modes that exist.

Table 1-1 lists common command modes with associated CLI prompts, access and exit methods, and a brief description of how each mode is used.

Command Mode	Access Method	Prompt	Exit Method	Mode Usage
User EXEC	Log in.	Router>	Issue the logout or exit command.	Change terminal settings.Perform basic tests.Display device status.
Privileged EXEC	From user EXEC mode, issue the enable command.	Router#	Issue the disable command or the exit command to return to user EXEC mode.	 Issue show and debug commands. Copy images to the device. Reload the device. Manage device configuration files. Manage device file systems.
Global configuration	From privileged EXEC mode, issue the configure terminal command.	Router(config)#	Issue the exit command or the end command to return to privileged EXEC mode.	Configure the device.

Table 1-1 CLI Command Modes

Command Mode	Access Method	Prompt	Exit Method	Mode Usage
Interface configuration	From global configuration mode, issue the interface command.	Router(config-if)#	Issue the exit command to return to global configuration mode or the end command to return to privileged EXEC mode.	Configure individual interfaces.
Line configuration	From global configuration mode, issue the line vty or line console command.	Router(config-line)#	Issue the exit command to return to global configuration mode or the end command to return to privileged EXEC mode.	Configure individual terminal lines.

Table 1-1 CLI Command Modes (continued)

Command Mode	Access Method	Prompt	Exit Method	Mode Usane
ROM monitor	From privileged EXEC mode, issue the reload command. Press the Break key during the first 60 seconds while the system is booting.	rommon # > The # symbol represents the line number and increments at each prompt.	Issue the continue command.	 Run as the default operating mode when a valid image cannot be loaded. Access the fall-back procedure for loading an image when the device lacks a valid image and cannot be booted. Perform password recovery when a CTRL-Break sequence is issued within 60 seconds of a power-on or reload event.
Diagnostic	 The router boots or enters diagnostic mode in the following scenarios. When a Cisco IOS XE process or processes fail, in most scenarios the router will reload. A user-configured access policy was configured using the transport-map command, which directed the user into diagnostic mode. The router was accessed using an RP auxiliary port. A break signal (Ctrl-C, Ctrl-Shift-6, or the send break command) was entered, and the router was configured to enter diagnostic mode when the break signal was received. 	Router (diag) #	If a Cisco IOS XE process failure is the reason for entering diagnostic mode, the failure must be resolved and the router must be rebooted to exit diagnostic mode. If the router is in diagnostic mode because of a transport-map configuration, access the router through another port or use a method that is configured to connect to the Cisco IOS XE CLI. If the RP auxiliary port was used to access the router, use another port for access. Accessing the router through the auxiliary port is not useful for customer purposes.	 Inspect various states on the router, including the Cisco IOS XE state. Replace or roll back the configuration. Provide methods of restarting the Cisco IOS XE software or other processes. Reboot hardware, such as the entire router, an RP, an ESP, a SIP, a SPA, or other hardware components. Transfer files into or off of the router using remote access methods such as FTP, TFTP, and SCP.

Table 1-1CLI Command Modes (continued)

EXEC commands are not saved when the software reboots. Commands that you issue in a configuration mode can be saved to the startup configuration. If you save the running configuration to the startup configuration, these commands will execute when the software is rebooted. Global configuration mode is the highest level of configuration mode. From global configuration mode, you can enter a variety of other configuration modes, including protocol-specific modes.

ROM monitor mode is a separate mode that is used when the software cannot load properly. If a valid software image is not found when the software boots or if the configuration file is corrupted at startup, the software might enter ROM monitor mode. Use the question symbol (?) to view the commands that you can use while the device is in ROM monitor mode.

```
rommon 1 > ?
alias set and display aliases command
boot boot up an external process
confreg configuration register utility
cont continue executing a downloaded image
context display the context of a loaded image
cookie display contents of cookie PROM in hex
.
.
.
rommon 2 >
```

The following example shows how the command prompt changes to indicate a different command mode:

```
Router> enable
Router# configure terminal
Router(config)# interface ethernet 1/1
Router(config-if)# ethernet
Router(config-line)# exit
Router(config)# end
Router#
```

```
Note
```

A keyboard alternative to the end command is Ctrl-Z.

Using the Interactive Help Feature

The CLI includes an interactive Help feature. Table 1-2 describes how to use the Help feature.

Command	Purpose
help	Provides a brief description of the Help feature in any command mode.
?	Lists all commands available for a particular command mode.
partial command?	Provides a list of commands that begin with the character string (no space between the command and the question mark).
partial command< Tab>	Completes a partial command name (no space between the command and <tab>).</tab>
command ?	Lists the keywords, arguments, or both associated with the command (space between the command and the question mark).
command keyword ?	Lists the arguments that are associated with the keyword (space between the keyword and the question mark).

 Table 1-2
 CLI Interactive Help Commands

Γ

The following examples show how to use the help commands:

help

Router> help

Help may be requested at any point in a command by entering a question mark '?'. If nothing matches, the help list will be empty and you must backup until entering a '?' shows the available options.

Two styles of help are provided:

1. Full help is available when you are ready to enter a command argument (e.g. 'show ?') and describes each possible argument.

2. Partial help is provided when an abbreviated argument is entered and you want to know what arguments match the input (e.g. 'show pr?'.)

?

```
Router# ?
Exec commands:
    access-enable
    access-profile
    access-template
    alps
    archive
<snip>
```

Create a temporary access-List entry Apply user-profile to interface Create a temporary access-List entry ALPS exec commands manage archive files

partial command?

Router(config)# **zo?** zone zone-pair

partial command<Tab>

Router(config)# we<Tab> webvpn

command?

```
Router(config-if)# pppoe ?
enable Enable pppoe
max-sessions Maximum PPPOE sessions
```

command keyword?

```
Router(config-if)# pppoe enable ?
  group attach a BBA group
  <cr>
```

Understanding Command Syntax

Command syntax is the format in which a command should be entered in the CLI. Commands include the name of the command, keywords, and arguments. Keywords are alphanumeric strings that are used literally. Arguments are placeholders for values that a user must supply. Keywords and arguments may be required or optional.

Specific conventions convey information about syntax and command elements. Table 1-3 describes these conventions.

Symbol/Text	Function	Notes
< > (angle brackets)	Indicate that the option is an argument.	Sometimes arguments are displayed without angle brackets.
A.B.C.D.	Indicates that you must enter a dotted decimal IP address.	Angle brackets (< >) are not always used to indicate that an IP address is an argument.
WORD (all capital letters)	Indicates that you must enter one word.	Angle brackets (< >) are not always used to indicate that a WORD is an argument.
LINE (all capital letters)	Indicates that you must enter more than one word.	Angle brackets (< >) are not always used to indicate that a LINE is an argument.
<cr> (carriage return)</cr>	Indicates the end of the list of available keywords and arguments, and also indicates when keywords and arguments are optional. When <cr> is the only option, you have reached the end of the branch or the end of the command if the command has only one branch.</cr>	

The following examples show syntax conventions:

```
Router(config)# ethernet cfm domain ?
WORD domain name
Router(config)# ethernet cfm domain dname ?
level
Router(config)# ethernet cfm domain dname level ?
<0-7> maintenance level number
Router(config)# ethernet cfm domain dname level 7 ?
<cr>
Router(config)# snmp-server file-transfer access-group 10 ?
protocol protocol options
<cr>
Router(config)# logging host ?
Hostname or A.B.C.D IP address of the syslog server
ipv6 Configure IPv6 syslog server
```

Understanding Enable and Enable Secret Passwords

Some privileged EXEC commands are used for actions that impact the system, and it is recommended that you set a password for these commands to prevent unauthorized use. Two types of passwords, enable (not encrypted) and enable secret (encrypted), can be set. The following commands set these passwords and are issued in global configuration mode:

- enable password
- enable secret *password*

Using an enable secret password is recommended because it is encrypted and more secure than the enable password. When you use an enable secret password, text is encrypted (unreadable) before it is written to the config.text file. When you use an enable password, the text is written as entered (readable) to the config.text file.

Each type of password is case sensitive, can contain from 1 to 25 uppercase and lowercase alphanumeric characters, and can start with a number. Spaces are also valid password characters; for example, "two words" is a valid password. Leading spaces are ignored, but trailing spaces are recognized.

Note

Both password commands have numeric keywords that are single integer values. If you choose a number for the first character of your password followed by a space, the system will read the number as if it were the numeric keyword and not as part of your password.

When both passwords are set, the enable secret password takes precedence over the enable password.

To remove a password, use the **no** form of the commands: **no enable** *password* or **no enable** *secret password*.

For more information about password recovery procedures for Cisco products, see the following:

http://www.cisco.com/en/US/products/sw/iosswrel/ps1831/ products_tech_note09186a00801746e6.shtml

Using the Command History Feature

The command history feature saves the commands that you enter during a session in a command history buffer. The default number of commands saved is 10, but the number is configurable within the range of 0 to 256. This command history feature is particularly useful for recalling long or complex commands.

To change the number of commands saved in the history buffer for a terminal session, issue the **terminal history size** command:

```
Router# terminal history size num
```

A command history buffer is also available in line configuration mode with the same default and configuration options. To set the command history buffer size for a terminal session in line configuration mode, issue the **history** command:

```
Router(config-line) # history [size num]
```

To recall commands from the history buffer, use the following methods:

- Press Ctrl-P or the Up Arrow key—Recalls commands beginning with the most recent command. Repeat the key sequence to recall successively older commands.
- Press Ctrl-N or the Down Arrow key—Recalls the most recent commands in the history buffer after they have been recalled using Ctrl-P or the Up Arrow key. Repeat the key sequence to recall successively more recent commands.



The arrow keys function only on ANSI-compatible terminals such as the VT100.

• Issue the **show history** command in user EXEC or privileged EXEC mode—Lists the most recent commands that you entered. The number of commands that are displayed is determined by the setting of the **terminal history size** and **history** commands.

The command history feature is enabled by default. To disable this feature for a terminal session, issue the **terminal no history** command in user EXEC or privileged EXEC mode or the **no history** command in line configuration mode.

Abbreviating Commands

Typing a complete command name is not always required for the command to execute. The CLI recognizes an abbreviated command when the abbreviation contains enough characters to uniquely identify the command. For example, the **show version** command can be abbreviated as **sh ver**. It cannot be abbreviated as **s ver** because **s** could mean **show**, **set**, or **systat**. The **sh v** abbreviation also is not valid because the **show** command has **vrrp** as a keyword in addition to **version**.

Using Aliases for CLI Commands

To save time and the repetition of entering the same command multiple times, you can use a command alias. An alias can be configured to do anything that can be done at the command line, but an alias cannot move between modes, type in passwords, or perform any interactive functions.

Table 1-4 shows the default command aliases.

Command Alias	Original Command			
h	help			
lo	logout			
р	ping			
s	show			
u or un	undebug			
W	where			

Table 1-4 Default Command Aliases

To create a command alias, issue the **alias** command in global configuration mode. The syntax of the command is **alias** *mode command-alias original-command*. Following are some examples:

- Router(config)# alias exec prt partition—privileged EXEC mode
- Router(config)# alias configure sb source-bridge—global configuration mode
- Router(config)# alias interface rl rate-limit—interface configuration mode

To view both default and user-created aliases, issue the show alias command.

For more information about the **alias** command, see http://www.cisco.com/en/US/docs/ios/fundamentals/command/reference/cf_book.html.

Using the no and default Forms of Commands

Most configuration commands have a **no** form that is used to reset a command to its default value or to disable a feature or function. For example, the **ip routing** command is enabled by default. To disable this command, you would issue the **no ip routing** command. To re-enable IP routing, you would issue the **ip routing** command.

Configuration commands may also have a **default** form, which returns the command settings to their default values. For commands that are disabled by default, using the **default** form has the same effect as using the **no** form of the command. For commands that are enabled by default and have default settings, the **default** form enables the command and returns the settings to their default values. To see what **default** commands are available on your system, enter **default** ? in the appropriate command mode of the command-line interface.

The **no** form is documented in the command pages of Cisco IOS command references. The **default** form is generally documented in the command pages only when the **default** form performs a function different than that of the plain and **no** forms of the command.

Command pages often include a "Command Default" section as well. The "Command Default" section documents the state of the configuration if the command is not used (for configuration commands) or the outcome of using the command if none of the optional keywords or arguments is specified (for EXEC commands).

Using the debug Command

A **debug** command produces extensive output that helps you troubleshoot problems in your network. These commands are available for many features and functions within Cisco IOS XE software. Some **debug** commands are **debug all**, **debug aaa accounting**, and **debug mpls packets**. To use **debug** commands during a Telnet session with a device, you must first enter the **terminal monitor** command. To turn off debugging completely, you must enter the **undebug all** command.

For more information about **debug** commands, see the *Cisco IOS Debug Command Reference* at http://www.cisco.com/en/US/docs/ios/debug/command/reference/db_book.html.

Caution

Debugging is a high priority and high CPU utilization process that can render your device unusable. Use **debug** commands only to troubleshoot specific problems. The best times to run debugging are during periods of low network traffic and when few users are interacting with the network. Debugging during these periods decreases the likelihood that the **debug** command processing overhead will affect network performance or user access or response times.

Filtering Output Using Output Modifiers

Many commands produce lengthy output that may use several screens to display. You can use output modifiers to filter this output to show only the information that you want to see.

The following three output modifiers are available:

- **begin** *regular-expression*—Displays the first line in which a match of the regular expression is found and all lines that follow.
- include *regular-expression*—Displays all lines in which a match of the regular expression is found.

• **exclude** *regular-expression*—Displays all lines except those in which a match of the regular expression is found.

To use one of these output modifiers, type the command followed by the pipe symbol (I), the modifier, and the regular expression that you want to search for or filter. A regular expression is a case-sensitive alphanumeric pattern. It can be a single character or number, a phrase, or a more complex string.

The following example illustrates how to filter output of the **show interface** command to display only lines that include the expression "protocol."

```
Router# show interface | include protocol
```

```
FastEthernet0/0 is up, line protocol is up
Serial4/0 is up, line protocol is up
Serial4/1 is up, line protocol is up
Serial4/2 is administratively down, line protocol is down
Serial4/3 is administratively down, line protocol is down
```

Understanding CLI Error Messages

You may encounter some error messages while using the CLI. Table 1-5 shows the common CLI error messages.

Error Message	Meaning	How to Get Help		
% Ambiguous command: "show con"	You did not enter enough characters for the command to be recognized.	Reenter the command followed by a space and a question mark (?). The keywords that you are allowed to enter for the command appear.		
% Incomplete command.	You did not enter all the keywords or values required by the command.	Reenter the command followed by a space and a question mark (?). The keywords that you are allowed to enter for the command appear.		
% Invalid input detected at "^" marker.	You entered the command incorrectly. The caret (^) marks the point of the error.	Enter a question mark (?) to display all the commands that are available in this command mode. The keywords that you are allowed to enter for the command appear.		

Table 1-5 Common CLI Error Messages

For more system error messages, see Cisco IOS XE System Messages.

Saving Changes to a Configuration

To save changes that you made to the configuration of a device, you must issue the **copy running-config startup-config** command or the **copy system:running-config nvram:startup-config** command. When you issue these commands, the configuration changes that you made are saved to the startup configuration and saved when the software reloads or power to the device is turned off or interrupted.

The following example shows the syntax of the copy running-config startup-config command:

```
Router# copy running-config startup-config
Destination filename [startup-config]?
```

You press Enter to accept the startup-config filename (the default), or type a new filename and then press Enter to accept that name. The following output is displayed indicating that the configuration was saved:

```
Building configuration...
[OK]
Router#
```

On most platforms, the configuration is saved to NVRAM. On platforms with a Class A flash file system, the configuration is saved to the location specified by the CONFIG_FILE environment variable. The CONFIG_FILE variable defaults to NVRAM.

Additional Information

• "Part 1: Using the Cisco IOS Command-Line Interface (CLI)" of the Cisco IOS XE Configuration Fundamentals Configuration Guide

http://www.cisco.com/en/US/docs/ios/ios_xe/fundamentals/configuration/guide/2_xe/cf_xe_book. html

or

"Using Cisco IOS XE Software" chapter of the Cisco ASR 1000 Series Aggregation Services Routers Software Configuration Guide

http://www.cisco.com/en/US/docs/routers/asr1000/configuration/guide/chassis/Using_CLI.html

Cisco Product Support Resources

http://www.cisco.com/go/techdocs

• Support area on Cisco.com (also search for documentation by task or product)

http://www.cisco.com/en/US/support/index.html

• Software Download Center (downloads; tools; licensing, registration, advisory, and general information) (requires Cisco.com user ID and password)

http://www.cisco.com/kobayashi/sw-center/

 Error Message Decoder, a tool to help you research and resolve error messages for Cisco IOS XE software

http://www.cisco.com/cgi-bin/Support/Errordecoder/index.cgi

 Command Lookup Tool, a tool to help you find detailed descriptions of Cisco IOS XE commands (choose Select an index: IOS > Select a release: All IOS Commands) (requires Cisco.com user ID and password)

http://tools.cisco.com/Support/CLILookup

• Output Interpreter, a troubleshooting tool that analyzes command output of supported **show** commands

https://www.cisco.com/pcgi-bin/Support/OutputInterpreter/home.pl



Cisco Unified Border Element (SP Edition) Overview

This chapter presents an overview of Cisco Unified Border Element (SP Edition) on the Cisco ASR 1000 Series Aggregation Services Routers—its signaling and media functions, unified and distributed deployment models, supported features, and supported MIBs.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Contents

- Cisco Unified Border Element (SP Edition) on the Cisco ASR 1000 Series Routers, page 2-1
- Cisco Unified Border Element (SP Edition) on the Cisco ASR 1001 Series Routers, page 2-5
- Supported MIBs, page 2-5

Cisco Unified Border Element (SP Edition) on the Cisco ASR 1000 Series Routers

Cisco Unified Border Element (SP Edition) enables direct IP-to-IP interconnect between multiple administrative domains for session-based services providing protocol interworking, security, and admission control and management. Cisco Unified Border Element (SP Edition) is a voice over IP (VoIP) device that sits on the border of a network and controls call admission to that network.

Cisco Unified Border Element (SP Edition) protects the interior of the network from excessive call load and malicious traffic. Cisco Unified Border Element (SP Edition) provides additional functions such as media bridging and billing services.

Cisco Unified Border Element (SP Edition) is integrated into the Cisco IOS Software and does not require any additional hardware to run.

The SBC service includes two functional areas:

• Signaling SBC function—Managed by the signaling border element (SBE), controls access of VoIP signaling messages to the core of the network, and manipulates the contents of these messages. It does this by acting as a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA).

• Media SBC function—Managed by the data border element (DBE), controls access of media packets to the network, provides differentiated services and quality of service (QoS) for different media streams, and prevents service theft. It does this by acting as a real-time transport protocol (RTP) proxy.

For Cisco IOS XE Release 2.4, Cisco Unified Border Element (SP Edition) can operate in two modes or deployment models:

- Unified—In the unified model, both the SBE and DBE logical entities co-exist on the same network element. In this model, the signaling entity controls the media local to the router. Simply put, the SBE handles the SIP and H.323 packets and the DBE handles the RTP and RTCP packets.
- Distributed—In the distributed model, the SBE and the DBE entities reside on two different network elements. Logically, each of the SBE entities controls multiple DBE elements, and each DBE could be controlled by multiple SBE entities. The SBE interacts with the DBE entities using a session controller interface (SCI). The SCI interface supports the H.248 protocol.

In this model, the bearer always flows through the DBE, and the SBE participates only in the signaling flow. This model is typically used in conjunction with a third-party SBE that supports the DBE H.248 profile.

Note

It is important to note that the DBE configuration is still required when running in the unified model because the DBE configuration provides the information necessary for the RTP media to flow.

Note

For Cisco IOS XE Release 2.3 and earlier, the SBC supports only DBEs in the distributed model.

Figure 2-1 illustrates the unified mode. Figure 2-2 illustrates the relationships between SBEs, DBEs, and other network elements.



Figure 2-1 Relationships Between SBEs/DBEs and Other Network Elements in the Unified



Figure 2-2 Relationships Between SBEs/DBEs and Other Network Elements in the Distributed Model

Cisco Unified Border Element (SP Edition) on the Cisco ASR 1001 Series Routers

Table 2-1 list the scaling and performance that is supported on the Cisco ASR 1001 Series Routers.

Table 2-1 Scaling and Performance Supported on Cisco ASR 1001 Series Routers

Platform	HT=180	RP CPU	QFP CPU	Degradation CPS %	Reasons for Congestion	Memory Setup	Throughput Value	Feature
ASR 1001 1 RU	CPS=60	33%	40%	NA	Memory	Default	5000K	NA
ASR 1001 1 RU	CPS=57	66%	78%	NA	Memory	Default	2500L	NA

Supported MIBs

The following MIBs are supported Cisco IOS XE Release 2.4 and later for the SBC on the Cisco ASR 1000 Series Router:

- CISCO-SESSION-BORDER-CONTROLLER-EVENT-MIB
- CISCO-SESSION-BORDER-CONTROLLER-CALL-STATS-MIB

For more information about MIB support on a Cisco ASR 1000 Series Routers, refer to the *Cisco ASR* 1000 Series Aggregation Services Routers MIB Specifications Guide at:

http://www.cisco.com/en/US/docs/routers/asr1000/mib/guide/asr1kmib.html

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at:

http://tools.cisco.com/ITDIT/MIBS/servlet/index

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

To access Cisco MIB Locator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you.



Configuring Cisco Unified Border Element (SP Edition)

This chapter describes how to configure the data border element (DBE) and signaling border element (SBE) for Cisco Unified Border Element (SP Edition).

Note that the DBE configuration is still required when running in the unified model because the DBE configuration provides the information necessary for the RTP media to flow.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Configuring Unified Model

This section contains the following information on configuring the unified model:

- Configuring SBE in the Unified Model, page 3-1
- Memory Alerting, page 3-10
- Configuring Memory Alerting, page 3-11
- Configuring DBE in the Unified Model, page 3-12
- Image Upgrade Procedure for Cisco Unified Border Element (SP Edition), page 3-14

Configuring SBE in the Unified Model

This section describes how to configure a SBE on a Cisco ASR 1000 Series Routers:

Prerequisites

• In the unified mode, you must configure the SBE before the DBE.

- You need to configure blacklisting to override default blacklisting thresholds when the SBE is configured and before you start using Cisco Unified Border Element (SP Edition). See the Dynamic Blacklisting Behavior, page 12-5 for configuration information.
- When running Cisco Unified Border Element (SP Edition) with 500 or more active calls, configure the huge buffer size to 65535 bytes with the **buffer huge size 65535** command. The increased buffer size is required because by default Cisco IOS software sets the "huge" buffer size to be 18084 bytes, which is not large enough for audit responses when there are more than 500 active calls.

Configuration Tip

We strongly recommend you use different addresses for signaling and media addresses to avoid scenarios where reservation for media port range can prevent call signaling packets from reaching the route processor (RP). In this scenario, if the SBC attempts to receive a call using a port that has been reserved by the SBC for media, packets will be dropped, rather than forwarded to the RP. This type of scenario is more likely to occur for H.323 and SIP calls using TCP transport.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. signaling-address ipv4 ipv4_IP_address
- 6. signaling-port *port_num*
- 7. remote-address ipv4 ip-address ip-mask
- 8. signaling-peer peer_name
- 9. signaling-peer-port port_num
- 10. attach
- 11. exit
- 12. adjacency sip adjacency-name
- 13. signaling-address ipv4 ipv4_IP_address
- 14. signaling-port *port_num*
- 15. remote-address ipv4 ip-address ip-mask
- **16.** signaling-peer peer_name
- 17. signaling-peer-port port_num
- 18. attach
- 19. call-policy-set policy-set-id
- 20. first-call-routing-table table-name
- 21. rtg-src-adjacency-table table-id
- 22. entry entry-id
- 23. action
- 24. dst-adjacency target-adjacency
- **25**. match-adjacency *key*
- 26. exit
- 27. entry entry-id
- 28. action
- **29.** dst-adjacency target-adjacency
- **30.** match-adjacency key
- 31. complete
- **32.** active-call-policy-set policy-set-id
- 33. activate
- 34. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example:	Use the <i>adjacency-name</i> argument to define the name of the service.
	Router(config-sbc-sbe)# adjacency sip Access	Note A functional SBC needs a minimum of two adjacencies configured.
Step 5	<pre>signaling-address ipv4 ipv4_IP_address</pre>	Specifies the local IPv4 signaling address of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 88.103.29.100	
Step 6	<pre>signaling-port port_num</pre>	Specifies the local signaling port of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060	
Step 7	remote-address ipv4 <i>ip-address ip-mask</i>	Restricts the set of remote signaling peers contacted over the adjacency to those with the given IP address prefix.
	Example: Router(config-sbc-sbe-adj-sip)# remote-address 200.200.200.0 255.255.255.0	
Step 8	<pre>signaling-peer peer_address</pre>	Specifies the remote signaling peer for the SIP adjacency to use.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer 200.200.200.118</pre>	
Step 9	<pre>signaling-peer-port port_num</pre>	Specifies the remote signaling-peer port for the SIP adjacency to use.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060	

	Command or Action	Purpose
Step 10	attach	Attaches the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)#	
Step 11	exit	Exits SBE SIP adjacency configuration mode and enters SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 12	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example : Router(config-sbc-sbe)# adjacency sip Core	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 13	<pre>signaling-address ipv4 ipv4_IP_address</pre>	Specifies the local IPv4 signaling address of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 88.103.33.100	
Step 14	signaling-port port_num	Specifies the local signaling port of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060	
Step 15	<pre>remote-address ipv4 ip-address ip-mask</pre>	Restricts the set of remote signaling peers contacted over the adjacency to those with the given IP address prefix.
	Example: Router(config-sbc-sbe-adj-sip)# remote-address 200.200.200.0 255.255.255.0	
Step 16	<pre>signaling-peer peer_address</pre>	Specifies the remote signaling peer for the SIP adjacency to use.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer 200.200.200.118</pre>	
Step 17	<pre>signaling-peer-port port_num</pre>	Specifies the remote signaling-peer port for the SIP adjacency to use.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060</pre>	
Step 18	attach	Attaches the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# attach	

	Command or Action	Purpose
Step 19	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity, creating a new policy set, if necessary.
	<pre>Example: Router(config-sbc-sbe-callpolicy)# call-policy-set 1</pre>	Note There can only be one call policy set at any given time.
Step 20	<pre>first-call-routing-table table-name Example: Router(config-sbc-sbe-callpolicy)# first-call-routing-table start-table</pre>	Configures the name of the first policy table to process when performing the routing stage of policy for new-call events.
Step 21	<pre>rtg-src-adjacency-table table-id Example: Router(config-sbc-sbe-callpolicy)# rtg-src-adjacency-table start-table</pre>	Enters the configuration mode of a routing table (creating one if necessary) within the context of an SBE policy set whose entries match the source adjacency.
Step 22	entry entry-id Example:	Enters the mode for configuring an entry in a routing table, creating the entry if necessary.
	Router(config-sbc-sbe-callpolicy-rtgtable)# entry 1	
Step 23	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-callpolicy-rtgtable-entry)# action complete</pre>	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
_		• Reject the indicated action using the reject keyword.
Step 24	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	<pre>Example: Router(config-sbc-sbe-callpolicy-rtgtable-entry) # dst-adjacency Core</pre>	
Step 25	<pre>match-adjacency target-adjacency Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)</pre>	Configures the match value of an entry in a number analysis or routing table whose entries match against the source adjacency.
Step 26	<pre># match-adjacency Access exit</pre>	Exits mode for configuring an entry in a routing table and enters configuration mode of a routing table to create an
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit</pre>	chuy.

	Command or Action	Purpose
Step 27	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry if necessary.
	<pre>Example: Router(config-sbc-sbe-callpolicy-rtgtable)# entry 2</pre>	
Step 28	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-callpolicy-rtgtable-entry)# action complete</pre>	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 29	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	<pre>Example: Router(config-sbc-sbe-callpolicy-rtgtable-entry)# dst-adjacency Access</pre>	
Step 30	match-adjacency target-adjacency	Configures the match value of an entry in a number analysis or routing table whose entries match against the source
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-adjacency Core</pre>	aujacency.
Step 31	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # complete</pre>	
Step 32	active-call-policy-set policy-set-id	Sets the active routing policy set within an SBE entity.
	Example: Router(config-sbc-sbe)# active-call-policy-set 1	
Step 33	activate	Initiates the SBC service.
	Example: Router(config-sbc-sbe)# activate	
Step 34	end	Exits SBC-DBE configuration mode and returns to EXEC mode.
	Example: Router(config-sbc-sbe)# end	

Modifying Existing Call Policy Set

A policy set is a group of policies that can be active on Cisco Unified Border Element (SP Edition) at any one time. If a policy set is active, then Cisco Unified Border Element (SP Edition) uses the rules defined within it to apply policy to events. Routing and number analysis are configured in a call policy set.

Only one policy set of each type can be active at any given time. You can switch the active policy set at any time. You cannot modify the currently active policy set without deactivating it. However you can modify policy sets that are not active. A policy set can be deleted, provided that it is not the active policy set.

To modify an existing call policy set, you must first deactivate it with the no active call-policy-set command and then execute a **no complete** command.

The following task deactivates the active call-policy-set.

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. no active-call-policy-set policy-set-id
- 5. no complete
- 6. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example:	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	no active-call-policy-set <i>policy-set-id</i>	deactivates the active routing policy set within an SBE entity.
	Example: Router(config-sbc-sbe)# no active-call-policy-set 1	

	Command or Action	Purpose
Step 5	no complete	Does not complete the active routing policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # no complete	
Step 6	exit	Exits the current mode of the configuration.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit</pre>	

Replacing an Existing Call Policy Set

Only one policy set of each type can be active at any given time. You can replace or switch the active policy set at any time. To do that, first deactivate the existing call policy set. Then activate the new call policy set for it to take effect.

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. no active-call-policy-set policy-set-id
- 5. active-call-policy-set policy-set-id
- 6. complete
- 7. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	no active-call-policy-set <i>policy-set-id</i>	Deactivates the active routing policy set within an SBE entity.
	<pre>Example: Router(config-sbc-sbe)# no active-call-policy-set 1</pre>	
Step 5	active-call-policy-set policy-set-id	Activates the new active routing policy set that is replacing the prior active routing policy set.
	<pre>Example: Router(config-sbc-sbe)# active-call-policy-set 6</pre>	
Step 6	complete	Completes the active routing policy set.
	Example: Router(config-sbc-sbe-cacpolicy- cactable-entry)# no complete	
Step 7	exit	Exits the current mode of the configuration.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# exit	

Memory Alerting

The Memory Alerting feature enables you to configure the number of active calls on an SBC based on the amount of free memory available on the device.

For example, an ASR1000 may support 5000 maximum active calls and support other features as well. In a scenario where the upper limit to the active calls is not configured, and other non-SBC features are also in use, there is a possibility that the SBC might use the system memory to a point that even the basic functions of the ASR1000 gets affected due to memory fragmentation or lack of memory.

The Memory Alerting feature enables you to configure thresholds and drop rates for various memory availability levels. This prevents the SBC from consuming memory for new calls or call registrations.

The Memory Alerting feature consists of four levels, Minor, Major, Critical, and Halt. The levels are defined based on the amount of processor memory available at a given time. Processor memory is checked after every ten new calls to determine the memory available.

You can configure the percentage of memory available to trigger each level, and define the number of calls to be rejected (0 to 10) from a set of 10 calls.

Table 3-1 represents the default percentages and drop rates.

Level	Default Percentage of Memory Remaining	Number of Calls Rejected
Minor	<= 25%	0 of 10
Major	<= 20%	4 of 10

Table 3-1 SBC Memory Alerting Levels, Default Memory Percentages, and Calls Rejected

Level	Default Percentage of Memory Remaining	Number of Calls Rejected	
Critical	<= 15%	9 of 10	
Halt	<= 10%	10 of 10	

Table 3-1 SBC Memory Alerting Levels, Default Memory Percentages, and Calls Rejected

Note

You cannot configure or modify the level Halt. If only 10 or lesser percentage of memory is available on the device, SBC stops accepting new calls.

Whenever a memory level change occurs, a message similar to the following is displayed on the console:

```
*July 2010 10:25:56.489:%SBC_COMP-3-MEMORY_ALERT: SBC memory congestion level has changed from CRITICAL to MINOR.
Usage: 883638296 of 1774290032 bytes.
```

Use the **[no] reject-threshold** [*level*] **memory** [*percentage*] [*reject rate*] command to configure the memory threshold and reject rate for new calls.

Configuring Memory Alerting

This task configures the reject threshold and reject rate for new calls.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. reject-threshold
- 5. end
- 6. show sbc sbc-name sbe call-stats reject-threshold

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enables entry into the mode of an SBC service.
		Use the <i>sbc-name</i> argument to define the name of the SBC.
	Example:	
	Router(config) # sbc mySbc	

	Command or Action	Purpose
Step 3	sbe	Enables entry into the mode of an SBE entity within an SBC service.
	Example:	
	Router(config-sbc)# sbe	
Step 4	reject-threshold	Configures the memory threshold and reject rate for new calls.
	Example:	
	Router(config-sbc-sbe)# reject-threshold major memory 20 5	
Step 5	end	Enable exit from the config-sbc-sbe mode.
	Example:	
	Router(config-sbc-sbe)# end	
Step 6	<pre>show sbc sbc-name sbe call-stats reject-threshold</pre>	Shows the reject threshold details for the selected SBC.
	Example:	
	Router# show sbc mySbc sbe call-stats reject-threshold	

Configuring DBE in the Unified Model

This section describes how to configure a DBE on a Cisco ASR 1000 Series Routers in the unified model.

The DBE configuration is still required when running in the unified model because the DBE configuration provides the information necessary for the RTP media to flow.

Prerequisites

When running Cisco Unified Border Element (SP Edition) with 500 or more active calls, configure the huge buffer size to 65535 bytes with the **buffer huge size 65535** command to ensure the buffer is large enough for audit responses.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. media-address ipv4 A.B.C.D
- 4. activate
- 5. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC-DBE configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	<pre>media-address ipv4 {A.B.C.D}</pre>	Adds the IPv4 address which can be used by the DBE as a local media address. This address is the SBC virtual
	Example: Router(config-sbc)# media-address ipv4 1.1.1.1	interface address.
Step 4	activate	Initiates the SBC service, for DBE and SBE.
	Example: Router(config-sbc)# activate	
Step 5	end	Exits SBC-DBE configuration mode and returns to Exec mode.
	Example: Router(config-sbc)# end	

Configuring Cisco Unified Border Element (SP Edition) Unified Model: Example

The following is an example of a Cisco Unified Border Element (SP Edition) unified model configuration:

```
Router# show run sbc
Generating configuration....
sbc test
  sbe
    adjacency sip Access
      signaling-address ipv4 88.103.29.100
      signaling-port 5060
      remote-address ipv4 200.200.200.0 255.255.255.0
      signaling-peer 200.200.200.118
      signaling-peer-port 5060
      attach
    adjacency sip Core
      signaling-address ipv4 88.103.33.100
      signaling-port 5060
      remote-address ipv4 200.200.200.0 255.255.255.0
      signaling-peer 200.200.200.118
      signaling-peer-port 5060
      attach
    call-policy-set 1
```

```
first-call-routing-table start-table
   rtg-src-adjacency-table start-table
      entry 1
       action complete
       dst-adjacency Core
       match-adjacency Access
      entry 2
        action complete
        dst-adjacency Access
       match-adjacency Core
    complete
  active-call-policy-set 1
media-address ipv4 88.103.29.100
media-timeout 30
deactivation-mode normal
activate
```

Configuring Memory Alerting: Example

The following example shows how to configure memory threshold and reject rate for new calls:

```
Router# configure
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# reject-threshold minor memory 30 0
Router(config-sbc-sbe)# reject-threshold major memory 20 5
Router(config-sbc-sbe)# reject-threshold critical memory 15 9
Router(config-sbc-sbe)# end
Router# show sbc mySbc sbe call-stats reject-threshold
      Memory Trigger Action
Level
_____
minor < 30 percent 0 in 10 calls dropped
major < 20 percent 5 in 10 calls dropped
critical < 15 percent 9 in 10 calls dropped
halt < 10 percent 10 in 10 calls dropped
Current level: NORMAL
Total calls rejected due to low memory threshold: 0
```

Image Upgrade Procedure for Cisco Unified Border Element (SP Edition)

The following procedures describe how to perform an image upgrade.

- Copy the Cisco Unified Border Element (SP Edition) image from the tftp location onto your hard disk: Step 1 Check if the node has two RP cards using the show platform command. Step 2 If the node has two RP cards, copy the image to the standby card using the following command: Router# copy harddisk:asr1000rp1new_image.bin stby-harddisk:asr1000rp1new_image.bin Step 3
- Do a no boot system of the existing image on the Active RP using the following command:

Router(config)# no boot system harddisk:asr1000rp1old_image.bin

Step 4 Start the upgrade using the following command:

RTP-ASR-1(config)# boot system harddisk:asr1000rp1<new_image>.bin

- **Step 5** Do a **show run** to check if the changes are reflected.
- **Step 6** Reload the node using the **reload** command:

```
Router# reload
```

```
System configuration has been modified. Save? [yes/no]: y
Building configuration...
[OK]
Proceed with reload? [confirm] y
```

- Step 7 To verify that the new image is loaded after the "reload," use the show version command.
- **Step 8** After the upgrade, check that all the cards have come up in the Active state by using the **show platform** command.





Media Address Pools

You can configure Cisco Unified Border Element (SP Edition) with a single media address or a range of media addresses. In addition you can define one or more permissible port ranges for the configured addresses. This feature allows the administrator to configure or restrict the data border element (DBE) address by address pool with or without port range, and define class of service (CoS) affinity for each port range.



For Cisco IOS XE Release 2.4, this feature is supported in both the unified and distributed models.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Media Address Pools

Release	Modification
Cisco IOS XE Release 2.1	This feature was introduced on the Cisco IOS XR.
Cisco IOS XE Release 2.4	Added support for SBC unified model.
Cisco IOS XE Release 3.2S	Added support for media address pool selection using port range tags.

Contents

This chapter contains the following sections:

- Prerequisites—Implementing Media Address Pools, page 4-2
- Restrictions for Configuring Media Address Pools, page 4-2
- Media Address Pools, page 4-3
- Configuring Media Address Pools, page 4-3

- Configuring the Port Range Tag for the CAC Policy, page 4-6
- Configuring Media Address Pools: Example, page 4-10
- Configuring a Port Range Tag for the CAC Policy: Example, page 4-11

Prerequisites—Implementing Media Address Pools

The following prerequisites are required to implement media address pools:

• Before implementing media address pools, you must create a static route.



• Creating a static route will fail if the remote peer is on the same VLAN as the interface VLAN of the media address.

• Before implementing media address pools, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions for Configuring Media Address Pools

The restrictions for configuring media address pools are:

- The ending address must be numerically higher than the starting address.
- The minimum port must be numerically lower than the maximum port.
- Port ranges may not overlap.
- Address ranges may not overlap.
- Address ranges and single addresses may not overlap.
- Where a range of addresses are defined in a single command, they will share any port ranges assigned. If there is a requirement to have different port ranges for different media addresses, then the addresses must be configured separately.
- Media addresses and port ranges may only be deleted before the DBE is activated. After DBE activation, the DBE must be deactivated in order to delete addresses and port ranges.
- After you configure media addresses and pools of addresses, you cannot delete them unless you delete the DBE.
- The port range tag is supported by only the signaling border element (SBE), and not the DBE.
- The media address and the signaling address should not be identical. If the media address and the signaling address are identical, and the Cisco ASR 1000 Series Router selects an ephemeral port to send out signaling packets, the port may overlap with the port range of the media address. As a result, the signaling packets do not get punted up to the RP, and get dropped by the media packet filter. This may result in events such as incomplete TCP handshakes during the second leg of a call through the SBC.
- The media address of the SBC must be unique, which means that:
 - It is not used by any features on the Cisco ASR 1000 Series Router other than sending and receiving call media.
 - It is not used by SBC call signaling.

Media Address Pools

If you do not specify a port range, all possible VoIP port numbers are valid. The full VoIP port range extends from 16384 to 32767 inclusive.

You can define a CoS affinity for each port range. The set of CoS is consistent with those used for Quality of Service (QoS) packet marking, and consists of voice and video. If you do not define an associated CoS affinity, then the affinity is for all call types.

You can modify the extent of the existing port ranges or the class of service (CoS) affinities of the existing port ranges, or delete an existing port range. Note that the configuration changes do not apply to the existing calls, but to the calls being set up after the configuration is committed.

From Cisco IOS Release 3.2S, support for selecting the media address pools using the port range tags has been added. A port range tag is a user-configured string that can be applied to a call in the Call Admission Control (CAC) policy in the SBC. A user can match the normal subset of call attributes when configuring a policy that applies a port range tag to a call, as with all the CAC policy fields. Similarly, tags can be added during the port range configurations on media addresses or media address pools.

When a call arrives at the SBC, it is passed to CAC as part of call setup. If a configured CAC policy matches the call, the policy assigns the port range tag to the call, after which the value is passed to the media component.

When selecting a local media address and port for a call, the SBC selects a port from a port range that can meet the following characteristics, which are applied in the order specified:

- 1. The media address range is in the requested VPN.
- 2. The media address range has an IP realm that matches the request for the media stream, if a media stream has been requested.
- **3.** The port-range either has the same CoS configured as requested for the media stream, or has the "Any" CoS configured.
- 4. If the media stream has a port range tag specified, the port-range must have an identical port range tag configured. However, if the media stream does not have a port range tag specified, the port-range must have the default zero-length port range tag configured on it.

Configuring Media Address Pools

This section contains the steps for configuring media address pools.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. media-address {ipv4 | ipv6} {addr} [nat-mode twice-nat | vrf vrf-name | managed-by {dbe | mgc}]
 or

media-address pool {**ipv4** | **ipv6**} {*start-addr*} {*end-addr*} [**nat-mode twice-nat** | **vrf** *vrf-name* | **managed-by** {**dbe** | **mgc**}]

- 4. port-range *min-port max-port* [any | voice | video | signaling | fax | tag *tag-string*]
- 5. end
- 6. show sbc service-name sbe addresses

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Creates the SBC service on the Cisco Unified Border Element (SP Edition) and enters the SBC
	Example: Router(config)# sbc MySBC	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	<pre>media-address {ipv4 ipv6} {addr} [nat-mode twice-nat vrf vrf-name managed-by {dbe mgc}]</pre>	Adds an IPv4 or IPv6 address to the set of addresses that can be used by the DBE as a local media address.
	Example: Router(config-sbc)# media-address ipv4 10.10.10.1	• <i>addr</i> —Local IPv4 or IPv6 address on an SBC interface that can be used for media arriving on the DBE.
		• nat-mode twice-nat —(Optional) Allows local addresses to be reserved for Twice-NAT pinholes.
		• vrf <i>vrf-name</i> —(Optional) Specifies that the IP address is associated with a specific VPN routing and forwarding (VRF) instance. If the VRF is not specified, the address is assumed to be an address on the global VPN.
		• managed-by —(Optional) Specifies whether the DBE or the MGC is allowed to select these addresses as local addresses for flows.
		• dbe —(Optional) Specifies that only the DBE is allowed to select these addresses as local addresses for flows.
		• mgc —(Optional) Specifies that only the media gateway controller (MGC) is allowed to select these addresses as local addresses for flows.

Command or Action	Purpose
<pre>media-address pool {ipv4 ipv6} {start-addr} {end-addr} [nat-mode twice-nat vrf vrf-name managed-by {dbe mgc}]</pre>	Creates a pool of sequential IPv4 and IPv6 media addresses that can be used by the SBC as local media addresses, and enters the SBC media address pool configuration mode.
Example: Router(config-sbc)# media-address pool ipv4	• <i>start-addr</i> —Starting IPv4 and IPv6 media address in a range of addresses.
10.10.10.1 10.10.20	• <i>end-addr</i> —Ending IPv4 and IPv6 media address in a range of addresses. The ending address must be numerically greater than the starting address.
	• nat-mode twice-nat —(Optional) Allows local addresses to be reserved for Twice-NAT pinholes.
	• vrf <i>vrf-name</i> —(Optional) Specifies that the IP addresses are associated with a specific VRF instance. If the VRF instance is not specified, the address is assumed to be an address on the global VPN.
	• managed-by —(Optional) Specifies whether the DBE or the MGC is allowed to select these addresses as local addresses for flows.
	• dbe —(Optional) Specifies that only the DBE is allowed to select these addresses as local addresses for flows.
	• mgc —(Optional) Specifies that only the MGC is allowed to select these addresses as local addresses for flows.
<pre>port-range min-port max-port [any voice video signaling fax tag tag-string] Example:</pre>	Creates a pool of sequential IPv4 media addresses that can be used by the SBC as local media addresses, and enters the SBC media address pool configuration mode.
Router(config-sbc-media-address-pool)# port-range 16384 30000 video	In the SBC media address pool configuration mode, the CoS for the port range is video.
end	Returns to the Privileged EXEC mode.
Example: Router(config-sbc)# end	
show sbc sbe addresses	Lists the addresses configured on the SBEs.
Example: Router# show sbc dmsbc-node9 sbe addresses	

Step

Step

Step



There is a known issue for the **media-address** command. If a secondary IP address under an interface SBC is configured as a media-address, when you use the **no** form of the **media-address** command to remove that media-address, the corresponding secondary IP address under that interface SBC will be removed as well. Furthermore, if that secondary IP address is configured under some interface SBC both on Active and Standby (in B2B redundancy), removing that media-address will also remove that secondary IP address under interface SBC are the same as that of secondary IPv4 address under interface SBC.

Configuring the Port Range Tag for the CAC Policy

This section contains the steps to configure the port range tag for applying to a call in the CAC policy in the SBC.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type {policy-set | limit {list of limit tables}}
- 7. entry entry-id
- 8. cac-scope {list of scope options}
- 9. caller port-range-tag {adj-name | none | string tag-string}
- 10. callee port-range-tag {adj-name | none | string tag-string}
- **11.** action [next-table goto-table-name | cac-complete]
- 12. exit
- 13. exit
- 14. complete
- 15. exit
- 16. cac-policy-set global policy-set-id
- 17. end
- 18. show sbc sbc-name sbe cac-policy-set id table name entry id

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.
Step 5	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one if necessary) within the contact of on SPE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	the context of an SBE poncy set.
Step 6	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)#	When the policy-set keyword is specified, use the cac-scope command to configure the scope in each entry at which limits are applied in a CAC Policy Set table.
	cable-type policy-set	Note In Policy Set tables, the event, call, or message is applied to all the entries.
Step 7	entry entry-id	Enters the CAC table entry mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy- cactable)# entry 1	

	Command or Action	Purpose
Step 8	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each entry at which limits are applied in a policy set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope <i>category</i>	- account —Events that are from the same account.
		 adjacency—Events that are from the same adjacency.
		 adj-group—Events that are from members of the same adjacency group.
		- call—Scope limits are per single call.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		- dst-number —Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server)

	Command or Action	Purpose
Step 9	<pre>caller port-range-tag {adj-name none string tag-string}</pre>	Configures the port range tag for a caller. This tag is used when selecting the media address and port.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller port-range-tag adj-name</pre>	• adj-name —Uses the source adjacency name as a port range tag.
		• none —Prompts the SBC to not use a port range tag for calls matching the CAC entry, and removes previously found strings, if any.
		• string <i>tag-string</i> —Specifies the explicit port range tag string.
Step 10	<pre>callee port-range-tag {adj-name none string tag-string}</pre>	Configures the port range tag for a callee. This tag is used when selecting the media address and port.
	Example:	• adj-name —Uses the destination adjacency name as a port range tag.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee port-range-tag string GenericCorePortRange	• none —Prompts the SBC to not use a port range tag for calls matching the CAC entry, and removes previously found strings, if any.
		• string <i>tag-string</i> —Specifies the explicit port range tag string.
Step 11	action [next-table goto-table-name cac-complete]	Configures the action to perform after this entry in an admission control table. Possible actions are:
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name argument</i> .
	# action cac-complete	• Stop processing for this scope using the cac-complete keyword.
Step 12	exit	Exits the entry mode, and enters the CAC table mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 13	exit	Exits the CAC table mode, and enters the CAC policy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 14	complete	Completes the CAC policy set after you commit the entire set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 15	exit	Exits the CAC policy mode, and enters the SBE configuration mode.
	Example: Router(config-sbc-sbe-cacpolicy)# exit	

	Command or Action	Purpose
Step 16	cac-policy-set global policy-set-id	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Router(config-sbc-sbe)# cac-policy-set global 23	• <i>policy-set-id</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 17	end	Exits the CAC table mode, and enters the privileged EXEC mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# end	
Step 18	show sbc sbc-name sbe cac-policy-set <i>id</i> table name entry <i>id</i>	Lists detailed information, such as caller and callee port range tags, pertaining to a given entry in a CAC policy table.
	Example: Router# show sbc MySBC sbe cac-policy-set 1 table StandardListByAccount entry 1	

Configuring Media Address Pools: Example

This section provides sample configurations for media address pools. The following example shows the creation of a static route for the media pool address.

At the Route Processor (RP):

Router(config)# **ip route 87.87.29.8 255.255.255.248 87.87.29.100** !

The following example creates a pool of IPv4 media addresses that can be used by the DBE as local media addresses:

```
Router(config)# sbc test dbe
Router(config-sbc-dbe)# media-address pool ipv4 87.87.29.8 87.87.29.15
```

The following sample script adds a single address (10.10.10.1), and two ranges of addresses (10.10.11.1 through 10.10.11.10 and 10.10.11.21 through 10.10.11.30) to the media address pool.

Two port ranges are configured on the single address. The first port range is for voice traffic, and runs from port 16384 to 20000 inclusively. The second one is for video traffic, and runs from port 20001 to 65535 inclusively.

The first range of addresses also has two similar port ranges configured that apply to all ten addresses within the range. The second range of addresses has a single port range defined, and no service class associated with it.

```
Router(config)# sbc test dbe
Router(config-sbc-dbe)# media-address ipv4 10.10.10.1
Router(config-sbc-dbe-media-address pool)# port-range 16384 20000 voice
Router(config-sbc-dbe-media-address pool)# exit
Router(config-sbc-dbe)# media-address ipv4 10.10.10.1
```

```
Router(config-sbc-dbe-media-address pool)# port-range 20001 65535 video
Router(config-sbc-dbe-media-address pool)# exit
```

```
Router(config-sbc-dbe)# media-address pool ipv4 10.10.11.1 10.10.11.10
Router(config-sbc-dbe-media-address pool)# port-range 16384 30000 voice
Router(config-sbc-dbe)# media-address pool ipv4 10.10.11.1 10.10.11.10
Router(config-sbc-dbe)# media-address pool)# port-range 30001 40000 video
Router(config-sbc-dbe-media-address pool)# exit
Router(config-sbc-dbe-media-address pool)# exit
Router(config-sbc-dbe)# media-address pool ipv4 10.10.11.21 10.10.11.30
```

Router(config-sbc-dbe-media-address pool)# port-range 20000 40000 any

The following example shows how to add an IPv4 address to the set of addresses that can be used by the SBE as a local media address, and how to configure a port range tag:

```
sbc MySBC
media-address ipv4 10.33.33.1
port-range 2000 4000 voice tag GoldCustomerA
port-range 4001 6000 video tag HighBwCustomer
port-range 10000 12005 tag Adjacency_IMS_Core
no port-range 10000 12005 tag
```

The following example shows how to create a pool of IPv6 media addresses that can be used by the SBE as local media addresses, and how to configure a port range tag:

```
sbc MySBC
```

```
media-address pool ipv6 CAFE:1234:1234:1234::0011 CAFE:1234:1234:1234::0012
port-range 2000 4000 voice tag LowBW@CustomerA
port-range 4001 6000 signaling
port-range 10000 12005 fax tag FaxGWAdjacency23
```

Configuring a Port Range Tag for the CAC Policy: Example

This section provides a sample configuration of a port range tag for applying to a call in a CAC policy set in the SBC:

```
sbc MySBC
  sbe
    cac-policy-set 1
      cac-table Table1
        table-type policy-set instigate
        entry 1
         cac-scope global
          caller port-range-tag adj-name
          callee port-range-tag adj-name
          action next-table Table2
     cac-table Table2
       table-type limit account
       entrv 1
         match-value GoldAccount
         caller port-range-tag string LargeBWPorts
         callee port-range-tag none
```



Implementing Multi-VRF on Cisco Unified Border Element (SP Edition)

Cisco Unified Border Element (SP Edition) provides support for multi-VRF (VPN routing and forwarding) on customer edge (CE) devices. This feature provides the capability of suppressing provider edge (PE) checks to prevent loops when the PE is performing a mutual redistribution of packets.

VRF is only supported in DBE media address and SBE AAA/H248 control address; DBE H248 control address does not support VRF.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XE Release 2.4, this feature is supported in both the unified and distributed model.

Feature History for Implementing Multi-VRF on Cisco Unified Border Element (SP Edition)

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco ASR 1000 Series Routers.
Cisco IOS XE Release 3.2S	SBC Voice traffic support over tunnel-interface (GRE, IPSec, MPLS, TE tunnel, BBA) was introduced on the Cisco ASR 1000 Series Routers.

Contents

This module contains the following sections:

- Prerequisites—Implementing Multi-VRF, page 5-2
- Information About Implementing Multi-VRF, page 5-2
- Implementing Multi-VRF, page 5-3
- Configuration Examples for Implementing Multi-VRF, page 5-7
- Supporting the SBC Voice Traffic over Tunnel Interfaces, page 5-13

Prerequisites—Implementing Multi-VRF

The following prerequisite is required to implement multi-VRF on Cisco Unified Border Element (SP Edition):

• Before implementing multi-VRF, Cisco Unified Border Element (SP Edition) must already be configured.

Information About Implementing Multi-VRF

Cisco Unified Border Element (SP Edition) support for multi-VRF on customer edge (CE) devices, such as customer premises routers, provides the capability of suppressing PE checks that are needed to prevent loops when the PE is performing a mutual redistribution of packets. Multi-VRF allows for the use of only one router to accomplish the tasks that multiple routers usually perform. It runs on a network without the requirement of MPLS and BGP installed.

When VRF is used on a router that is not a PE, the checks can be turned off to allow for correct population of the VRF routing table with routes to IP prefixes. Multi-VRF is also important because virtual private network (VPN) functionality is not completely supported on low-end systems. Multi-VRF provides logical separation of routing instances (and by the implication address space) within one router.

The following summarizes the features of multi-VRF:

- Allows a single physical router to be split into multiple virtual routers, where each router contains its own set of interfaces, routing table, and forwarding table. Cisco Unified Border Element (SP Edition) supports multiple (overlapping and independent) routing tables (addressing) per customer. Virtual routing contexts are used to separate routing domains within a single router.
- Multi-VRF can be used where multiple routers are required but only one is available.
- When using multi-VRF, the domain name server (DNS) queries are per VRF.
- One physical interface can belong to multiple virtual routers through the use of subinterfaces (Frame Relay, ATM, VLANs).
- BGP and MPLS are not used.
- No connectivity is provided between VRFs (would require using BGP for internal exporting and importing between VRFs).
- When a call is placed between two endpoints in the same VPN site, Cisco Unified Border Element (SP Edition) can route the media directly between them, to reduce network utilization.
- Multi-VRF on Cisco Unified Border Element (SP Edition) provides optimization where both endpoints are on the same VPN by turning media bypass on.
- When a VRF is removed from a SBC interface that is in use by an activated SBC, the IP addresses are not removed automatically by the SBC. The user has to manually remove the IP addresses when the SBC is deactivated.

For Cisco IOS XE Release 2.4, by default, all adjacencies on the same VPN have media bypass turned on. Media bypass can be turned off by using the **media-bypass-forbid** command (this command is implemented for CAC policies only).



The vrf name under the adjacency must match the context name.



Media termination occurs prior to route leaking, therefore media cannot be terminated on leaked routes.

Implementing Multi-VRF

Implementing multi-VRF is described in the following sections:

- Associating a SIP Adjacency with a VRF, page 5-3
- Configuring DBE with VRF—Distributed Model Only, page 5-5

Associating a SIP Adjacency with a VRF

This task associates a SIP adjacency with a VPN.



When an adjacency is assigned to a particular VRF, all the addresses relating to the adjacencies, such as signalling-address and remote-address, must also be routable within the VRF.

SUMMARY STEPS

- 1. adjacency sip adjacency-name
- 2. vrf vrf_name
- 3. signaling-address ipv4 local_signaling_IP_address
- 4. signaling-port *port_num*
- 5. remote-address ipv4 local_signaling_IP_address/prefix
- 6. local-id host name
- 7. signaling-peer peer_address
- 8. signaling-peer-port port_num
- 9. account account-name
- 10. media-bypass (optional)
- 11. media-bypass-forbid
- 12. attach

DETAILED STEPS

	Command or Action	Purpose
Step 1	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sip_vrf1	• Use the <i>adjacency-name</i> argument to define the name of the service.
Step 2	<pre>vrf_name</pre>	Ties a SIP adjacency to a specific VPN.
	Example: Router(config-sbc-sbe-adj-sip)# vrf my_vrf1	Note The vrf name under the adjacency must match the context name.
Step 3	<pre>signaling-address ipv4 ipv4_IP_address</pre>	Specifies the local IPv4 signaling address of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 88.88.88.88	
Step 4	<pre>signaling-port port_num</pre>	Specifies the local signaling port of the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060	
Step 5	<pre>remote-address ipv4 remote_IP_address/prefix</pre>	Restricts the set of remote signaling peers contacted over the adjacency to those with the given IP address prefix.
	Example: Router(config-sbc-sbe-adj-sip)# remote-address ipv4 10.10.101.4 255.255.255.255	
Step 6	local-id host address	Configures the local identity name on a SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# local-id host 88.88.101.11	
Step 7	signaling-peer peer_address	Specifies the remote signaling peer for the SIP adjacency to use.
	Example:	
	Router(config-sbc-sbe-adj-sip)# signaling-peer 10.10.101.4	
Step 8	<pre>signaling-peer-port port_num</pre>	Specifies the remote signaling-peer port for the SIP adjacency to use.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060	
Step 9	account account_name	Defines the SIP adjacency as belonging to an account on an SBE.
	Example: Router(config-sbc-sbe-adj-sip)# account sip-vrf1	

	Command or Action	Purpose
Step 10	media-bypass	(Optional) Configures the adjacency to allow media traffic to bypass the DBE.
	Example: Router(config-sbc-sbe-adj-sip)# media-bypass	This command is optional and only works on one adjacency.
Step 11	media-bypass-forbid	Configures the SIP adjacency to forbid media traffic to bypass the DBE.
	Example: Router(config-sbc-sbe-adj-sip)# media-bypass-forbid	If this is not configured, media traffic for calls originating and terminating on this adjacency flows directly between the endpoints and does not pass through the DBE, as long as both adjacencies are on the same VPN.
Step 12	attach	Attaches the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# attach	

Configuring DBE with VRF—Distributed Model Only

This task configures DBE with VRF in the distributed model.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name dbe
- 3. vdbe global
- 4. unexpected-source-alerting
- 5. local-port abcd
- 6. control-address h248 ipv4 A.B.C.D
- 7. controller h248 controller-index
- 8. remote-address ipv4 remote-address
- 9. remote-port [port-num]
- 10. transport [udp | tcp]
- 11. attach-controllers
- 12. media-address pool ipv4 A.B.C.D E.F.G.H vrf vrfname
- 13. media-timeout timeout
- 14. overload-time-threshold time
- 15. deactivation-mode
- 16. activate

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Accesses the configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name dbe	Creates the DBE service on the SBC and enters into SBC-DBE configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	vdbe [global]	Enters into vDBE configuration submode.
	Example:	Note In the initial release only one vDBE (the global vDBE) is supported. The vdbe name is not required. If specified, it must be global.
	Router(config-sbc-dbe)# vdbe	
Step 4	unexpected-source-alerting	Sets alerting for unexpected source addresses.
	<pre>Example: Router(config-sbc-dbe-vdbe-global)# unexpected-source-alerting</pre>	The no form of this command removes alerting for any unexpected source addresses that are received.
Step 5	<pre>local-port {abcd}</pre>	Configures a DBE to use a specific local port.
	Example: Router(config-sbc-dbe)# local-port 5090	
Step 6	control-address h248 ipv4 A.B.C.D	Configures a DBE to use a specific IPv4 H.248 control address.
	Example: Router(config-sbc-dbe)# control-address h248 ipv4 10.0.0.1	Note The control address cannot be in a VRF and must be routable in the global address table.
Step 7	controller h248 controller-index	Identifies the H.248 controller for the DBE and enters into Controller H.248 configuration mode.
	Example: Router(config-sbc-dbe)# controller h248 1	
Step 8	<pre>remote-address ipv4 remote-address</pre>	Configures the IPv4 remote address of the H.248 controller.
	Example: Router(config-sbc-dbe-vdbe-h248)# remote-address ipv4 1.1.1.1	
Step 9	remote-port [port-num]	Defines the port to connect to on the SBE for an H.248 controller.
	Example: Router(config-sbc-dbe-h248)# remote-port 2094	

	Command or Action	Purpose
Step 10	transport udp	Configures a DBE to use User Datagram Protocol (UDP) for H.248 control signaling.
	Example:	
	Router(config-sbc-dbe-h248)# transport udp	
Step 11	attach-controllers	Configure a DBE to attach to an H.248 controller.
	Example: Router(config-sbc-dbe)# attach-controllers	
Step 12	media-address pool ipv4 A.B.C.D E.F.G.H vrf vrfname	Create a pool of sequential IPv4 media addresses for an IPv4 address associated with a specific VRF instance.
	Example: Router(config-sbc-dbe)# media-address pool ipv4 10.10.10.1 10.10.10.20 vrf my_vrf1	Note The vrf name under the adjacency must match the context name.
Step 13	<pre>media-timeout timeout Example: Router(config-sbc-dbe)# media-timeout 10</pre>	Sets the maximum time a DBE waits after receiving the last media packet on a call and before cleaning up the call resources.
Step 14	overload-time-threshold time	Configures the threshold for media gateway (MG) overload control detection.
	<pre>Example: Router(config-sbc-dbe)# overload-time-threshold 400</pre>	
Step 15	deactivation-mode normal	Specifies that the DBE of an SBC signals a service change and terminates all calls upon deactivation of the DBE
	Example: Router(config-sbc-dbe)# deactivation-mode normal	service.
Step 16	activate	Initiates the SBC service.
	Example: Router(config-sbc-dbe)# activate	

Configuration Examples for Implementing Multi-VRF

This section provides the following configuration examples:

- Configuring SBC Unified Model with VRF: Example, page 5-8
- Configuring Multi-VRF: Example, page 5-9
- Associating a SIP Adjacency with a VRF: Example, page 5-9
- Configuring DBE with Multi-VRF (Distributed Model Only): Example, page 5-11

Configuring SBC Unified Model with VRF: Example

You can configure the Cisco ASR 1000 Series Router so that traffic is routed to the SBC adjacency address. This is achieved by creating a VRF instance on the router.

The following is an example, which uses VLAN trunks to get the traffic into the SBC. In this example, a VRF is created to route traffic from the 100.0.0/24 network to the 12.0.0.0/24 network, where the SIP signaling address and media address reside for a particular SBC connection.

The **interface sbc** command is needed, whenever a VRF is being used. You must have a secondary IP address defined if the media IP address is going to be different than the signaling address. However, in this case the secondary IP address is automatically added when the **media-address ipv4** command is used. It must not be manually entered.

```
vrf definition cust100side// Create a VRF instance
 Т
address-family ipv4
 exit-address-family
interface SBC100// Create an interface in the VRF space
vrf forwarding cust100side
 ip address 12.0.0.30 255.255.255.0 secondary// This contains the IP address for the
                                             // media, if different to the signaling
                                             // address. The line is not entered, but
                                             // appears automatically after the DBE
                                             // configuration is entered (see
                                             // `media-address' CLI later.)
 ip address 12.0.0.20 255.255.255.0
                                              // This is the SIP adjacency address
interface GigabitEthernet0/1/0
 no ip address
media-type sfp
negotiation auto
                                              // VLAN identifier 100 defined here
interface GigabitEthernet0/1/0.100
vrf forwarding cust100side
 encapsulation dot1Q 100
 ip address 100.0.0.1 255.255.255.0
                                           // This IP is where the remote side or external
                                          // router can send traffic to, in order to get
                                         // to the internal 12.0.0.0/24 network
interface GigabitEthernet0/1/0.200
                                          // Other VLANS that are being trunked.
vrf forwarding cust200side
 encapsulation dot1Q 200
 ip address 200.0.0.1 255.255.255.0
sbc ted
  sbe
    adjacency sip adj_cust100
      vrf cust100side
      . . .
      signaling-address 12.0.0.20
                                          // This is the local address where call traffic
                                         // will get routed to/from
                                         // This is an address for the remote side, where
      remote-address ipv4 100.0.0.14
                                         // traffic will be routed
      attach
  media-address ipv4 12.0.0.30 vrf cust100side// The media address is also on the
                                             // internal network. When the line
                                             // is entered, the interface SBC
                                             // will show a secondary address
                                             // containing this IP address.
  activate
```

Configuring Multi-VRF: Example

This sample configuration shows how the Service Virtual Interface (SVI) and adjacencies are added to associate a VPN to them.

1. Configure the line card interface associated with vrf my_vrf1 on the route processor (RP).

```
vrf definition my_vrf1
rd 55:1111
!
address-family ipv4
exit-address-family
!
2. Configure the line card interface associated with vrf, my_vrf1, on the route processor.
interface GigabitEthernet1/3
description ''Connected to CAT-3550-101 Fa 0/13 vlan919''
ip address 10.122.3.3 255.255.255.0
interface GigabitEthernet1/3.99
encapsulation dot1q 99
vrf forwarding my_vrf1
ip address 10.122.3.3 255.255.255.0
!
```

3. Configure the media address pools.

media-address pool ipv4 88.88.101.12 88.88.101.15 vrf my_vrf1 activate

Associating a SIP Adjacency with a VRF: Example

This example configuration creates a SIP adjacency associated with a VPN.

```
ip route 10.10.0.0 255.255.0.0 101.101.101.100 ip route 20.20.20.0 255.255.255.0
101.101.101.4
domain default-domain
sbc mysbc
  sbe
    adjacency sip 7200-1
     vrf mv vrf1
     inherit profile preset-core
     preferred-transport udp
     redirect-mode pass-through
     authentication nonce timeout 300
     signaling-address ipv4 101.101.101.3
     signaling-port 5061
      remote-address ipv4 0.0.0.0 0.0.0.0
      signaling-peer 101.101.101.5
     signaling-peer-port 5060
     account sip-core
     attach
    adjacency sip 7200-2
     vrf my_vrf1
      inherit profile preset-access
     preferred-transport udp
      redirect-mode pass-through
     authentication nonce timeout 300
     signaling-address ipv4 101.101.101.3
```

```
signaling-port 5060
  remote-address ipv4 0.0.0.0 0.0.0.0
  signaling-peer 101.101.101.4
 signaling-peer-port 5060
 account sip-core
  attach
adjacency sip 7200-3
 vrf my_vrf1
 nat force-on
 inherit profile preset-core
 preferred-transport udp
 redirect-mode pass-through
 authentication nonce timeout 300
  signaling-address ipv4 101.101.101.3
 signaling-port 5063
 remote-address ipv4 0.0.0.0 0.0.0.0
  signaling-peer 101.101.101.5
  signaling-peer-port 5063
  account sip-core
 reg-min-expiry 3000
 attach
sip inherit profile preset-standard-non-ims
retry-limit 3
call-policy-set 1
  first-call-routing-table invite-table
  first-reg-routing-table start-table
 rtg-src-adjacency-table invite-table
    entry 1
      action complete
      dst-adjacency 7200-2
     match-adjacency 7200-3
    entry 2
     action complete
      dst-adjacency 7200-3
     match-adjacency 7200-2
 rtg-src-adjacency-table start-table
    entry 1
     action complete
      dst-adjacency 7200-1
     match-adjacency 7200-2
    entry 2
      action complete
      dst-adjacency 7200-2
      match-adjacency 7200-1
  complete
active-call-policy-set 1
network-id 2
sip max-connections 2
sip timer
  tcp-idle-timeout 120000
 tls-idle-timeout 3600000
 udp-response-linger-period 32000
 udp-first-retransmit-interval 500
 udp-max-retransmit-interval 4000
  invite-timeout 180
```

blacklist
```
global
redirect-limit 2
deactivation-mode normal
activate
media-address ipv4 101.101.101.160 vrf my_vrf1 port-range 11000 20000 any
location-id 0
media-timeout 30
deactivation-mode normal
activate
```

Configuring DBE with Multi-VRF (Distributed Model Only): Example

To make use of Multi-VRF when Cisco Unified Border Element (SP Edition) is running in the distributed mode, both the configuration and the corresponding H.248 messages are required to be VRF-aware.

The following sample configuration creates media pool that is tied to a particular VRF. This media pool can only be used to assign media addresses for that particular VRF and can overlap with addresses from different VRF's or from the global address space.

```
vrf definition moon
vpn id 22AA:33334411
!
interface SBC1
 ip vrf forwarding moon
  ip address 90.0.0.1 255.0.0.0
T
sbc global dbe
vdbe global
h248-version 3
h248-napt-package napt
local-port 2979
 control-address h248 ipv4 200.50.1.9
 controller h248 1
 remote-address ipv4 200.50.1.254
 remote-port 2979
attach-controllers
location-id 1
media-address ipv4 90.0.0.1 vrf moon
port-range 10000 20000 any
activate
!
```

The H.248 configuration is specified in the H.248 package/Extended VPN Discrimination/ (EVPND). This package has two methods, GVPNID and VRF_NAME, of specifying to which VRF the media addresses belong. These parameters are mutually exclusive but they are independent on a per side basis. For example, side A may use the VRF_NAME method for specifying the VRF and side B may use the GVPNID method.

The VRF_NAME is a quoted ASCII string corresponding to the name of the VRF in the configuration. In the following example, the name would be "moon."

M {

```
TS { SI = IV },
    ST = 1 {
        O \{ MO = IN,
            EVPND/VRF_NAME = "moon"
          },
        R {
            v=0
            c=IN IP4 3.0.0.3
            m=application 5000 udp 0
        },
        L {
            v=0
            c=IN IP4 $
            m=application $ udp 0
        }
    }
}
```

The GVPNID is the identification number for the VRF in RFC2685 format. This is specified in the configuration as follows:

vrf definition moon
 vpn id 22AA:33334411
!

The H.248 format is then specified as:

```
M {
    TS { SI = IV },
    ST = 1 {
        O \{ MO = IN,
            EVPND/GVPNID = 22AA33334411
          },
        R {
            v=0
            c=IN IP4 3.0.0.3
            m=application 5000 udp 0
        },
        L {
            v=0
            c=IN IP4 $
            m=application $ udp 0
        }
    }
}
```

Supporting the SBC Voice Traffic over Tunnel Interfaces

The Cisco IOS XE Release earlier than Cisco IOS XE Release 3.2S did not support the SBC traffic over the tunnel interfaces. The Cisco IOS XE Release 3.2S provides support to the SBC traffic over the tunnel interfaces (PPPoE, GRE, MPLS-TE, IPsec SVTI or DVTI, DMVPN). The following topology diagrams (Figure 5-1 and Figure 5-2) illustrate the broadband deployment scenario and tunnel interface scenarios in which the SBC voice traffic is supported over the tunnel interfaces:









Implementing Adjacencies on Cisco Unified Border Element (SP Edition)

Accounts and adjacencies are the key objects used to control signaling. An account represents a service relationship with a remote organization on the signaling border element (SBE), with which Cisco Unified Border Element (SP Edition) will interact. Within each account, the user defines one or more signaling adjacencies, which connect Cisco Unified Border Element (SP Edition) to devices within that organization. The account is used to:

- Define customer-specific admission control
- Define routing policy configurations
- Organize billing records

An adjacency represents a signaling relationship with a remote call agent. There is one adjacency defined per external call agent. The adjacency is used to define protocol-specific parameters as well as admission control and routing policy. Each adjacency belongs within an account.

Each incoming call is matched to an adjacency, and each outgoing call is routed out over a second adjacency. Adjacencies can also be associated with a media gateway location, so that the most appropriate virtual data border element (vDBE) can be selected for a given call leg. Typically, an Cisco Unified Border Element (SP Edition) has at least one account representing the internal network.

You can assign each adjacency to an adjacency group, so you can enable and disable features per interface. For example, you can turn off high bandwidth features on all adjacencies to customers on a known low-bandwidth link.

This chapter also discusses the SIP Over Transport Layer Security (TLS) feature, an encryption feature that provides a secure, encrypted transport to carry all SIP messages from the caller to the callee's domain.



Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Implementing Adjacencies and SIP Over TLS

Release	Modification		
Cisco IOS XE Release 2.4	This feature and SIP over TLS were introduced on the Cisco IOS XR along with support for the unified model.		
Cisco IOS XE Release 2.6	The following features were added:		
	• Configurable Mutual TLS Authentication Per Interface.		
	• TLS Transport Parameter in Record Route Headers.		
Cisco IOS XE Release 3.1S	The Redundant Peer Addresses feature was added.		
Cisco IOS XE Release 3.2S	S The SIP peer availability detection feature was added.		
	The Public Key Infrastructure (PKI) High Availability (HA) support was added.		

Contents

This module contains the following sections:

- Prerequisites for Implementing Adjacencies, page 6-2
- Restrictions, page 6-3
- Information About Implementing Adjacencies, page 6-3
- How to Implement Adjacencies, page 6-7
- Configuration Examples for Implementing Adjacencies, page 6-14
- SIP UAS Failure Detection, page 6-15
- SIP Outbound Flood Protection, page 6-18
- SIP Over TLS, page 6-20
- SIP Peer Availability Detection, page 6-32
- Redundant Peer Addresses, page 6-34

Prerequisites for Implementing Adjacencies

The following prerequisite is required to implement adjacencies:

• Before implementing adjacencies, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions

H.323 adjacencies are not supported in Cisco IOS XE Release 2.4 and earlier.

Information About Implementing Adjacencies

Adjacencies are used to enable call signaling between the SBE and other voice over IP (VoIP) devices. Cisco Unified Border Element (SP Edition) supports adjacencies in Session Initiation Protocol (SIP) network deployments.

In a SIP network, the devices might be user agents, proxies, softswitches, or back-to-back user agents (B2BUAs). When you configure a SIP adjacency, the SBE functions as a B2BUA within the SIP network.

Adjacencies can represent both trunking and subscriber signaling relationships. The network topology and configuration of an adjacency determine its role.

The adjacency accepts packets from either the UDP or TCP socket specified in the signaling port configuration line. For SIP, the default is port 5060. When sending packets out the adjacency, the transport used is specified using the **preferred-transport** [**tcp** | **udp**] command. The default is to use UDP. Note that there is no dependency between the input and output adjacencies. It is valid to have one adjacency use TCP for the signaling and the other use UDP.

Further overview details about implementing adjacencies are described in the following sections:

- Properties Common to SIP Adjacencies
- About SIP Adjacencies in the Deployment
- How Adjacencies Affect Media Routing

Properties Common to SIP Adjacencies

The following properties are common to SIP adjacencies:

- Adjacencies are known by name. The name makes it easy for a Cisco Unified Border Element (SP Edition) policy to reference the adjacency.
- An adjacency has a local address and port for incoming call setup.
- An adjacency has a peer address and port. This is the point of contact for outgoing calls. In the SIP case, this is only true if the "force-signaling-peer" option is set for that adjacency.
- An adjacency forms the output of a routing policy decision. In other words, the routing phase for a call results in selection of an outgoing adjacency for that call. Normally, adjacency selection is done based on a destination telephone number prefix. However, two adjacencies can also be bridged together by using a source adjacency as a routing input.

About SIP Adjacencies in the Deployment

Figure 6-1 shows a simple SIP network where:

- SIP subscribers register with the SIP proxy, which acts as a single point of contact for all of them.
- The softswitch is a gateway between the SIP network and the public switched telephone network (PSTN).

• The softswitch routing policy assigns a particular phone prefix to each SIP proxy, allowing calls from the PSTN network to be routed through the proxy to a given subscriber. (In other deployments, subscribers may register directly with a softswitch without going through a proxy first.)

Figure 6-1 SIP Network



Figure 6-2 shows placement of a Cisco Unified Border Element (SP Edition) in two possible positions within the SIP network, with the adjacencies noted. Each adjacency enables call setup to one or more neighboring devices, as follows:

- ADJ_SIP1A allows call setup between SBC1 and the softswitch.
- ADJ_SIP1B allows call setup between SBC1 and the proxy.
- ADJ_SIP2A allows call setup between SBC2 and the proxy.
- ADJ_SIP_SUBSCRIBERS allows call setup between SBC2 and the subscribers.

In the case of SBC2, SIP registrations are being routed through the SBC. Registrations received on ADJ_SIP_SUBSCRIBERS are being routed to the proxy over ADJ_SIP2A.

The key difference between subscriber and nonsubscriber adjacencies is that:

- Nonsubscriber adjacencies have a configured single point of contact, the peer address for the adjacency.
- Subscriber adjacencies do not have a single point of contact and are instead configured to accept registrations.

SIP registrations require a routing policy to determine which is the correct outgoing adjacency for a given registration. This works in a very similar way to a call routing policy. See the procedures described in the Implementing Cisco Unified Border Element (SP Edition) Policies module.



Figure 6-2 Adjacencies in a SIP Network Deployment

How Adjacencies Affect Media Routing

For a distributed Cisco Unified Border Element (SP Edition) deployment, each adjacency is configured with a *media location*. The media location is an ID used to select the data border elements (DBEs) suitable for relaying media traffic for calls set up over the adjacency.

If a call is routed out over the same or different adjacency, the media may bypass a DBE. The media bypass feature allows the media packets to bypass the Cisco Unified Border Element (SP Edition) to enable the endpoints to communicate directly to each other. Media packets flow directly without going through the DBE component of the SBC after the call signaling is done. Signaling packets still flow through the SBC as usual.

The configuration is set per adjacency, and allows media bypass across different adjacencies. Media-bypass configuration is enabled under adjacency configuration. Media bypass is useful when two endpoints are on the same subnet, but the DBE is located elsewhere on the network.

Figure 6-3 and Figure 6-4 illustrate how adjacency configuration controls media routing. In this example:

- Adjacency A connects to Peer1
- Adjacency B connects to Peer2a and 2b
- Adjacency C connects to Peer3

Adjacencies A and B are configured with media location 1. In other words, calls routed over them will use the same DBE (or set of DBEs) for media. Adjacency C is configured with media location 2.

Figure 6-3 How Adjacency Configuration Controls Media Routing



Now consider three calls: Peer1-Peer3, Peer1-Peer2a, and Peer2a-Peer2b. The media for these calls is routed as shown in Figure 6-4.

- The first call traverses two adjacencies with different media locations. Its media is relayed through two DBEs.
- The second call traverses two adjacencies with the same media location. Its media is relayed through a single DBE.
- The third call traverses a single adjacency with media by pass enabled. Its media is sent directly between the two peers without involving a DBE.

Figure 6-4 Media Routing for Three Calls: Peer1-Peer3, Peer1-Peer2a, and Peer2a-Peer2b



- 4. adjacency sip adjacency-name
- 5. force-signaling-peer

1. configure terminal 2. sbc service-name

attach 6.

3. sbe

7. exit

SUMMARY STEPS

The following sections describe implementing a SIP adjacency, depending on your implementation requirements:

Adjacencies are the key objects used to control signaling. The user defines one or more signaling adjacencies, which connect the Cisco Unified Border Element (SP Edition) to devices within that organization. Each incoming call is matched to an adjacency, and each outgoing call is routed out over

an adjacency. The adjacencies are then attached to the appropriate account. Adjacencies can be

associated with a media gateway DBE location, so that the most appropriate DBE can be selected to route

The default behavior for Cisco Unified Border Element (SP Edition) is to route INVITE requests to the device specified in the Request URI. If instead, the user wishes requests to be routed to the signaling peer, then 'force-next-hop' behavior should be enabled by configuring the **force-signaling-peer**

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- Configuring a SIP Adjacency, page 6-8

command on the outbound adjacency.

Implementing Adjacencies on Cisco Unified Border Element (SP Edition)

• Assigning SIP Adjacencies to Adjacency Groups, page 6-13

Configuring Force-Signaling-Peer Adjacency

This task configures a force-signaling-peer adjacency.

Note

How to Implement Adjacencies

media for a given call leg.

Chapter 6

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DETAILED STEPS

Command or Action		Purpose	
Step 1	configure terminal	Enables global configuration mode.	
	Example: Router# configure terminal		
Step 2	sbc service-name	Enters the mode of an SBC service.	
	Example: Router(config)# sbc umsbc-node10	Use the <i>service-name</i> argument to define the name of the service.	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.	
	Example: Router(config-sbc)# sbe		
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.	
	Example: Router(config-sbc-sbe)# adjacency sip 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the SIP adjacency.	
Step 5	force-signaling-peer	Forces SIP messages to go to the configured signaling peer.	
	Example: Router(config-sbc-sbe-adj-sip)# force-signaling-peer		
Step 6	attach	Attaches the adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# attach		
Step 7	exit	Exits the sip mode to the sbe mode.	
	Example: Router(config-sbc-sbe-adj-sip)# exit		

Configuring a SIP Adjacency

You can only modify adjacencies when the adjacency is detached. Before modifying an adjacency, you can detach the adjacency first with the **no attach** command. The adjacency stays in the going down state when a call is active or when the ping enable feature is running. During this state, existing calls are not torn down and new calls are not accepted. The adjacency does not go to detached state until all calls have ended. An adjacency cannot be attached until the adjacency is in detached state.

If you wish to override the option to wait till active calls on the adjacency end, the adjacency can be detached immediately using the following commands:

- no attach force abort—Executes a forced detach, tearing down calls without signaling their end.
- no attach force normal—Executes a forced detach, tearing down calls gracefully.

To check the state of the adjacency, you can use the show sbc sbe adjacencies command.



Adjacencies can only be modified when the status is detached. Before modifying an adjacency, use the **no attach** command first.



For User-to-Network Interface (UNI) registration support for a SIP inherit profile, you have the option of using the default value or a preset-access or a preset-core value. When using the default value for those adjacencies without specific per adjacency configuration, the **sip inherit profile preset-standard-non-ims** command in the SBE configuration mode (config-sbc-sbe) is applied to the adjacencies by default, and UNI registration support is enabled for this default configuration. When configuring a a preset-access or a preset-core value, use the **inherit profile preset-p-cscf-access** command on the adjacency facing subscribers and the **inherit profile preset-p-cscf-access** command on the adjacency facing the SIP proxy. If you use other combinations (for example, if both the inbound and outbound adjacencies are configured as preset-core, Cisco Unified Border Element (SP Edition) will not store the registration information, nor will it rewrite the Contact: header to make sure it's on the signaling path of future messages.

This task configures two session initiation protocol (SIP) adjacencies. The first adjacency is configured for a gateway/endpoint. The second adjacency is configured with proxy/softswitch.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. sip inherit profile {preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-standard-non-ims}
- 5. adjacency sip adjacency-name
- 6. signaling-address ipv4 ipv4_IP_address
- 7. signaling-port port_num
- 8. remote-address ipv4 ipv4_IP_address/prefix
- 9. signaling-peer peer_address
- 10. signaling-peer-port port_num
- **11.** account account-name
- 12. registration rewrite-register
- 13. attach
- 14. exit
- 15. adjacency sip adjacency-name
- 16. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}
- 17. signaling-address ipv4 ipv4_IP_address
- 18. signaling-port port_num

- **19**. **remote-address ipv4** *ipv4_IP_address/prefix*
- **20**. fast-register disable
- **21**. **signaling-peer** *peer_name*
- 22. signaling-peer-port *port_num*
- **23.** account account-name
- 24. registration target address host_address
- **25**. registration target port port_num
- 26. attach
- 27. exit
- 28. end
- **29**. show

DETAILED STEPS

	Command or Action	Purpose	
Step 1	configure terminal	Enables global configuration mode.	
	Example: Router# configure terminal		
Step 2	sbc service-name	Enters the mode of an SBC service.	
	Example:Router Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.	
	Example: Router(config-sbc)# sbe		
Step 4	<pre>sip inherit profile {preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal preset-p-cscf-access preset-p-cscf-core preset-standard-non-ims}</pre>	Configures the global default inherit profile for all adjacencies.	
	Example: Router(config-sbc-sbe)# sip inherit profile preset-standard-non-ims		
Step 5	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.	
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.	
Step 6	<pre>signaling-address ipv4 ipv4_IP_address</pre>	Specifies the local IPv4 signaling address of the SIP adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 88.88.141.3		

	Command or Action	Purpose	
Step 7	signaling-port port_num	Specifies the local signaling port of the SIP adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060		
Step 8	<pre>remote-address ipv4 ipv4_IP_address/prefix</pre>	Restricts the set of remote signaling peers contacted over the adjacency to those with the given IP address prefix.	
	Example: Router(config-sbc-sbe-adj-sip)# remote-address ipv4 10.10.121.0/24		
Step 9	signaling-peer peer_address	Specifies the remote signaling peer for the SIP adjacency to use.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer 10.10.121.10</pre>		
Step 10	<pre>signaling-peer-port port_num</pre>	Specifies the remote signaling-peer port for the SIP adjacency to use.	
	Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060		
Step 11	account account_name	Defines the SIP adjacency as belonging to an account on an SBE.	
	Example: Router(config-sbc-sbe-adj-sip)# account iosgw		
Step 12	registration rewrite-register	Configures SIP REGISTER request rewriting.	
	Example: Router(config-sbc-sbe-adj-sip)# registration rewrite-register		
Step 13	attach	Attaches the adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# attach		
Step 14	exit	Exits adj-sip mode to sbe mode.	
	Example: Router(config-sbc-sbe-adj-sip)# exit		
Step 15	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.	
	Example: Router(config-sbc-sbe)# adjacency sip sipPROXY	Use the <i>adjacency-name</i> argument to define the name of the service.	

	Command or Action	Purpose	
Step 16	<pre>inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}</pre>	Configures an inherit profile for the SIP adjacency.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-standard-non-ims</pre>		
Step 17	<pre>signaling-address ipv4 ipv4_IP_address</pre>	Specifies the local IPv4 signaling address of the SIP adjacency.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 88.88.141.11</pre>		
Step 18	<pre>signaling-port port_num</pre>	Specifies the local signaling port of the SIP adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060		
Step 19	<pre>remote-address ipv4 ipv4_IP_address/prefix</pre>	Restricts the set of remote signaling peers contacted over the adjacency to those with the given IP address prefix.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# remote-address ipv4 200.200.200.0/24</pre>		
Step 20	fast-register disable	Disables fast register support on the SIP adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# fast-register disable		
Step 21	<pre>signaling-peer peer_address</pre>	Specifies the remote signaling peer for the SIP adjacency to use.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer 200.200.200.98</pre>		
Step 22	<pre>signaling-peer-port port_num</pre>	Specifies the remote signaling-peer port for the SIP adjacency to use.	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060</pre>		
Step 23	account account_name	Defines the SIP adjacency as belonging to an account on an SBE.	
	Example: Router(config-sbc-sbe-adj-sip)# account COREvlan		

Command or Action		Purpose	
Step 24	registration target address host_address	Sets the address to use if rewriting an outbound SIP REGISTER request.	
	Example: Router(config-sbc-sbe-adj-sip)# registration target address 200.200.200.98		
Step 25	registration target port port_num	Sets the port to use if rewriting an outbound SIP REGISTER request.	
	Example: Router(config-sbc-sbe-adj-sip)# registration target port 5060		
Step 26	attach	Attaches the adjacency.	
	Example: Router(config-sbc-sbe-adj-sip)# attach		
Step 27	exit	Exits adj-sip mode to sbe mode.	
	Example: Router(config-sbc-sbe-adj-sip)# exit		
Step 28	end	Exits the sbe mode and returns to Privileged EXEC mode.	
	Example: Router(config-sbc-sbe)# end		
Step 29	show	Shows contents of configuration.	
	Example: Router# show		

Assigning SIP Adjacencies to Adjacency Groups

Use the procedure in this section to assign an SIP adjacency to an adjacency group.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. group adjacency-group-name
- 6. end
- 7. show

DETAILED STEPS

	Command or Action	Purpose	
Step 1	configure terminal	Enables global configuration mode.	
	Example: Router# configure terminal		
Step 2	sbc service-name	Enters the mode of an SBC service.	
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.	
	Example: Router(config-sbc)# sbe		
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.	
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.	
Step 5	group adjacency-group-name	Assigns the SIP adjacency to an adjacency group.	
	Example: Router(config-sbc-sbe-adj-sip)# group InternetEth0	Use the <i>adjacency-group-name</i> argument to define the group name.	
Step 6	end	Exits adj-sip mode to sbe mode and returns to Privileged EXEC mode.	
	Example: Router(config-sbc-sbe-adj-sip)# end		
Step 7	show	Shows contents of configuration.	
	Example: Router# show		

Configuration Examples for Implementing Adjacencies

This section provides the following configuration example:

• Configuring a SIP Adjacency: Example, page 6-15

Configuring a SIP Adjacency: Example

The following example configures two SIP adjacencies. The first adjacency is configured for a gateway/endpoint. The second adjacency is configured with proxy/softswitch.

1. Activate SBE, as follows:

```
sbc sip-signal
sbe
activate
exit
```

2. Activate DBE, as follows:

```
sbc mySbc dbe
media-address ipv4 88.88.141.2
activate
exit
```

3. Create the SIP adjacencies, as follows:

```
sbc sip-signal
sbe
```

4. Create the SIP adjacency for gateway/endpoint:

```
adjacency sip sipGW
signaling-address ipv4 88.88.141.3
signaling-port 5060
remote-address ipv4 10.10.121.0/24
signaling-peer 10.10.121.10
signaling-peer-port 5060
account iosgw
registration rewrite-register
attach
exit
!
```

5. Create the SIP adjacency for proxy/softswitch:

```
adjacency sip sipPROXY
signaling-address ipv4 88.88.141.11
signaling-port 5060
remote-address ipv4 200.200.200.0/24
fast-register disable
signaling-peer 200.200.200.98
signaling-peer-port 5060
account COREvlan
registration target address 200.200.200.98
registration target port 5060
attach
```

SIP UAS Failure Detection

A User Agent Server (UAS) is a logical entity that generates a response to a SIP request. UAS failure detection is used to periodically monitor the state of a SIP network entity specified as the signaling peer on a SIP adjacency. SIP OPTIONS messages are sent to these network entities as a ping mechanism and a response from the device is expected. If a response is not received from the device, it is considered unreachable and removed from the routing calculations. Calls which cannot be routed through an alternate device are immediately responded to with a 604 Does Not Exist Anywhere message.

Cisco Unified Border Element (SP Edition) by default acts as an UAS that responds to OPTION pings when OPTION pings are sent to it. SIP UAS Failure Detection enables Cisco Unified Border Element (SP Edition) to send a SIP OPTIONS message to the device specified in the SIP Adjacency Destination Address. If an acceptable response is received within the SIP transaction timeout period then the routing tables are updated and the device is considered routable.

A ping failure occurs when no acceptable response is received within the SIP transaction timeout period. If ping-fail-count failures occur, then the device is considered to be unreachable. The signaling peer is considered offline as far as routing is concerned. Cisco Unified Border Element (SP Edition) sends pings at the rate specified in the period.

Note

When the SBC has a TCP-based adjacency with OPTION ping enabled and that adjacency does not have a valid peer with which a TCP connection can be established, then that adjacency must be in the "no attach" state. This prevents the SBC from attempting to set up a TCP connection to a non-existent peer to send an OPTIONS ping message.

Use the procedure in this section to configure SIP UAS Failure Detection:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ping-enable
- 6. ping-interval interval
- 7. ping-lifetime *duration*
- 8. ping-fail-count fail-count
- 9. exit

DETAILED STEPS

Command or Action		Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	ping-enable	Configures the adjacency to poll its remote peer by sending SIP OPTIONS pings to it and enters the ping option submode
	Example: Router(config-sbc-sbe-adj-sip)# ping-enable	submode.
Step 6	<pre>ping-interval interval</pre>	Configures the interval between SIP OPTIONS pings sent to the remote peer.
	Example: Router(config-sbc-sbe-adj-sip-ping)# ping-interval 100	
Step 7	ping-lifetime duration	Configures the duration for which SBC waits for a response to an options ping for the adjacency.
	Example: Router(config-sbc-sbe-adj-sip-ping)# ping-lifetime 100	
Step 8	<pre>ping-fail-count fail-count</pre>	Configures the number of consecutive pings that must fail before the adjacencies peer is deemed to be unavailable.
	Example: Router(config-sbc-sbe-adj-sip-ping)# ping-fail-count 10	
Step 9	exit	Exits adj-sip mode to sbe mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

SIP UAS Failure Detection: Example

In the following configuration example, PING is enabled on each of three adjacencies. A round robin call policy is set so that calls are distributed between the three adjacencies in a weighted random manner. If a UAS is unreachable, calls will be distributed between the remaining two adjacencies.

```
sbc mySBC
sbe
adjacency sip CallMgrA
signaling-address ipv4 88.103.29.100
remote-address ipv4 200.200.200.0 255.255.255.0
signaling-peer 200.200.200.118
ping-enable
ping-interval 5
ping-fail-count 3
ping-lifetime 32
attach
adjacency sip CallMgrB
signaling-address ipv4 88.103.29.100
remote-address ipv4 200.200.200.0 255.255.255.0
signaling-peer 200.200.200.117
```

```
ping-enable
        ping-interval 5
       ping-fail-count 3
       ping-lifetime 32
      attach
    adjacency sip CallMgrC
      signaling-address ipv4 88.103.29.100
      remote-address ipv4 200.200.200.0 255.255.255.0
      signaling-peer 200.200.200.200.115
      ping-enable
       ping-interval 5
       ping-fail-count 3
       ping-lifetime 32
      attach
call-policy-set 1
      first-call-routing-table DestAddr
      rtg-dst-address-table DestAddr
        entry 1
          action next-table RoundRobin
         match-address 12
         prefix
      rtg-round-robin-table RoundRobin
        entry 1
          action complete
          dst-adjacency CallMgrB
        entry 2
          action complete
          dst-adjacency CallMgrC
        entry 3
          action complete
          dst-adjacency CallMgrA
      complete
    active-call-policy-set 1
```

SIP Outbound Flood Protection

SIP Outbound Flood Protection protects other network elements from excessively high valid traffic in unusual situations, such as a protection from a flood of generated BYE messages when a neighboring network element fails.

SIP Outbound Flood Protection sets a maximum rate of outgoing request messages and prevents the rate of outgoing request messages exceeding this maximum rate. If the limit is reached, outgoing requests are failed or dropped instead.

SIP Outbound Flood Protection is an addition to the normal CAC policy mechanisms and does not replace CAC policy. CAC policy allows fine grain control of calls, like, for example, rate limiting of INVITE requests at configurable scopes. SIP Outbound Flood Protection is intended to provide a simple overall rate limit for outgoing requests and is especially useful for requests that currently do not involve CAC policy (such as BYE requests).

Flood protection may be required in the following situations:

• Adjacent network element terminating — If an adjacent network element terminates (either normally or due to error) Cisco Unified Border Element (SP Edition) is likely to detect that the calls that used this element are dead at approximately the same time and attempt to tear the calls down. With many active calls this can generate a flood of BYE requests (normally two BYEs for each call).

Rather than allow these BYE messages to transiently overload other network signaling elements the network administrator may prefer to drop or fail some BYE requests at the Cisco Unified Border Element (SP Edition).

• Local removal of configuration in the Cisco Unified Border Element (SP Edition) — If a SIP adjacency is unconfigured using normal deactivation mode then BYE requests will be sent for all active calls using the adjacency before they are destroyed.

Again it may be desirable for to limit the rate of outgoing requests prevent other network elements getting overloaded.

Use the procedure in this section to configure SIP Outbound Flood Protection:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. outbound-flood-rate *rate*
- 6. exit

DETAILED STEPS

	Command or Action	Purpose		
Step 1	configure terminal	Enables global configuration mode.		
	Example: Router# configure terminal			
Step 2	sbc service-name	Enters the mode of an SBC service.		
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.		
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.		
	Example: Router(config-sbc)# sbe			
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.		
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.		

	Command or Action	Purpose
Step 5	<pre>outbound-flood-rate rate Example: Router(config-sbc-sbe-adj-sip)# outbound-flood-rate 1000</pre>	Configures the maximum desired rate of outbound request signals on this adjacency (excluding ACK/PRACK requests) in signals per second.
Step 6	exit	Exits adj-sip mode to sbe mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

SIP Outbound Flood Protection: Example

The following configuration example sets an outbound flood rate of 100 outbound request signals per second.

```
sbc mySBC
sbe
adjacency sip CallMgrA
signaling-address ipv4 88.103.29.100
remote-address ipv4 200.200.200.0 255.255.255.0
signaling-peer 200.200.200.118
outbound-flood rate 100
attach
```

SIP Over TLS

This section describes the concepts for SIP over Transport Layer Security (TLS). This section contains the following topics:

- Security Configuration on an Adjacency, page 6-21
- SIP Over TLS Overview, page 6-21
- User Agent Server-Side Processing, page 6-24
- Routing Processing, page 6-24
- User Agent Client-Side Processing, page 6-24
- Configurable Mutual TLS Authentication Per Interface, page 6-24
- TLS Transport Parameter in Record-Route Headers, page 6-26
- Configuring SIP Over TLS on Cisco Unified Border Element (SP Edition), page 6-27
- SIP Over TLS Configuration Example, page 6-29
- SIP Over TLS Verification, page 6-30
- Enabling the Conversion of SIPS URIs to SIP URIs on a Trusted-Unencrypted Adjacency, page 6-30

Security Configuration on an Adjacency

You can independently configure client and server security support on a SIP adjacency, using the following options:

- Untrusted—Specifies that this adjacency is not secured by any means. Only unsecured calls (not the calls to SIPS URIs) are made out of this adjacency.
- Untrusted-Encrypted—Specifies that the adjacency is untrusted and SSL/TLS encryption is used.
- Trusted-Encrypted—Specifies that the encrypted signaling is used to ensure security on this adjacency. The default certificate and key of the router are used for encryption. Only secure calls (calls to SIPS URIs) are made out of this adjacency.
- Trusted-Unencrypted—Specifies that a non-encryption mechanism is used to guarantee secure signaling for all messages on this adjacency. For example, this mechanism could be a single trusted physical link. Either secure or unsecured calls are made out of this adjacency. This configuration allows endpoints that do not support encryption to participate in secure SIP calls.

SIP Over TLS Overview

SIP Over Transport Layer Security (TLS) encryption provides a secure, encrypted transport to carry all SIP messages from the caller to the callee's domain. From there, the request is sent securely to the callee.

Cisco Unified Border Element (SP Edition) provides the following support for SIP Over TLS:

- Secured SIP calls can flow through Cisco Unified Border Element (SP Edition).
- A SIP adjacency can be secured by encryption or by another mechanism (for example, a single trusted physical-layer link or an interface to a trusted network).
- Inbound and outbound connections are immediately closed if a remote peer attempts to use encryption when encryption is not supported.
- Inbound and outbound connections are immediately closed if a remote peer fails to use encryption when encryption is required.
- You can view the level of security support configured for a given SIP adjacency by using the **show sbc** *sbc-name* **sbe adjacencies** *adj name* **detail** command.
- Calls received on untrusted adjacencies are not routed over outbound secure-encrypted adjacencies.
- Adjacencies secured by means of encryption can listen by default on port 5061. The port is configured to a different value.
- The fully-qualified domain name (FQDN) in the certificate offered by the remote peer is checked against the domain from which the request is received. The signal is dropped if the two do not match.
- Advanced Encryption Standard (AES) 128-bit Secure Hash Algorithm (SHA) is supported.
- The PKI HA updates the standby router with the certificate and trustpoint configuration changes.

The following are main security factors that are used in routing or rejecting a call:

- Calls to a SIPS URI must be secure. Calls to a SIP URI do not have to be secure.
- Signals received on a trusted adjacency are considered secure. Signals received on an untrusted adjacency are considered unsecured.

The following security factors apply to untrusted encrypted adjacencies:

- Secure calls may not be received on untrusted adjacencies of any type.
 - Cisco Unified Border Element (SP Edition) allows unsecured calls to be received over the untrusted encrypted adjacency.
 - Cisco Unified Border Element (SP Edition) rejects secured call that it receives over the untrusted encrypted adjacency.
- Secure calls cannot be routed to untrusted adjacencies.
 - Cisco Unified Border Element (SP Edition) can route unsecured calls over the untrusted encrypted adjacency.
 - Cisco Unified Border Element (SP Edition) does not route secured calls over the untrusted encrypted adjacency.

Table 6-1 and Table 6-2 summarize how Cisco Unified Border Element (SP Edition) handles inbound and outbound calls based on the call type, trust relationship, and encryption.

Table 6-1 Inbound Call Policy

	Trusted Adjacency		Untrusted Adjacency	
SIP Call Type	Encrypted	Unencrypted	Encrypted	Unencrypted
Secure SIP	Allow	Allow	Reject	Reject
Unsecured SIP	Allow	Allow	Allow	Allow

Table 6-2 Outbound Call Policy

	Trusted Adjacency		Untrusted Adjacency	
SIP Call Type	Encrypted	Unencrypted	Encrypted	Unencrypted
Secure SIP	Allow	Allow	Reject	Reject
Unsecured SIP	Allow	Allow	Allow	Allow

Table 6-3 and Table 6-4 summarize how Cisco Unified Border Element (SP Edition) handles inbound and outbound registrations based on the registration type, trust relationship, and encryption.

Table 6-3 Inbound Registration Policy

SIP Registration	Trusted Adjacency		Untrusted Adjacency	
Туре	Encrypted	Unencrypted	Encrypted	Unencrypted
Secure SIP	Allow	Allow	Reject	Reject
Unsecured SIP	Allow	Allow	Allow	Allow

SIP Registration Type	Trusted Adjacency		Untrusted Adjacency	
	Encrypted	Unencrypted	Encrypted	Unencrypted
Secure SIP	Allow	Allow or Reject (depending on the configuration)	Reject	Reject
Unsecured SIP	Allow	Allow	Allow	Allow

 Table 6-4
 Outbound Registration Policy

For the SBC to be able to forward Secure SIP (SIPS) registrations to a trusted-unencrypted adjacency, all the following conditions must be met:

- The source adjacency must have a non-IP Multimedia Subsystem (non-IMS) or non-IMS access adjacency profile that specifies tracking of the registration state.
- The destination adjacency must have a non-IMS adjacency profile.
- The destination adjacency must be configured to accept a SIPS URI registration. This procedure is explained in the ?\$paranum>Enabling the Conversion of SIPS URIs to SIP URIs on a Trusted-Unencrypted Adjacency? section on page 6-30.

When the SBC forwards secure registrations to a trusted-unencrypted adjacency that meets these conditions, the outbound registration is modified as follows:

- The Address of Record (AoR) in the To and From headers is converted from a SIPS URI to a SIP URI.
- The Request URI is converted from a SIPS URI to a SIP URI. Note that the Request URI may not be identical to the AoR.
- The URIs in the Contact headers are converted from SIPS to SIP.
- The URIs in Record Route headers are passed through unchanged. Note that according to RFC 3261, Record Route headers must be ignored on receipt if they are present in REGISTER messages.
- The URIs in other SIP headers are passed through unchanged.



The SBC rejects registrations that contain a mix of SIP URIs and SIPS URIs in their AoRs and contacts. On receipt of the REGISTER response, the SBC reverses the changes and passes back SIPS URIs in the response forwarded to the endpoint.

The following are restrictions for this feature:

- There is no change in the processing of non-INVITE messages, such as SUBSCRIBE, NOTIFY, and PUBLISH, by the SBC. For these messages, the SBC does not convert SIPS URIs to SIP URIs.
- The SBC does not support registrations to trusted-unencrypted adjacencies in scenarios where either the inbound adjacency or the outbound adjacency has an IMS profile.

L

User Agent Server-Side Processing

Inbound requests are marked according to two factors: whether the caller is trusted, and whether the call is intended for a secure target.

The caller-trust is determined in the following ways:

- SIP requests arriving over trusted adjacencies are marked as trusted.
- SIP requests arriving over untrusted adjacencies are marked as untrusted.

Desired target security is determined in the following ways:

- Requests for SIPS URIs are marked to require the outbound-security.
- Requests for SIP URIs are marked not to require the outbound-security.

Inbound requests are rejected if the caller is untrusted and the target requires security. All other combinations are forwarded to routing processing.

Routing Processing

The Routing Policy System (RPS) policy determines where the requests are routed next, with the following default behaviors:

- If a call requires the outbound security, the RPS considers only the trusted outbound adjacencies.
- If a call does not require the outbound security, the RPS considers only untrusted or trusted-unencrypted outbound adjacencies.

If the RPS is unable to find a suitable outbound adjacency for a call, the call is rejected.

User Agent Client-Side Processing

Outbound adjacencies preserve the URI scheme of the original request, ensuring that if a call is originally targeted at a SIPS URI, it is sent out to a SIPS URI. Or, if the call is originally targeted at a SIP URI, it is sent out to a SIP URI.

Upon receipt of 3xx class responses and target-refresh indications, the contact set is examined. Untrusted adjacencies do not permit the target of the call to be rerouted to a SIPS target. Likewise, trusted adjacencies do not permit the target of the call to be rerouted to a SIP target. If this is attempted by the remote peer, the call is brought down.

Configurable Mutual TLS Authentication Per Interface

The Configurable Mutual TLS Authentication Per Interface feature helps you to configure unilateral or mutual TLS authentication on a per adjacency basis for SIP over TLS calls.

In a SIP over TLS call, SBC can either be a TLS client side or TLS server side. This feature is relevant only when the SBC is a TLS server side.

While negotiating a TLS connection, the server side sends its certificate to the client side to perform the server authentication. After the server authentication, the server may require a certificate from the client for client authentication. When client authentication is not used, the authentication is referred to as a unilateral authentication. When both — server and client — requires authentication, then it is referred to as a mutual authentication.

The message flow diagram, Figure 6-5, illustrates the negotiation process of a TLS connection. The bold line represents the messages required when mutual authentication is enabled on the server side:





When SBC acts as TLS client side, it can automatically negotiate with the server side to perform the client authentication. But when SBC acts as TLS server side, you need to configure SBC so that SBC can decide whether to send a CertificateRequest message to the client side to get the client's certificate to do client authentication.

Use the tls mutual-authentication command to configure mutual authentication.

Restrictions and Limitations—Configurable Mutual TLS Authentication

- The configuration on a SIP adjacency cannot be modified while the adjacency is attached.
- The security configuration of the adjacency must be trusted encrypted or untrusted encrypted.
- Multiple TLS-enabled adjacencies that use the same local address and port must have the same configuration. Otherwise, the configuration will be rejected and an error message will be displaced on the console.
- Configuring trust points on a per adjacency basis is not possible because SBC uses global trust points to validate peer's certification. This limitation will not pose any limitation for certificate verification on SBC because, SBC automatically searches for a matching certificate from its global trust points.

- SBC only supports one certificate while SBC is a TLS server side. It is not possible to configure different certificates for each adjacency. The certificate is picked from the primary trust point.
- Certificate chain is not synchronized in SSO config-sync redundancy mode and hence the TLS certificates are not replicated to the standby. The incoming TLS calls might fail because of non-availability of TLS certificates.

TLS Transport Parameter in Record-Route Headers

This feature allows you to add a **transport=tls** parameter to the SBC-originated Contact and Record-Route headers when using TLS. This feature is applicable when the security for the SBC inbound adjacency is configured as untrusted-encrypted.

The transport=tls parameter was deprecated in RFC3261 for better interoperability. With the implementation of RFC3261, the Contact and Record-Route header of 200(INVITE), back to caller, would use SIP URI without the transport parameter such as *Contact: <sip:192.168.1.1:5060>*, *Record-Route: <sip:192.168.1.1:5060;1r>*. Because of this, the subsequent mid-dialog requests—re-INVITE—are sent using TCP or UDP based on the SIP URI instead of TLS. Since SBC is expecting a TLS message on the port, the call is dropped.

Figure 6-6 shows the message flow where the SIP call is received over TLS, but the call was dropped. The ACK in response to the 2000K (TLS) message is sent from the SIP to SBC using TCP or UDP.



Figure 6-6 Message Flow During a SIP Call Over TLS

To avoid call drops, the caller is forced to use the TLS transport for the ACK by adding the transport=tls parameter. This feature is controlled on a per adjacency basis.

Use header-name [contact [add [tls-param]] | from{passthrough} | to{passthrough}] command in (config-sbc-sbe-adj-sip) mode to configure the transport=tls parameter in the Contact and Record-Route header.

Configuring SIP Over TLS on Cisco Unified Border Element (SP Edition)

Use the procedure in this section to configure SIP over TLS on Cisco Unified Border Element (SP Edition):

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. security trusted-encrypted
- 6. redirect-mode {pass-through | recurse}
- 7. authentication nonce
- 8. signaling-address ipv4 ipv4_IP_address
- 9. signaling-port port-num
- **10**. remote-address ipv4 *ip-address ip-mask*
- **11.** signaling-peer peer-name
- 12. signaling-peer-port port-num
- 13. dbe-location-id dbe-location-id
- 14. reg-min-expiry period
- **15.** attach *force* [abort | normal]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.

	Command or Action	Purpose
Step 5	security trusted-encrypted	Configures transport-level security on a Session Initiation Protocol (SIP) adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# security trusted-encrypted	
Step 6	redirect-mode {pass-through recurse}	Configures the behavior of SBC on receipt of a 3xx response to an invite from the SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# redirect-mode recurse</pre>	
Step 7	authentication nonce timeout	Configures the authentication nonce timeout for a SIP adjacency.
	Example: Router(config-sbe-adj-sip)# authentication nonce timeout 10	
Step 8	signaling-address ipv4 <i>ipv4_IP_address</i>	Defines the local IPv4 signaling address of a SIP or an H.323 adjacency.
	Example:	
	Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 10.10.10.10	
Step 9	signaling-port <i>port-num</i>	Defines the local port of signaling address of a SIP adjacency.
	Example:	
	Router(config-sbc-sbe-adj-sip)# signaling-port 5000	
Step 10	remote-address ipv4 <i>ip-address ip-mask</i> Fxamnle:	Configures a SIP adjacency to restrict the set of remote signaling peers that can be contacted over the adjacency to those with the given IP address prefix.
	Router(config-sbc-sbe-adj-sip)# remote-address ipv4 36.36.36.20 255.255.255.0	
Step 11	signaling-peer peer-name	Configures a SIP adjacency to use the given remote signaling-peer.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer 10.1.2.3</pre>	
Step 12	signaling-peer-port <i>port-num</i>	Configures a SIP adjacency to use the given remote signaling-peer's port.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 123</pre>	
Step 13	dbe-location-id dbe-location-id	Configures an adjacency to use a given media gateway DBE location when routing media.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# dbe-location-id 1</pre>	

	Command or Action	Purpose
Step 14	reg-min-expiry period	Configures the minimum registration period in seconds on the SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# reg-min-expiry 300</pre>	
Step 15	attach force [abort normal]	Attaches an adjacency to an account on an SBE.
	Example: Router(config-sbc-sbe-adj-h323)# attach	

SIP Over TLS Configuration Example

The following example shows a SIP over TLS configuration:

```
crypto pki trustpoint CA
 enrollment terminal
 serial-number
 subject-name ST=Some-State, C=AU, O=Internet Widgits Pty Ltd revocation-check none
rsakevpair the default !
crypto pki certificate chain CA
 certificate 01
  308201D7 30820140 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  46310B30 09060355 04061302 5553310C 300A0603 55040813 03525450 310B3009
  06035504 07130253 4A310E30 0C060355 040A1305 43495343 4F310C30 0A060355
  040B1303 53424330 1E170D30 39303230 35313030 3832385A 170D3130 30323035
  31303038 32385A30 45311330 11060355 0408130A 536F6D65 2D537461 7465310B
  30090603 55040613 02415531 21301F06 0355040A 1318496E 7465726E 65742057
  69646769 74732050 7479204C 7464305C 300D0609 2A864886 F70D0101 01050003
  4B003048 024100DC 18647810 B82F521B 40762B30 31646EB1 D567F0A6 E38DAD77
  1C41D825 E5274FFC A1F59E98 DCDFA617 161EA4D4 DBDC06E9 E1142752 9212D34D
  646E6B37 99D26502 03010001 A31A3018 30090603 551D1304 02300030 0B060355
  1D0F0404 030205A0 300D0609 2A864886 F70D0101 04050003 81810084 7E9A479B
  018F93F0 E683AA41 D3303705 6D89D44B 7798BD5F 15BCFAD5 EF55D72E 03365CD9
  BBCD955E 3C6D78B3 8E8C0675 772A7DE2 BCFBD6DF 760F9683 F0AB6F62 A87D9AC1
  AB2EA7E0 D831D33D 2F54582F 9E39F81D CBA33BD9 2466296C 4DCDAD0C 7D697AF7
  797AFEAA 05C3021F A7E89044 EA1796DC F422C82E 2B3894F6 3B98A7
       quit
 certificate ca 00F2D75C678DC7F7F2
  3082021A 30820183 A0030201 02020900 F2D75C67 8DC7F7F2 300D0609 2A864886
  F70D0101 04050030 46310B30 09060355 04061302 5553310C 300A0603 55040813
  03525450 310B3009 06035504 07130253 4A310E30 0C060355 040A1305 43495343
  4F310C30 0A060355 040B1303 53424330 1E170D30 39303230 35303931 3032395A
  170D3134 30323034 30393130 32395A30 46310B30 09060355 04061302 5553310C
  300A0603 55040813 03525450 310B3009 06035504 07130253 4A310E30 0C060355
  040A1305 43495343 4F310C30 0A060355 040B1303 53424330 819F300D 06092A86
  4886F70D 01010105 0003818D 00308189 02818100 BD3DBEEE A8CB6C51 9E2BBEC4
  35C2644F 92055B30 3543CA9D A1E1C0CB F59A2490 9296304D 43C19913 2A12EA80
  BDC6A1E3 0C164059 2C0DF132 E4AFF260 E88F38DC F23E866C DAFDD1BD F888BE90
  B74C49DA 4712E1E2 E249F444 FB3226B2 A5963DCD E75467B3 83669794 13BB8E7B
  CAFE3830 85091839 9658999B C72395E1 07AB35D1 02030100 01A31030 0E300C06
  03551D13 04053003 0101FF30 0D06092A 864886F7 0D010104 05000381 8100A7E5
  662FDE66 01FC63BA 399D1D17 0336C35B F9D9AEAF 87DA9E05 6AD13B90 D11CB984
  9B90FF8E 123F03B3 4E035D6B AC79D399 FF92A09C 2E62B759 E716D1D5 ABA46796
  41BB570F 96B7EE47 FB779AD4 0C8790FC 15FC65D6 47F60BE4 EB498B63 6DC2FBD3
  9DD51D82 0EB80125 D5A8F71B F7B61A63 5B601A6D FEFCAEB6 B33BF38B 9A10
```

The Cisco Unified Border Element (SP Edition) configuration example is illustrated here.

```
Router# configure
Router(config)# sbc sbc-3
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# security trusted-encrypted
Router(config-sbc-sbe-adj-sip)# redirect-mode pass-through
Router(config-sbc-sbe-adj-sip)# authentication nonce timeout 300
Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 10.130.10.25
Router(config-sbc-sbe-adj-sip)# remote-address ipv4 10.74.49.145 255.255.255.255
Router(config-sbc-sbe-adj-sip)# signaling-peer 10.74.49.145
Router(config-sbc-sbe-adj-sip)# signaling-peer 10.74.49.145
Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060
Router(config-sbc-sbe-adj-sip)# dbe-location-id 4294967295
Router(config-sbc-sbe-adj-sip)# reg-min-expiry 3000
Router(config-sbc-sbe-adj-h323)# attach
```

SIP Over TLS Verification

Use the following commands to check certificates on the node:

show crypto pki certificates—displays information about your certificate, the certification authority certificate (CA), and any registration authority (RA) certificates.

show crypto key pubkey-chain rsa—enters public key configuration mode (so you can manually specify and show other devices' RSA public keys).

show crypto key mypubkey rsa—displays the RSA public keys of your router.

Enabling the Conversion of SIPS URIs to SIP URIs on a Trusted-Unencrypted Adjacency

Use the procedure described in this section to enable the conversion of SIPS URIs to SIP URIs on a trusted-unencrypted adjacency. Performing this procedure is one of the requirements for configuring the SBC to forward secure registrations to a trusted-unencrypted adjacency. See ?\$paranum>SIP Over TLS Overview? section on page 6-21 for more information about this feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. security trusted-unencrypted
- 6. registration unencrypted-convert
- 7. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
		adjacency-name—Name of the adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	
Step 5	security trusted-unencrypted	Configures transport-level security on the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# security trusted-encrypted	
Step 6	registration unencrypted-convert	Enables the conversion of SIPS URIs to SIP URIs on the trusted-unencrypted adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# registration unencrypted-convert	
Step 7	end	Returns to the privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	

In the following example, the output of the **show sbc adjacencies** command shows that conversion of SIPS URIs to SIP URIs is enabled:

```
Router# show sbc MySBC sbe adjacencies ADJ1 detail

SBC Service MySBC

Adjacency ADJ1 (SIP)

Status: Attached

Signaling address: 192.0.2.36.1:5060, VRF sidd_sipp1

IPsec server port: 0

Signaling-peer: 192.0.2.37.1:5060 (Default)

.

.

Media Bypass Tag List:

Tag 1: tag1
```

Tag 2: tag2 Media Bypass Max Out Data Length: 1024 Register unencrypted covert: Enabled

SIP Peer Availability Detection

The SBC supports the SIP peer availability detection (OPTIONs ping) functionality. The SBC periodically sends an OPTION request to a configured peer. If the peer fails to respond to a set number of OPTION requests, the peer is declared dead, and the calls are routed through other peers.

To avoid congestion, when ping suppression is enabled, and if signaling traffic exchange between peers is active, the OPTIONS pings are not used to check peer availability.

Restrictions for SIP Peer Availability Detection

The SIP Peer Availability Detection feature has the following restrictions:

- The OPTION requests will use the SIP method congestion response codes.
- If the number of OPTIONS messages to the peer are reduced, the time taken to detect dead peer by the SBC increases substantially.

Configuring SIP Peer Availability Detection

Use the procedure described in this section to configure the detection of SIP peer availability:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ping-enable
- 6. ping-bad-rsp-codes ranges
- 7. ping-suppression
- 8. exit
- 9. end
- 10. show sbc sbc-name sbe adjacencies adjacency-name Detail
DETAILED STEPS

	Command or Action	Purpose		
Step 1	configure terminal	Enables the global configuration mode.		
	Example: Router# configure terminal			
Step 2	sbc service-name	Enters the mode of an SBC service.		
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.		
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.		
	Example: Router(config-sbc)# sbe			
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.		
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.		
Step 5	ping-enable	Configures the adjacency to poll the adjacency's remote peer by sending SIP OPTION pings to it, and enters the Ping option submode.		
	Example: Router(config-sbc-sbe-adj-sip)# ping-enable			
Step 6	ping-bad-rsp-codes ranges	Configures the congestion response codes on a SIP adjacency by sending SIP OPTION pings to the adjacency.		
	Example: Router(config-sbc-sbe-adj-sip-ping)# ping-bad-rsp-codes ranges 300,398	Use the <i>ranges</i> argument to specify the response code range (The range can be 300 to 399).		
Step 7	ping-suppression options	(Optional) Configures the SBC to ping, when required, on a SIP adjacency.		
	Example: Router(config-sbc-sbe-adj-sip-ping)#	<i>options</i> specifies one of the following strings used for ping suppression:		
	ping-suppression odd-request	• ood-request—The SBC considers a peer as reachable when an out-of-dialog or dialog-creating request is received, excluding the OPTIONS and REGISTER messages.		
		• ood-response—The SBC considers a peer as reachable when an out-of-dialog or dialog-creating 2xx response is received, excluding OPTIONS and REGISTER messages.		
		• ind-request—The SBC considers a peer as reachable when an in-dialog request is received.		
		• ind-response—The SBC considers a peer as reachable when an in-dialog 2xx response is received.		

	Command or Action	Purpose		
Step 8	exit	Exits the adj-sip-ping mode, and moves to the adj-sip mode.		
	Example: Router(config-sbc-sbe-adj-sip-ping)# exit			
Step 9	end	Exits the SBE mode and returns to the Privileged EXEC mode.		
	Example: Router(config-sbc-sbe)# end			
Step 10	show sbc sbc-name sbe adjacencies adjacency-name detail	Displays the details pertaining to the specified adjacency.		
	Example: Router# show sbc mysbc sbe adjacencies sipGW detail			

Example

The following example shows how to configure the congestion response codes on a SIP adjacency by sending SIP OPTIONS pings:

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip SipAdj1
Router(config-sbc-sbe-adj-sip)# ping-enable
Router(config-sbc-sbe-adj-sip-ping)# ping-bad-rsp-codes ranges 300,398
Router(config-sbc-sbe-adj-sip-ping)# exit
Router(config-sbc-sbe-adj-sip)#
```

Redundant Peer Addresses

The SBC may be required to interoperate with peer SIP devices, such as the PGW 2200 softswitch, which can start signaling from a different IP address following a VLAN failure when operating with redundant hosts on separate VLANs (for example, geographically separated hosts). Peer SIP devices, such as the PGW 2200 softswitch, have the following high availability (HA) strategies that include both VLAN and host redundancy:

- In a standard failover scenario, when one host fails, a backup takes over. This backup also takes over the virtual IP address used for SIP communication. The call state is maintained, and the failover is made invisible to the SBC.
- In a scenario where a VLAN failure occurs, the PGW 2200 softswitch host, which interoperates with the SBC, starts using an interface in a different VLAN. Because you cannot transfer a virtual IP address between VLANs, the IP address for SIP communication changes.
- In some networks, the primary and backup hosts are geographically redundant and unable to share a VLAN. Therefore, you cannot transfer a virtual IP address between the hosts, so the IP address being used for SIP communication changes if the primary host fails.

Figure 6-7 illustrates a redundant topology.



Figure 6-7 Redundant Topology

When peer IP addresses change, the call state on the corresponding peer device continues to be maintained, and key SIP parameters, such as the dialog tags, the Contact header, and route set, are unchanged. However, the Contact header is incorrect, and does not contain the new peer IP address.

The Redundant Peer Addresses feature allows the IP address of its peer device to change, and supports the following functionalities on SBC:

- Accepts incoming SIP messages from any of the redundant IP addresses.
- Ignores discrepancies between the IP addresses specified in the SIP Contact header (or other SIP headers) and the actual IP address being used by the peer.
- Pings each peer address to monitor the active addresses and sends the outgoing SIP messages destined for the peer to an IP address that is currently active.
- Configures the SBC with multiple redundant IP addresses for a SIP peer device that is not contained within a single remote address mask.
- Supports the following modes of operation that is configurable for each adjacency:
 - The SBC switches peer IP address when a higher priority address becomes active, even if the current address does not fail.
 - Elects a *current destination* for each adjacency, choosing the peer IP address with the highest-priority that is currently active, and continues to use that destination until it is detected to have failed, at which point the election process is repeated.
- Uses a single local IP address, port, and VPN for all communication with every peer IP address.

Restrictions for Redundant Peer Addresses

The Redundant Peer Addresses feature has the following restrictions:

- This is a signaling-only feature.
- Alternative redundant addresses for a peer cannot be automatically detected, and must therefore, be configured using the **ping-enable** command.
- The main peer address in an adjacency share the same priority values, ranging from 1 to 6, as the redundant peer addresses.
- A single load-balancing method is provided. The SBC selects the active peer IP address with the highest configured priority for all the outgoing SIP requests.
- The source address of fast-REGISTER requests cannot be changed.
- If a SIP request is sent to a peer address, and no response is received, the SBC subsequently detects that the peer address has failed. However, the destination address of the SIP request does not change, and it is retried to the failed address. New requests are sent to an active address.
- The destination addresses and ports configured for a given adjacency are not available in message editing configuration. Therefore, there is no per-destination equivalent for the existing signaling-peer and signaling-peer-port header filtering syntax.
- The optimization to send only pings when required (ping suppression) cannot be configured on the adjacencies facing redundant peers.

Configuring Redundant Peer Addresses

Use the procedure described in this section to configure redundant peer addresses:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. no attach
- 6. force-signaling-peer all
- 7. ping-enable
- 8. exit
- 9. redundant peer index
- 10. address address
- **11. port** *port*
- 12. network {IPv4 address netmask | IPv6 address netmask}
- 13. priority priority
- 14. activate
- 15. exit
- **16.** signaling-peer-switch {always | fail}

- 17. signaling-peer-priority priority
- 18. exit
- 19. end
- 20. show sbc sbc-name sbe adjacencies adjacency-name peers

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	no attach	Detaches the adjacency from an account on the SBE.
	Example: Router(config-sbc-sbe-adj-sip)# no attach	Note The adjacency must be detached before adding or removing a redundant peer.
Step 6	force-signaling-peer all	Forces SIP messages for both in-call requests and out-of-call requests to go to the configured signaling peer.
	Example: Router(config-sbc-sbe-adj-sip)# force-signaling-peer all	
Step 7	ping-enable	Configures the adjacency to poll its remote peer by sending SIP OPTIONS pings to it, and enters the ping option submode
	Example: Router(config-sbc-sbe-adj-sip)# ping-enable	
Step 8	exit	Exits the adj-sip-ping mode, and moves to adj-sip mode.
	Example: Router(config-sbc-sbe-adj-sip-ping)# exit	

	Command or Action	Purpose
Step 9	redundant peer index	Enters the mode of an SBE SIP adjacency peer to configure an alternative signaling peer for the adjacency. You can specify the index number of the peer ranging from 1 to 5
	<pre>Example: Router(config-sbc-sbe-adj-sip)# redundant peer 1</pre>	specify the index number of the peer, ranging from 1 to 5.
Step 10	address address	Configures either an IP address or a host name to act as the redundant peer.
	<pre>Example: Router(config-sbc-sbe-adj-sip-peer)# no address</pre>	
Step 11	port port	Configures a port for the redundant peer.
	Example: Router(config-sbe-adj-sip-peer)# port 2	Note By default, 5060 port is used.
Step 12	<pre>network {IPv4 address netmask IPv6 address netmask}</pre>	Configures either an IPv4 or IPv6 network on the redundant peer.
	<pre>Example: Router(config-sbc-sbe-adj-sip-peer)# network ipv4 33.33.36.2 255.255.255.0</pre>	
Step 13	priority priority	Configures the redundant peer's priority. The range is from 1 to 6.
	<pre>Example: Router(config-sbc-sbe-adj-sip-peer)# priority 1</pre>	
Step 14t	activate	Activates the redundant signaling peer.
	Example: Router(config-sbc-sbe-adj-sip-peer)# activate	
Step 15t	exit	Exits the adj-sip-peer mode, and moves to adj-sip mode.
	Example: Router(config-sbc-sbe-adj-sip-peer)# exit	
Step 16	signaling-peer-switch {always fail}	Configure a SIP adjacency to switch the signaling peer to an available destination.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-switch always</pre>	
Step 17	<pre>signaling-peer-priority priority</pre>	Configures the priority of a signaling peer on a SIP adjacency. The range is from 1 to 6.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-priority 1</pre>	
Step 18	exit	Exits the adj-sip mode, and moves to sbe mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

	Command or Action	Purpose
Step 19	end	Exits the sbe mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 20	show sbc sbc-name sbe adjacencies adjacency-name peers	Lists the configured peers for the specified adjacency.
	Example: Router# show sbc mysbc sbe adjacencies sipGW peers	

Redundant Peer Addresses Example

The following example shows a redundant peer addresses configuration:

```
sbc mat
sbe
  adjacency sip SIPPA
   force-signaling-peer all
   signaling-peer-switch on-fail
   inherit profile preset-access
   signaling-address ipv4 1.0.0.10
   statistics method summary
   signaling-port 5068
    remote-address ipv4 1.0.0.0 255.0.0.0
    signaling-peer 1.0.0.3
   signaling-peer-priority 6
   signaling-peer-port 5068
   registration rewrite-register
   registration target address 1.0.0.3
    registration target port 5068
   redundant peer 1
    network ipv4 5.5.5.5 255.255.255.255
    address 5.5.5.5
    priority 2
    activate
    redundant peer 2
    network ipv4 22.22.22.22 255.255.255.255
    address 22.22.22.22
    port 2222
    priority 3
    ping-enable
    attach
```

Redundant Peer Addresses Verification

Use the following commands to verify the peers:

- show sbc sbe adjacencies detail—Displays detailed configuration of a SIP adjacency.
- show sbc sbe adjacencies peer—Lists the configured peers for an adjacency.
- show sbc sbe all-peers—Displays a peer's information.

Redundant Peer Addresses

Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model



Implementing Cisco Unified Border Element (SP Edition) Policies

A Cisco Unified Border Element (SP Edition) policy is a set of rules that define how the Cisco Unified Border Element (SP Edition) treats different kinds of voice over IP (VoIP) events. A Cisco Unified Border Element (SP Edition) policy allows you to control the VoIP signaling and media that passes through the Cisco Unified Border Element (SP Edition) at an application level.

Note

From Cisco IOS XE Release 2.4, configuration of policies is supported in the unified model. Enhancements to this feature have been introduced in later releases.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to *Cisco Unified Border Element* (SP Edition) Command Reference: Unified Model at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Implementing	Cisco Unified Border	Element (SP Edition) Policies
---	-----------------------------	--------------------------------------

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 2.5	Subscriber Policy support, Regular expression based routing support, SIP trunk-group ID routing support, and the SIP media line removal feature were added on the Cisco ASR 1000 Series Routers.
	Support for H.323 call routing features: H.323 Hunting and multiARQ hunting, Picking a next Hop in Routing Policy, Support for H.323 Addressing, DNS Name Resolution, Number Validation and Editing, Load Balancing, and Inter-VPN Calling were added on the Cisco ASR 1000 Series Routers.
Cisco IOS XE Release 2.6	(Source) Number Analysis feature updated to include source number table and source prefix table.
Cisco IOS XE Release 3.1S	Support for Asymmetric payload types was added.

Cisco IOS XE Release 3.2S	The Number Analysis feature was updated to include source address manipulation. The following number analysis tables were changed:
	• na-src-number-table was changed to na-src-address-table.
	• na-dst-number-table was changed to na-dst-address-table.
	• na-dst-number-attr-table was changed to na-carrier-id-table.
	• first-number-analysis-table was renamed as first-inbound-na-table
	Also, first-outbound-na-table was introduced, active-call-policy-set was renamed as call-policy-set default, and active-cac-policy-set was renamed as cac-policy-set global.
	Administrative domains were introduced.
	The copy and swap procedure for Call Admission Control (CAC) and call policy sets was introduced.
	The Multiple CAC Averaging Period feature was added.
	The Privacy Service feature was added.
	The Multiple SBC Media Bypass feature was added.
Cisco IOS XE Release 3.3S	Message, Policy, and Subscriber Statistics enhancements were added.
Cisco IOS XE Release 3.4S	The Limiting Resource Usage feature was added.
Cisco IOS XE Release 3.5S	CAC-related enhancements were introduced. The branch command has been introduced as an alternative to the caller and callee command pair in some configuration scenarios.
Cisco IOS XE Release 3.6S	The Common IP Address Media Bypass feature was added.

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- Message, Policy, and Subscriber Statistics, page 7-50
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- How to Implement Policies, page 7-65
- Configuring Asymmetric Payload Types, page 7-143
- Limiting Resource Usage, page 7-145
- Configuration Examples for Implementing Policies, page 7-156

Prerequisites for Implementing Policies

The following prerequisites are required to implement Cisco Unified Border Element (SP Edition) policies:

Before implementing policies, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions

The following restrictions apply when you implement routing policies on the Cisco Unified Border Element (SP Edition):

- H.323 protocols are not supported in Cisco IOS XE Release 2.4 and earlier.
- Regular expression matching is only supported for text user names and domain names in source or destination URIs for SIP calls. Regular expression matching for telephone numbers used in H.323 calls is not supported.
- SBC does not allow addition, modification, or removal of trunk-group ID (TGID) information before call routing occurs.
- SBC does not allow regular expression matching when performing TGID routing.

Information About Implementing Policies

A policy is a set of rules that define how the Cisco Unified Border Element (SP Edition) treats different kinds of VoIP events. A Cisco Unified Border Element (SP Edition) policy allows you to control the VoIP signaling and media that passes through Cisco Unified Border Element (SP Edition) at an application level. Figure 7-1 shows an overview of policy control flow.



Figure 7-1 Policy Control Overview

Number analysis and routing are configured in one type of configuration set, admission control is configured in another.

Number analysis (NA) determines whether a set of source digits or dialed digits represents a valid telephone number (based on number validation, number categorization, or digit manipulation). Call routing determines the VoIP signaling entity to which a signaling request should be sent. A destination adjacency is chosen for the signaling message based on various attributes of the message (for example, based on source account or adjacency). Routing policy is applied to new call events and to subscriber registration events.

In releases earlier than Cisco IOS XE Release 3.2S, textual usernames would bypass NA and proceed to route analysis, where they could be matched. From Cisco IOS XE Release 3.2S, NA can validate both dialed digits and textual usernames.

Also, in releases earlier than Cisco IOS XE Release 3.2S, dst-address in NA could be edited, but not src-address. From Cisco IOS XE Release 3.2S, src-address in NA can also be edited. The task of editing src-address can only be performed on digit strings, as in the case of editing dst-address.

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In Cisco IOS XE Release 3.2S, **na-src-name-anonymous-table** command was introduced to determine whether the source number's display name or presentation number is anonymous.

Call Admission Control (CAC) limits the number of concurrent calls and registrations, and restricts the media bandwidth dedicated to active calls. It allows for load control on other network elements by rate limiting. Certain events can be completely blocked (using a blacklist) or freely allowed (using a whitelist), based on certain attributes.

Not all policies are mandatory:

- To call between subscribers, only endpoint routing policy is required.
- To call between telephone numbers, only call routing policy is required.
- Number analysis and admission control are optional, although they are likely to be required by the user.

Policies refer to accounts and adjacencies by name. Therefore, you may find it useful to configure and name adjacencies before configuring policies although this is not required.

The following sections describe the many concepts critical to understanding how to implement Cisco Unified Border Element (SP Edition) policies:

- Cisco Unified Border Element (SP Edition) Policies
- Number Analysis Policies
- Routing
- H.323 Call Routing Features
- Call Admission Control

Cisco Unified Border Element (SP Edition) Policies

This section describes the following Cisco Unified Border Element (SP Edition) policies:

- Policy Events
- Policy Stages
- Policy Sets
- Policy Tables

Policy Events

Policies are applied to the following events:

- New calls—When new SIP or H.323 calls are signaled to the Cisco Unified Border Element (SP Edition), Cisco Unified Border Element (SP Edition) applies a policy to determine what happens to the new call request and what constraints the call must satisfy during its lifetime.
- Call updates—If one of the endpoints in a call attempts to renegotiate new media parameters, Cisco Unified Border Element (SP Edition) applies policy to ratify the attempt.
- Subscriber registrations—If a subscriber attempts to register through Cisco Unified Border Element (SP Edition), Cisco Unified Border Element (SP Edition) applies policy to determine what happens to the registration request.

Policy Stages

In the context of SIP and H.323 calls, three distinct stages of a policy are applied in a sequence to the policy events. The stages are:

- Inbound number analysis
- Routing
- Outbound number analysis
- Admission control

Some of these policy stages are skipped for particular types of events. Figure 7-2 shows the sequence of the policy stages for each event type.

Figure 7-2 Policy Stages for Event Types



If the policy stages fail, the call is rejected and the failure is propagated back to the calling device (using either session initiation protocol (SIP) or H.323 signaling, as appropriate) with the error codes in Table 7-2.

Component	Resulting SIP Error Code	Resulting H.323 Error
Number analysis	604 "Does not exist anywhere"	ITU-T Q.931 Release Complete UUIE with H.225 Reason field unreachableDestination
Routing	604 "Does not exist anywhere"	ITU-T Q.931 Release Complete UUIE with H.225 Reason field unreachableDestination
Call Admission Control	503 "Service Unavailable"	ITU-T Q.931 Release Complete UUIE with H.225 Reason field noPermission



If the call fails at the routing or Call Admission Control phase, it is released. There is no attempt to retry. Whether or not to retry is left to the upstream (calling) device to decide.

The following sections describe policy stages in more detail:

- Number Analysis
- Routing
- Admission Control

Number Analysis

Number Analysis (NA) determines whether a set of dialed digits or source number represents a valid telephone number. This is achieved by configuring one or more tables of valid source number and dialed digit strings using a limited-form regular-expression syntax, then matching the actual source number or dialed digits against the different strings in the tables.

NA policy is applied only to new call events. If NA determines that a new call does not contain a valid set of source numbers or dialed digits, Cisco Unified Border Element (SP Edition) rejects the call, using the error code described in the ?\$paranum>Policy Stages? section.

NA rules are sensitive to the source account and source adjacency of a call, which allows different dial plans to be configured for different customer organizations, or even for different endpoints.

In addition to validating a source number and dialed number, NA policy can also:

- Reformat the dialed digits into canonical form; for example, E.164 format.
- Label the call with a category, which is used by the later stages of policy.

Routing

Routing determines the next-hop VoIP signaling entity to which a signaling request should be sent. Routing of VoIP signaling messages occurs in two stages:

- Policy-based routing—The first stage of routing. In policy-based routing, a destination adjacency is chosen for the signaling message, based on various attributes of the message, discussed later.
- Protocol-based routing—Takes place after policy-based routing. Protocol-based routing uses a VoIP protocol-specific mechanism to deduce a next-hop IP address from the signaling peer configured for the destination adjacency chosen by policy-based routing.

For example, if the destination adjacency is a SIP adjacency and the signaling peer is uk.globalisp.com, Cisco Unified Border Element (SP Edition) uses domain name server (DNS) or IP lookup to determine the IP address and port of the SIP server for the domain uk.globalisp.com, and forwards the appropriate signaling message to that IP address and port.

Routing policy is applied to new call events and to subscriber registration events.

If a new call event matches an existing subscription, the call is routed automatically to the source IP address and port of the original subscriber registration. No configured policy is required to achieve this, and no configured policy can influence the routing of such calls.

Routing policy is not applied to call update events; call update signaling messages are routed automatically to the destination adjacency that was chosen for the new call event that originated the call.

It is possible that an event cannot be routed, if its attributes do not match a suitable configured routing rule. In such cases, Cisco Unified Border Element (SP Edition) rejects the event using a suitable error code.

Regular expression based routing feature allows the user to configure routing rules that use regular expressions to match the user name or domain part of a source or destination SIP URI.

SBC supports SIP trunk-group ID routing which provides call routing based on the value of the source or destination TGID parameters in the received SIP INVITE message.

Note

A trunk in a network is a communication path connecting two switching systems used in the establishment of an end-to-end connection. A trunk-group is a set of trunks, traffic engineered as a unit, for the establishment of connections within or between switching systems in which all of the paths are interchangeable. TGID is a string that identifies a trunk-group uniquely within a given context.

Admission Control

Call admission control determines whether an event should be granted or refused based on configured limits for network resource utilization. There are two reasons for performing admission control.

- To defend load-sensitive network elements, such as softswitches, against potentially harmful levels of load precipitated by singular events, such as DoS attacks, natural or man-made disasters, or mass-media phone-ins.
- To police the Service Level Agreements (SLAs) between organizations, to ensure that the levels of network utilization defined in the SLA are not exceeded.

Call admission control policy is applied to all event types. If an event is not granted by admission control policy, then Cisco Unified Border Element (SP Edition) rejects it with a suitable error code.

Policy Sets

A policy set is a group of policies that can be active on Cisco Unified Border Element (SP Edition) at any one time. If a policy set is active, then Cisco Unified Border Element (SP Edition) uses the rules defined within it to apply policy to events. You can create multiple policy sets on a single Cisco Unified Border Element (SP Edition).

A policy set has two potential uses:

- It enables you to atomically modify the configured policy by creating a copy of the currently active policy set, making all necessary changes, reviewing the modified policy, and then switching the active policy set. If a problem is discovered with the new policy set after it is activated, Cisco Unified Border Element (SP Edition) can be switched back to using the previous policy set with a single command.
- It enables you to create different policy sets for use at different times and to switch between them at the appropriate times.

Number analysis and routing are configured in a call policy set. Admission control is configured in a CAC policy set.

A new policy set can either be created empty (that is, without any configured policies), or created as a copy of another policy set. A policy set can be deleted, provided that it is not the active policy set.

When the Cisco Unified Border Element (SP Edition) is initialized, there are no active policy sets. At any time after initialization, the active policy set can be undefined. While there is no active routing policy, each event that requires routing is rejected.

From Cisco IOS XE Release 3.2S, the administrative domain allows a user to create separate groups of start indexes for number analysis, route analysis, and a CAC policy that can point to different policy sets. The administrative domain is then attached to the adjacencies for both incoming and outgoing analysis stages.

You can designate an inactive call policy set as the active call policy set at any time. However, you cannot directly modify an active call policy set. To modify an active call policy set, perform the copy-and-swap procedure.

You can designate an inactive CAC policy set as the active CAC policy set at any time. You can also modify an active CAC policy set by adding a new table in the CAC policy set. Note that you can create an entry in an existing table of an active CAC policy set only if the table type is **limit all** or **policy-set**. To perform a modification of this type, you must perform the copy-and-swap procedure.

You can define multiple policy sets that are active and select policy sets that can be used at each call analysis stage based on the adjacency setting. To modify a policy that may be referenced by multiple administrative domains, perform the copy-and-swap procedure.

Modifying Active CAC Policy Sets

The procedure to modify an active CAC policy set is the same as the procedure to create a CAC policy set. This procedure is described in the ?\$paranum>Configuring Call Admission Control Policy Sets, CAC Tables, and Global CAC Policy Sets? section on page 7-115. The difference lies in the checks the system performs at the end of each of these procedures. The newly modified CAC policy set is activated only after it is determined that the following conditions are met by all the CAC policy tables that are reachable from the modified CAC policy table. A failure message is displayed if any of the CAC policy tables do not meet any of these conditions.

- The table is active.
- All table lookup actions in the table point to valid tables.
- None of the table lookup actions result in a CAC configuration loop.
- All table entry values are valid. For example, the scope name or match prefix length must meet the specified criteria.

Note that the modified CAC policy set is applied only to new incoming calls. Calls that were in progress before the modified CAC policy set is made active are not affected when the modified CAC policy set is made active.

Copy-and-Swap Procedure

To perform a copy and swap procedure, specify the source policy to be copied, and the destination policy to which the source policy is to be copied. The source policy must be an existing policy set, but the destination policy must not be an existing policy set. To protect policies from being overwritten, an error is generated if an attempt is made to copy to an existing policy set.

The old policy can be referenced by different administrative domains, and have multiple indexes within one administrative domain. When the policies are swapped, all the references pertaining to the source policy are replaced with the destination policy. The swap function replaces the default policy and global policy sets, including any policy set referenced in an administrative domain.

The new policy should be set to complete using the **complete** command before all the references to the old policy are replaced.

We recommend that the new policy is exercised globally before all the references to the old policy are replaced.

An error is generated if the old policy either does not exist or is in an incomplete state.

The following configuration example describes the steps involved in copying and swapping call policy set 2:

```
Router# show run | b call-policy-set 2
call-policy-set 2
description this is call policy 1
```

```
first-call-routing-table TAB1
first-reg-routing-table TAB2
rtg-src-adjacency-table TAB1
entry 1
 match-adjacency SIPP1A
 dst-adjacency SIPP1B
 action complete
entrv 2
 match-adjacency SIPP1B
 dst-adjacency SIPP1A
 action complete
rtg-src-adjacency-table TAB2
entry 1
 match-adjacency SIPP1A
 dst-adjacency Registrar
 action complete
entrv 2
 match-adjacency SIPP1B
 dst-adjacency Registrar
  action complete
complete
```

Step 1 Copy the existing call-policy-set 2 to a new call-policy-set 20:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc MySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set copy source 2 destination 20
```

Step 2 Modify the new call-policy-set with the necessary changes:

Router(config-sbc-sbe-rtgpolicy)# first-inbound-na-table InTable
Router(config-sbc-sbe-rtgpolicy)# first-outbound-na-table OutTable

Step 3 Set the new call-policy-set 20 to complete:

Router(config-sbc-sbe-rtgpolicy)# complete
Router(config-sbc-sbe-rtgpolicy)# exit

Step 4 Swap the policies so that references to policy set 2 are replaced with policy set 20. The swap function replaces the default and global policy sets, including any policy set referenced in an administrative domain:

Router(config-sbc-sbe)# call-policy-set swap source 2 destination 20

The following configuration example describes the steps involved in copying and swapping an existing CAC policy set 12:

```
Router# show run | b cac-policy-set 12
cac-policy-set 12
first-cac-table 1
cac-table 1
table-type limit adjacency
entry 2
match-value SIPP1B
media police strip
action cac-complete
complete
```

Step 1 Copy the existing cac-policy-set 12 to a new cac-policy-set 22:

```
Router# configure terminal
```

Enter configuration commands, one per line. End with CNTL/Z. Router(config)# **sbc MySBC** Router(config-sbc)# **sbe** Router(config-sbc-sbe)# **cac-policy-set copy source 12 destination 22**

Step 2 Modify the new cac-policy-set with the necessary changes:

Router(config-sbc-sbe-cacpolicy)# cac-table TAB1
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# \$max-call-rate-per-scope 100

Step 3 Set the new cac-policy-set 22 to complete:

```
Router(config-sbc-sbe)# cac-policy-set 22
Router(config-sbc-sbe-cacpolicy)# complete
Router(config-sbc-sbe-cacpolicy)# exit
```

Step 4 Swap the policies so that references to policy set 12 are replaced with policy set 22:

Router(config-sbc-sbe)# cac-policy-set swap source 12 destination 22

Policy Tables

All policies on the SBE is configured in a set of tables. This section describes the overall structure of the policy tables, as described in the following sections:

- Nomenclature
- Application of Policy
- Policy Selection
- Policy Table Example

Nomenclature

This section defines some terms that we later use when discussing policy tables.

A policy table has the following properties:

- A name that uniquely identifies the table within the scope of a single policy set. Tables in different policy sets may have the same name.
- A type, which defines the criterion that is used to select an entry from the table.
- A collection of table entries.

A policy table entry is a member of a policy table. It has the following properties:

- A value to match on (the match value). The semantics of this value are determined by the table type. No two entries in the same table may have identical match values.
- An optional *action* to perform on the event, if it matches this entry.
- An optional name of the next table to search for policy, if the event matches this entry.

Application of Policy

The policy tables are searched whenever an event occurs. The policy to be applied to the event is built up as the tables are searched.

The policy sets contains the following properties, which define which policy tables are searched at each stage of the policy calculation. The call policy set contains:

• First NA policy table to process

- First routing policy table to process for calls
- First routing policy table to process for endpoint registrations

The CAC policy set contains the first admission control policy table.

When an event occurs, the policy tables are searched as follows. This procedure is followed once for every stage of policy to which an event is subjected.

- The first table for the particular stage of the policy calculation is obtained from the active configuration set.
- The type of the table defines which of the event's attributes (for example, the destination number or the source adjacency) is being examined by this table.
- This attribute is compared against the match value of every entry in the table. This results in either exactly one entry matching the event, or no entries matching the event.
- If an entry matches the event, then the action associated with that entry is performed. After the action is performed, if the entry contains the name of a next table, that table is processed. If there is no next table, then the policy calculation is complete and processing for this stage of policy ends.
- If no entry matches the event, then the policy calculation is complete and processing for this stage of policy ends.

Policy Selection

From Cisco IOS Release 3.2S, the SBC can have multiple active configuration sets. However, by using administrative domains, you can select different policy sets for inbound number analysis, routing, CAC, and outbound number analysis for messages based on their source and destination adjacencies. Figure 7-3 explains the call processing flow using the policy sets.

The policy set that is to be used for a given administrative domain is defined in the admin-domain mode. Call policy sets specified in the admin-domain mode is given a priority. The priority is required because more than one administrative domain can be specified on an adjacency. The SBC will use the policy-set with the highest priority.

The policy sets must be in a complete state before they are assigned to an administrative domain. A default call-policy-set must be configured before the administrative domain mode is entered. If an inbound NA set, a routing set, or an outbound NA set is undefined, the administrative domain uses the values defined within the default call-policy-set. For more information on administrative domains, see the Administrative Domains, page 7-59 section.

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Call Policy

A signaling event is assigned to the default call policy set if an admin-domain is not specified on the adjacency.

However, you can use different sets of incoming and outgoing number analysis tables based on the administrative domains configured for the incoming and outgoing adjacencies respectively. You can also configure a different routing policy set on a per-adjacency basis.

If more than one administrative domain is associated with the incoming adjacency, the SBC will use the policy set with the highest priority. You should not configure two routing policy sets with the same priority, two inbound NA policy sets with the same priority, or any two outbound NA policy sets with the same priority. The SBC logs an error but uses the policy with the highest index value.

If the adjacencies list any administrative domains that is not listed in the admin-domain mode, they use the priority in the global policy. The SBC logs a configuration warning if an adjacency references an undefined administrative domain.

CAC Policy

All events are limited by the applicable CAC policies indicated by the source and destination administrative domains and the global CAC policy.

The user can configure a CAC policy using different sets of tables based on the administrative domains configured on both the incoming and outgoing adjacencies. It is not required by the administrative domain to specify a CAC policy-set.

Policy Table Example

The following example illustrates the flow of control as policy tables are parsed at a particular stage of policy for a particular event. The event in this example is a new call, received from source account with destination number **129**. The stage of policy considered here is **routing**.

This example is provided for illustrative purposes only; routing tables are described in detail in the ?\$paranum>Routing? section.

Figure 7-4 shows the relevant routing tables.

Figure 7-4 Policy Table Example



sbc sbe ro match src-	uting-policy-ta account	able RtgAnalyzeSourceAccount				
match	action	next table				
csi	[none]	RtgAnalyzeDestCSINumber		sbc sbe ro match des	outing-policy-table R	tgAnalyzeDestCSINumber
				match	action	next table
pepsi		HtgAnalyzeDestPepsiNumber		1xx	dst-adjacency csi-chester	[none]
coca-cola	[none]	RtgAnalyzeDestCokeNumber		[2-5]xx	dst-adjacency csi-enfield	[none]
				61xx	dst-adjacency csi-alameda	[none]

The policy calculation begins by looking up the first policy table to be used by the routing stage. This is the table with name RtgAnalyzeSourceAccount. This table is processed as follows:

7xx

8xx

dst-adjacency

csi-edinburgh

csi-reston dst-adjacency [none]

[none]

- The table type of the table is src-account, so the source account of the new call event is compared with each of the entries in this table.
- The table entry that matches on csi provides a match for this new call event. There is no action associated with this entry, but the entry points to a next table with name RtgAnalyzeDestCSINumber.

The flow of control then passes to the table with name RtgAnalyzeDestCSINumber. This table is processed as follows:

- The cac-scope of the table is dst-number, so the destination number of the new call event is compared with each of the entries in this table.
- The table entry that matches on 1xx provides a match for this new call event. The action associated with this entry is performed; that is, the destination adjacency for the new call event is set to csi-chester.
- This entry does not point to a next table, so the policy calculation for the routing stage ends.

This example shows successful routing of the new call. The outcome is successful because the destination adjacency of the new call is selected before the policy calculation finishes. It is entirely possible for the outcome of routing to be unsuccessful for a new call if the routing policy tables do not

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assign a destination adjacency to the call before the routing policy calculation ends. For example, the routing policy illustrated above does not successfully route a new call whose source account is csi and whose destination number is 911.

In this example, a single entry is selected from each table that is traversed during the calculation. In general, at most one entry in any policy table matches an event to which policy is being applied. In cases in which more than one entry would match an event, the best matching entry is selected.

Number Analysis Policies

The following Number Analysis (NA) policies are configured within NA tables and are applied simultaneously to new calls and are described in the following sections:

- Number Validation
- Number Categorization
- Digit Manipulation
- Text Addresses
- Outbound Number Analysis

Number Validation

Number validation is fundamental to the process of traversing number analysis policy tables. A number is validated if the NA tables are traversed and the final entry examined contains an action of **accept**. A number is not valid if the NA tables are traversed, and the final entry examined contains an action of **reject**. A number also is not valid if, at any stage of processing the NA tables, a table with no matching entries is encountered.

Number analysis tables can be one of the following types:

- **dst-number**—Tables of this type contain entries whose match values represent complete numbers of Destination. In such tables, an entry matches an event if the entire dialed digit string exactly matches the match value of the entry.
- **dst-prefix**—Tables of this type contain entries whose match values represent number prefixes of Destination. In such tables, an entry matches an event if there exists a subset of the dialed digit string, consisting of consecutive digits taken from the front of the dialed digit string, that exactly matches the match value of the entry.
- **src-number**—Tables of this type contain entries whose match values represent complete numbers of Source. In such tables, an entry matches an event if the entire source digit string exactly matches the match value of the entry.
- **src-prefix**—Tables of this type contain entries whose match values represent number prefixes of Source. In such tables, an entry matches an event if there exists a subset of the source digit string, consisting of consecutive digits taken from the front of the source digit string, that exactly matches the match value of the entry.
- **src-account**—Tables of this type contain entries whose match values are the names of accounts. In such tables, an entry matches an event if the name of the source account of the event exactly matches the match value of the entry.
- **src-adjacency**—Tables of this type contain entries whose match values are the names of adjacencies. In such tables, an entry matches an event if the name of the source adjacency of the event exactly matches the match value of the entry.

• carrier-id—Tables of this type contain entries matching the carrier ID.

Digit Matching NA Tables

The format of the match values of entries in NA tables that match on the destination number or destination number prefix is a limited-form, regular expression string representing a string of dialed digits. The syntax used is described in Table 7-1.

Syntax Element	Description
X	Any numerical digit 0 – 9.
()	The digit within the parentheses is optional. For example, (0) XXXX represents 0XXXX and XXXX .
[]	One of the digits within the square brackets is used. For example, [01]XXX represents 0XXX and 1XXX . A range of values can be represented within the square brackets. For example, [013-5]XXX represents 0XXX , 1XXX , 3XXX , 4XXX and 5XXX .
*	The * key on the telephone.
#	The # key on the telephone.
-	Digit delimiter
,	Digit delimiter
a-f/A-F	Hexadecimal digits

Table 7-1 Syntax Used for Digit Matching NA Tables

In such tables, it is always possible that more than one entry in the table may match a particular digit string. For example, entries that match 1xx and 12x both match a digit string 129. However, a single entry must be chosen from each table, so the Cisco Unified Border Element (SP Edition) chooses the best matching entry by applying the following rules in the order given.

Step 1 Choose the longest explicit match.

If the NA table is a dst-prefix type, it is possible that more than one entry specifies an explicit number (that is, one that contains no X characters or [] constructs) and matches the dialed number of the event. In this situation, the entry with the longest number has priority.

For example, the dialed number begins 011, the number validation table is a dst-prefix type, and there are two matching entries with numbers 01 and 011. The entry with the number 011 takes priority, because it is a longer number.

Step 2 If there is no explicit match, choose the longest wildcard match.

If the table does not contain an explicit entry to match the dialed number of the event, the longest wildcard entry that matches takes priority.

Step 3 If there are multiple wildcard matches of the same length, choose the most explicit where possible.

For example, the dialed number is 01234567890, the NA table is a dst-number type, and there are two matching entries with match values 0123XXXXXX and 0123456XXXX. In the first entry, the fifth digit is a wildcard; in the second entry, the eighth digit is a wildcard, so the second entry takes priority.

If the same number is dialed, and a different NA table has matching entries [01]234XXXXXX and 0XXXXXXXXX, the second entry takes priority, because in the first entry the first digit is a wildcard.

Number Categorization

Events can be placed into user-defined categories during NA processing. This is achieved by specifying a categorization action in an entry of an NA table. Categories are useful, because they may be referred to later during the admission control policy stage.

At most, one category may be associated with an event. If, during processing of the NA tables, categories are assigned to an event multiple times, then the last category to be assigned is used. When a category is assigned to an event, it cannot be deleted, only replaced with another category.

Digit Manipulation

During number analysis (NA), it is often a requirement to normalize numbers—in other words, convert them from the internal format used by a particular organization or service provider to a canonical format understood globally in the Internet and PSTN.

This is achieved by specifying one or more of the following actions in an entry of an NA table:

- *del-prefix N*—This action removes the leading *n* digits from the dialed digit string, or deletes the entire string if it is *n* or fewer digits long.
- *del-suffix n*—This action removes the final *n* digits from the dialed digit string, or deletes the entire string if it is *n* or fewer digits long.
- add-prefix digit string—This action adds the given digit string to the front of the dialed digit string.
- replace digit string—This action replaces the entire dialed digit string with the given digit string.

Text Addresses

From Cisco IOS XE Release 3.2S, NA supports both textual username and digit matching. The table name na-src-number-table was changed to na-src-address-table, na-dst-number-table to na-dst-address-table, and na-dst-number-attr-table to na-carrier-id-table.

To match the text addresses, the existing match number is modified to read the match address. The **match-address** command can include a suffix of digits or regex.

In number analysis, you can define the following matching criteria types:

- Digit matching matches the dialed digit strings using specialized digit regex.
- Regex matching is applicable only to textual usernames, and offers a basic regular expression (BRE) syntax.

Note

Comparison of dialed digits and regex is possible. To compare a fixed string, a regex without any regex metacharacters should be used.

Outbound Number Analysis

Outbound Number Analysis allows the configuration of the source and destination numbers from the canonical form to a form that is appropriate for the destination administrative domains. The configuration of Outbound Number Analysis is similar to that of Inbound Number Analysis, which is converted from the source administrative domain form to the canonical form.

Outbound Number Analysis is performed automatically after successful routing. Outbound Number Analysis is processed using the **call-policy-set outbound-na** command in the destination administrative domain.

Routing

This section describes the following routing policies:

- Routing Tables and Adjacencies
- Number Manipulation
- Hunting
- Regular Expression-Based Routing

Routing Tables and Adjacencies

This section explains how routing tables are configured on the Cisco Unified Border Element (SP Edition).

The inputs to the policy-based routing stage are as follows:

- The destination number of the event, which is the post-NA dialed digit string (that is, it may have been modified from the original dialed digit string)—This input is present only if the event is a new call.
- The source number of the event—This input is present only if the event is a new call.
- The source adjacency of the event.
- The source account of the event.

The routing policy tables examine some or all of these inputs, and produce one of the following outputs:

- A single destination adjacency.
- A group of adjacencies used for load balancing. One of these is chosen, depending on the load previously sent to the adjacencies in this group.

Routing tables represent one of the following types:

• **dst-address**—Tables of this type contain entries matching the dialed number (after number analysis). These values are either complete numbers or number prefixes (depending on whether the *prefix* parameter is given). Without the *prefix* parameter, an entry matches an event if the dialed digit string exactly matches the match value of the entry. With the *prefix* parameter, an entry matches an event if there exists a subset of the dialed digit string, consisting of consecutive digits taken from the front of the dialed digit string that exactly matches the match value of the entry.

Routing actions also match text user name using a regular expression rather than a literal text string. Routing actions are considered to match if the regular expression matches at least one part of the address.

• **src-address**—Tables of this type contain entries matching the dialer's number or SIP user name. These values are either complete numbers or number prefixes (depending on whether the *prefix* parameter is given). Without the *prefix* parameter, an entry matches an event if the entire digit string representing the calling number exactly matches the match value of the entry. With the *prefix* parameter, an entry matches an event if there exists a subset of the digit string that represents the calling number, consisting of consecutive digits taken from the front of this string that exactly match the match value of the entry. Routing actions also match text user name using a regular expression rather than a literal text string. Routing actions are considered to match if the regular expression matches at least one part of the address.

- **src-account**—Tables of this type contain entries matching the names of accounts. In such tables, an entry matches an event if the name of the source account of the event exactly matches the match value of the entry.
- **src-adjacency**—Tables of this type contain entries matching the names of adjacencies. In such tables, an entry matches an event if the name of the source adjacency of the event exactly matches the match value of the entry.
- **src-domain**—Tables of this type contain entries matching the source domain names.

Routing actions also match domain names using full regular expressions rather than the limited range of regular expression matching. Routing actions are considered to match if the regular expression matches at least one part of the domain.

• **dst-domain**—Tables of this type contain entries matching the destination domain names.

Routing actions also match domain names using full regular expressions rather than the limited range of regular expression matching. Routing actions are considered to match if the regular expression matches at least one part of the domain.

- carrier-id—Tables of this type contain entries matching the carrier ID.
- **round-robin-table**—A group of adjacencies are chosen for an event if an entry in a routing table matches that event and points to a round-robin adjacency table in the next-table action. A round-robin adjacency table is a special type of policy table, whose events do not have any match-value parameters, nor next-table actions. Its actions are restricted to setting the destination adjacency and performing digit manipulation.
- **category**—Tables of this type contain entries matching on the category that was assigned to the call during number analysis. You assign the category during number analysis.
- **time**—Tables of this type contain entries matching on a user-configured time. The entries can have overlapping match periods. Time periods can be specified by year, month, date, day of the week, hour, or minute.
- **least-cost**—Tables of this type contain entries matching on the user-configured precedence (cost) of the entries. If more than one entry has an equal cost, an entry is selected based on a user-configured weight or an entry is selected based on the number of active calls on each route. If routing fails, then the adjacency with the next lowest cost is selected.
- **src-trunk-group-id**—Tables of this type contain entries matching the source TGID or TGID context parameters and action type to perform the call routing.
- **dst-trunk-group-id**—Tables of this type contain entries matching the destination TGID or TGID context parameters and action type to perform the call routing.

The rules specified in the ?\$paranum>Digit Matching NA Tables? section on page 7-15 govern the format and matching rules of the match-values of the entries in routing tables of type dst-number, dst-prefix, src-number and src-prefix.

Number Manipulation

The number manipulation feature enables you to specify various number manipulations that can be performed on a dialed number after a destination adjacency has been selected. Number manipulation can be configured as a routing policy.

This enhancement affects the billing functionality as it allows the Cisco Unified Border Element (SP Edition) to display both the original and the edited dialed number for a call. For example:

```
<party ty"e="0"ig" pho"e="01234567890"/>
<party ty"e="t"rm" pho"e="23456789"31" editphone="111111111111"/>
```



The phone numbers in the above example are not real.

The number manipulation feature requires that the edit action be allowed in the routing policy entries. The edit action takes the same parameters as the edit action for the number analysis tables, enabling you to delete a number of characters from the beginning or end of the dialed string, add digits to the start of the string, or replace the entire string with another. For example, if the following table were matched:

```
call-policy-set 1
  rtg-src-adjacency-table table1
   entry 1
    match SipAdj1
   edit del-prefix 3
   dst-adjacency SipAdj2
   action complete
   end
  end
```

then the dialed string would have the first three of its digits deleted.

In the number analysis stage you can specify categories as shown below.

```
call-policy-set 1
    first-inbound-na-table check-accounts
   na-src-account-table check accounts
    entry 1
     match-account hotel_foo
     action next-table hotel_dialing_plan
     entry 2
     match-account hotel_bar
     action next-table hotel_dialing_plan
     entrv 3
     match-account internal
     action accept
    na-dst-prefix-table hotel_dialing_plan
     entry 1
      match-prefix XXX
      category internal
      action accept
     entry 2
     match-prefix 9XXX
      category external
      action accept
```

Later during routing, the calls are routed based on assigned categories.

```
call-policy-set 1
first-call-routing-table start_routing
rtg-category-table start_routing
entry 1
match-category internal
action next-table internal_routing
entry 2
match-category external
action next-table external_routing
rtg-src-adjacency-table internal_routing
entry 1
match-adjacency sip_from_foo
```

Γ

```
dst-adjacency sip_to_foo
  action complete
 entrv 2
 match-adjacency sip_from_bar
 dst-adjacency sip_to_bar
 action complete
rt-dst-address-table external_routing
entry 1
 match-address 208111
 prefix
 dst-adjacency sip_to_foo
 action complete
entry 2
 match-address 208222
 prefix
 dst-adjacency sip_to_bar
 action complete
entrv 3
 match-address 208333
 prefix
 dst-adjacency sip_to_softswitch
 action complete
```

```
<u>Note</u>
```

The category of a call cannot be changed in a routing table. Categories are only assigned during number analysis.

You can also specify various number manipulations to be performed on a dialing or dialed number after a destination adjacency is selected.

The following example adds a prefix of "123" to the source number, for all calls coming in on "SipAdj1" adjacency and destined to "SipAdj2".

```
call-policy-set 1
rtg-src-adjacency-table table1
entry 1
match SipAdj1
edit-src add-prefix 123
dst-adjacency SipAdj2
action complete
```

Hunting

Cisco Unified Border Element (SP Edition) can hunt for other routes or destination adjacencies in case of a failure. Hunting means the route is retried. Cisco Unified Border Element (SP Edition) supports hunting of SIP and H.323 calls. Hunting can be configured as a routing policy.

There are several ways in which failures can occur, including the following:

• CAC policy refusing to admit a call

If a CAC policy rejects a call, the SBC automatically attempts to reroute the call using the Routing Policy Service (RPS). RPS decides where to route onward signaling requests by using the configured policy in the RPS. The call is then tested against CAC policy again.

- Routing Policy Services being unable to route a call
- Call setup failure being received from SIP or H.323.

When the SBC receives a call setup failure notification from H.323 or SIP, it is notified whether or not it should attempt to reroute the call, depending upon the error code.

If an SIP or H.323 adjacency attempts to route a call, and the attempt fails, it receives an error code. You can configure which error codes trigger hunting or rerouting.

- If the error code received by the adjacency matches an entry on this list, RPS is signalled to reroute the call. Rerouting then occurs unless the number of attempts exceeds the limit set as the maximum number of routing attempts that SBC makes. The default is three attempts.
- If the error code received by the adjacency does not match an entry on this list, RPS is signalled not to reroute the call.

For both SIP and H.323 call, you can configure a list of error codes or failure return codes to trigger hunting or rerouting for a particular adjacency by using the **sip hunting-trigger error-codes** or **hunting-trigger error-codes** commands.

You can also configure a list of H.323 error codes at a global level, by using the **hunting-trigger** command in the global H.323 configuration mode.

SIP error codes are numeric error codes. H.323 error codes are textual. See the ?\$paratext[TC_TableCap,TCW_TableCapW,TCPr_TableCapPref,TCWPr_TableCapWPref,TCF_TableCapPartFirst,TCN_TableCapPartNext,TCWF_TableCapWPartFirst,TCWN_TableCapWPartNext]>? table.

Hunting finishes when one of the following conditions is met:

- The call is successfully routed.
- The SBC receives a call setup failure notification with the instruction not to continue hunting, in which case the call fails.
- The SBC has made the number of specified routing attempts and the call has not been successfully routed, in which case the call fails.
- The SBC has tried all available adjacencies, and the call has not been successfully routed, in which case the call fails.

H.323 hunting has the additional hunting modes of alternate endpoints and multiARQ hunting. See the ?\$paranum>H.323 Call Routing Features? section on page 7-24.

For information on configuring SIP and H.323 hunting, see the ?\$paranum>Configuring Hunting? section on page 7-107.

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Table 7-2 lists the supported error codes that you can configure to trigger hunting of SIP or H.323 calls.

 Table 7-2
 Configurable Error Codes to Trigger Hunting

Supported SIP Error Codes	Supported H.323 Error Codes
400 - Bad Request	unreachableDestination
401 - Unauthorized	noPermission
402 - Payment Required	noBandwidth
403 - Forbidden	destinationRejection
404 - Not Found	gatewayResources
405 - Method Not Allowed	badFormatAddress
406 - Not Acceptable	securityDenied
407 - Proxy Authentication Required	the internally-defined value "connectFailed"
408 - Request Timeout	
409 - Conflict	
410 - Gone	
411 - Length Required	
413 - Request Entity Too Large	
414 - Request URI Too Long	
415 - Unsupported Media Type	
416 - Unsupported URI Scheme	
420 - Bad Extension	
421 - Extension Required	
423 - Interval Too Brief	
480 - Temporarily Unavailable	
481 - Call/Transaction Does Not Exist	
482 - Loop Detected	
483 - Too Many Hops	
484 - Address Incomplete	
485 - Ambiguous	
486 - Busy Here	
487 - Request Terminated	
488 - Not Acceptable Here	
491 - Request Pending	
493 - Undecipherable	
500 - Server Internal Error	
501 - Not Implemented	
502 - Bad Gateway	
503 - Service Unavailable	
504 - Server Time-Out	_

Supported SIP Error Codes	Supported H.323 Error Codes
505 - Version Not Supported	—
513 - Message Too Large	—
600 - Busy Everywhere	_
603 - Declined	_
604 - Does Not Exist Anywhere	_
605 - Not Acceptable	—

Table 7-2 Configurable Error Codes to Trigger Hunting (continued)

Regular Expression-Based Routing

Regular expression based routing allows the user to configure routing rules that use regular expressions to match the user name or domain part of a source or destination SIP URI.

Routing actions match text user name using a regular expression rather than a literal text string when "regex" keyword is used. Routing actions are considered to match if the regular expression matches at least one part of the address.

Table 7-3 shows the basic regular expression (BRE) implementation for the supported regex characters.

Metacharacter	Description
	Matches any single character. Within POSIX bracket expressions, the dot character matches a literal dot. For example, a.c matches "abc", etc., but [a.c] matches only "a", ".", or "c".
[]	A bracket expression. Matches a single character that is contained within the brackets. For example, [abc] matches "a", "b", or "c". [a-z] specifies a range which matches any lowercase letter from "a" to "z". The - character is treated as a literal character if it is the last or the first character within the brackets, or if it is escaped with a backslash: [abc-], [-abc], or [a\-bc].
[^]	Matches a single character that is not contained within the brackets. For example, [^abc] matches any character other than "a", "b", or "c". [^a-z] matches any single character that is not a lowercase letter from "a" to "z". As above, literal characters and ranges can be mixed.
٨	Matches the starting position of the string.
\$	Matches the ending position of the string.
\(\)	Defines a marked subexpression. The string matched within the parentheses can be recalled later (see the next entry, \n).
\n	Matches what the nth marked subexpression matched, where n is a digit from 1 to 9. This construct is theoretically irregular and was not adopted in the POSIX ERE syntax. Some tools allow referencing more than nine capturing groups.
*	Matches the preceding element zero or more times.
$\{m,n\}$	Matches the preceding element at least m and not more than n times. For example, a\{3,5\} matches only "aaa", "aaaa", and "aaaaa".

Table 7-3 BRE Implementation

The rtg-src-address and rtg-dst-address tables contain entries matching the dialed number (after number analysis). At run-time, when the Request-URI is processed, the username is parsed to determine if the username is considered to be "textual" or "dialed-digits". It is initially assumed that the username is a dialed-digit string, and the username will considered to be textual only if non-dialed digit characters are encountered. Having determined this type, only policy entries matching this type are evaluated.

When configuring policy entries which match on rtg-src-address or rtg-dst-address table, it is important to configure the match-address correctly to ensure the policy entry is evaluated. In order to assist in configuration, the type of match address will be assessed and configured automatically if not specifically configured.

You can configure one of the following three choices explicitly:

match-address address [digits] (limited digit string regex)

match-address address [string] (string (textual) comparison on textual username only)

match-address address [regex] (regular expression on string (textual) usernames only)

Example:

Valid entries:

```
match-address (0)1234[56] digits
match-address username string
match-address [Uu]sername regex
```

Invalid entries:

```
match-address 1234 string (cannot perform a string match on dialed digits)
match-address 1234 regex (cannot perform a regex match on dialed-digits)
match-address [abc] regex (abc are valid dialed digits and #, * and d are also valid
dialed digits)
```

In this case the entry is evaluated at configuration time and error responses generated if there is a perceived mismatch in the type and match-address.

H.323 Call Routing Features

In addition to the features described in the ?\$paranum>Routing? section on page 7-17 that also apply to H.323 calls, Cisco Unified Border Element (SP Edition) supports various H.323-specific call routing features.

The H.323 call routing features are:

- H.323 Hunting, page 7-25
- Picking a Next Hop in Routing Policy, page 7-26
- Support for H.323 addressing, page 7-26
- DNS Name Resolution, page 7-26
- Number Validation and Editing, page 7-26
- Load Balancing, page 7-27
- Inter-VPN Calling, page 7-27

H.323 Hunting

Cisco Unified Border Element (SP Edition) supports hunting of H.323 calls. Cisco Unified Border Element (SP Edition) hunts for other routes or destination adjacencies in the event of a failure. Hunting re-routes the call in response to a specific user-configured event or error code.

H.323 hunting or re-routing operates in the following ways based on whether the adjacency is a gatekeeper or non-gatekeeper adjacency:

• For a gatekeeper adjacency, the SBC can cycle through a list of potential signaling next hops based on input from the gatekeeper. Alternate Endpoints and MultiARQ are two methods that allow the gatekeeper to provide the SBC with this list.

If H.323 has a list of alternate endpoints for a call, H.323 tries each of these in turn before reporting a routing failure to the RPS.

MultiARQ is described in the ?\$paranum>MultiARQ Hunting? section.

• For a non-gatekeeper adjacency, or where all the next hops on a gatekeeper adjacency have been exhausted, the SBC can re-route the call to a different adjacency in the "hunt group" (specifically, the round-robin-table or least-cost routing table). For more information on routing tables, see the ?\$paranum>Routing Tables and Adjacencies? section on page 7-17.

MultiARQ Hunting

Cisco Unified Border Element (SP Edition) supports a non-standard H.323 mechanism for hunting for other routes or destination adjacencies. This is based on issuing multiple Admission Requests (ARQs) to a Gatekeeper for a single call.

The SBC sends an ARQ (Admission Request) when an incoming call is received on a gatekeeper adjacency, or an outgoing call needs to be made on a gatekeeper adjacency. For an outgoing call, the gatekeeper returns the signaling address of the endpoint that the SBC should contact.

MultiARQ hunting occurs under the following circumstances:

- The H.323 endpoint sends an ARQ to a Gatekeeper as part of establishing an outbound call leg.
- The Gatekeeper contacts other network entities and identifies one or more potential endpoints.
- The Gatekeeper returns an admissionConfirm (ACF) containing a single destinationInfo and no alternateEndpoints.
- The H.323 endpoint attempts to contact the endpoint identified in the ACF. The endpoint either rejects the call or is unreachable.

The MultiARQ hunting continues until one of the following conditions is met.

- An endpoint is contacted and the call completes.
- A Gatekeeper ARQ retry is required, but the hard-coded limit on the number of permitted retry ARQs has been reached. This number is a customizable constant in h323cust.h, and is currently set to 32.
- The Gatekeeper returns an admissionReject, indicating that there are no further suitable endpoint identifiers.
- An endpoint returns a rejectReason which is not configured as a hunting trigger.
- An endpoint cannot be contacted, and connectFailed is not configured as a hunting trigger.

For information on configuring MultiARQ Hunting, see the ?\$paranum>Configuring H.323 MultiARQ Hunting? section on page 7-112.

Picking a Next Hop in Routing Policy

When receiving an incoming H.323 call, Cisco Unified Border Element (SP Edition) carries out routing to determine the next hop for the call.

SBC policy allows calls to be routed to one of the following:

- signaling peer (such as a gateway)
- outgoing gatekeeper

When a gatekeeper is used, the gatekeeper is responsible for resolving the called party number to a next hop address.

In a SBC configuration, a routing next hop is identified by an adjacency name. The adjacency is configured with the address of the next hop gateway or gatekeeper.

Support for H.323 addressing

All H.323 calls through Cisco Unified Border Element (SP Edition) need to specify a called party number. A called party number may optionally be supplied in the Q.931 calledPartyNumber or the H.225 destinationAddress, with the former taking priority. If a called party number is not present in either of these fields, then the SBC rejects the call.

Finally, the connected number may also optionally be supplied in the Q.931 connectedNumber or the H.225 connectedAddress, with the former taking priority. The connected number indicates the party the call ends up connecting with because during call setup, the call might be redirected or the called number might be edited along the way.

When an H.323 endpoint sends out a Q.931/H.225 message, the called and calling numbers are always placed in the Q.931 fields, not the H.225 fields.

DNS Name Resolution

Domain name server (DNS) name resolution enables you to use the domain name instead of the IP address in an adjacency configuration. You can configure both gatekeeper and non-gatekeeper adjacencies with DNS names.

If you use a DNS name in an adjacency configuration, the name is resolved each time a call is routed out over that adjacency. This process allows DNS-based load-balancing.

Number Validation and Editing

Cisco Unified Border Element (SP Edition) allows validation, editing and categorization of the called and calling party number through a Number Validation configuration.

This can be used for comparing or editing of source or destination telephone numbers or textual usernames. This process is called Number Analysis (NA). Number Analysis (NA) determines whether a set of source or destination digits, or source or destination textual addresses represents a valid address (based on number validation, number categorization, and/or digit manipulation). This is achieved by configuring one or more tables of valid addresses and editing rules in the tables. Matching for digit strings uses a limited-form of specialized regular-expression syntax and matching for textual addresses is done on the basis of the Basic Regular Expression syntax. In both cases, either the entire address or part of the address can be matched.

NA can be optionally configured as a step within the call policy set.

For more information, see the ?\$paranum>Number Analysis Policies? section on page 7-14 section and the ?\$paranum>Number Analysis? section on page 7-6 in the Implementing Cisco Unified Border Element (SP Edition) Policies chapter.

Load Balancing

Cisco Unified Border Element (SP Edition) can load balance between H.323 adjacencies using Round Robin or Least Cost Routing configurations.

Round Robin load balancing distributes calls evenly between adjacencies. Least Cost load balancing assigns a priority to each adjacency.

For example, routing might route two consecutive calls onto two different adjacencies.

- For gatekeeper adjacencies, the calls will be admitted on two different gatekeepers. It is up to the gatekeeper routing configuration to determine whether the signaling next hop for each call is the same.
- For non-gatekeeper adjacencies, the signaling next hop will be set to two different gateways (or terminals).

If a gatekeeper adjacency loses contact with the gatekeeper, it is temporarily taken out of service - meaning that the SBC will not attempt to route new calls through it. If there is an alternative route, call setup will continue on the alternative route. You can also manually deactivate an adjacency, which has the same effect.

Inter-VPN Calling

Cisco Unified Border Element (SP Edition) can peer with H.323 devices in different VPNs simultaneously.

You configure VPNs on a per-adjacency basis. Therefore, inter-VPN calling is simply a matter of your configuring a routing policy that routes calls between adjacencies in different VPNs.

Call Admission Control

This section describes the following:

- Call Admission Control Overview, page 7-28
- Compound Scopes, page 7-28
- Policy Scopes, page 7-29
- Policy Set Tables and Limit Tables, page 7-32
- Limit Tables, page 7-32
- CAC Table Entry Configuration Commands, page 7-33
- Media Line Removal, page 7-38
- Multiple SBC Media Bypass, page 7-39
- Common IP Address Media Bypass, page 7-43
- CAC Rate Limiting, page 7-45
- Multiple CAC Averaging Periods, page 7-46
- Subscriber Policy, page 7-46

• Privacy Service, page 7-47

Call Admission Control Overview

Call Admission Control (CAC) allows you to configure policy for accepting or rejecting calls. It allows you to apply detailed policies to certain call options to limit the number of concurrent calls and registrations. CAC can restrict the media bandwidth dedicated to active calls. It allows for load control on other network elements by rate limiting. Certain events can be completely blocked (using a blacklist) or freely allowed (using a whitelist), based on certain attributes.

CAC determines whether an event should be granted or refused based on configured limits for network resource utilization. There are two reasons for performing call admission control.

- To defend load-sensitive network elements, such as softswitches, against potentially harmful levels of load precipitated by singular events, such as DoS attacks, natural or man-made disasters, or mass-media phone-ins.
- To police the Service Level Agreements (SLAs) between organizations, to ensure that the levels of network utilization defined in the SLA are not exceeded.

Call admission control is the final stage of the call policy, so it is applied after number analysis and routing policy. CAC policy is applied to all event types, such as new calls, subscriber registrations, and call updates. If an event is not granted by the CAC policy, then Cisco Unified Border Element (SP Edition) rejects it with a suitable error code.

A CAC policy consists of the following.

• A limit or limits that must not be exceeded.

Limits, for example, can be set on the maximum number of concurrent calls, the maximum rate of calls, or the maximum bandwidth consumed by calls.

• A scope at which the limits are applied.

This can be global, per-account, per-adjacency, or any of the scopes defined in Policy scopes. Combinations of scopes can also be used, such as "per account, per number category." Scope is part of the policy itself. For example, in the policy "maximum 20Kb per call," the scope is "per call."

To define an admission control policy, you must define the limit and the scope at which it is applied. For example, you can define a policy such that not more than 10 concurrent calls (limit) could ever be made from a single account (scope).

Although the scope and limits define the policy, they do not determine when the policy is applied. For example, you cannot name a particular account, such as "account1," as the scope for your policy. Instead, the table-type and match value are used to determine when a policy is applied. Setting "account" as the table-type and "account1" as the match value matches call events from account1.

Compound Scopes

Compound scopes provide a more elaborate set of options for configuring policy. Certain policy scopes can be combined to create compound scopes. To combine scopes, configure each scope using a separate **first-cac-scope** or **cac-scope** command.

The following are examples of compound scopes:

• If you want to restrict the number of calls between any pair of adjacencies to 20, you could create a policy with MaxCalls = 20 and a scope of "src_adjacency, dst_adjacency." This policy would restrict the number of calls between any pair of adjacencies to 20. However, it would not limit the total number of calls out of any adjacency, nor the total number of calls into any adjacency.
• You can define an admission control policy at a compound scope of "source adjacency and category," and set the maximum concurrent calls in this scope to 10. This policy would restrict the number of concurrent calls of the same category that each adjacency could make to 10. The scope field value is src-adjacency, category.

Policy Scopes

Table 7-4 defines the scopes in which call admission policies can be applied and specifies whether each of these scopes can be combined with other scopes.

Scope Option or Value of Scope Field	Scope	Description	Can Scope Be Combined?
account	Per account	The limits specified in this scope apply to all the events from the same account.	Yes, except the dst-account and src-account scopes
adjacency	Per adjacency	The limits specified in this scope apply to all the events from the same adjacency.	Yes, except the src-adjacency, dst-adjacency, src-adj-group, and dst-adj-group scopes
adj-group	Per adjacency group	The limits specified in this scope apply to all events sent to or received from the same adjacency group. For example, you can restrict the total number of concurrent calls that can be sent to or received from the adjacencies in a single adjacency group by configuring limits in this scope.	Yes, except the adjacency, src-adj-group, and dst-adj-group scopes
call	Per call	The limits specified in this scope apply to any single call. For example, you can restrict the per-call bandwidth or the allowed call update rate by configuring limits in this scope. Note that some limits are invalid in this scope.	No
category	Per category	The limits specified in this scope apply to all events that have been placed in the same category by the number analysis policy tables. For example, you can restrict the total number of concurrent calls in any single category by configuring limits in this scope.	Yes
dst-account	Per destination account	The limits specified in this scope apply to all events sent to the same account. For example, you can restrict the total number of concurrent calls that can be sent to any single account by configuring limits in this scope.	Yes, except the account scope

Table 7-4 Policy Scope Definitions

Scope Option or Value of Scope Field	Scope	Description	Can Scope Be Combined?
dst-adj-group	Per destination adjacency group	The limits specified in this scope apply to all events sent to the same adjacency group. For example, you can restrict the total number of concurrent calls that can be sent to the adjacencies in a single adjacency group by configuring limits in this scope.	Yes, except the adj-group scope
dst-adjacency	Per destination adjacency	The limits specified in this scope apply to all events sent to the same adjacency. For example, you can restrict the total number of concurrent calls that can be sent to any single adjacency by configuring limits in this scope.	Yes, except the adjacency scope
dst-number	Per dialed number	The limits specified in this scope apply to all events that have the same destination number. For example, you can restrict the total number of concurrent calls to any single valid number by configuring limits in this scope.	Yes
global	Global	The limits specified in this scope apply to SBC as a whole.	No
src-account	Per source account	The limits specified in this scope apply to all events received from the same account. For example, you can restrict the total number of concurrent calls that can be initiated from any single account by configuring limits in this scope.	Yes, except the account scope
src-adj-group	Per source adjacency group	The limits specified in this scope apply to all events received from the same adjacency group. For example, you can restrict the total number of concurrent calls that can be initiated from the adjacencies in a single adjacency group by configuring limits in this scope.	Yes, except the adjacency and adj-group scopes
src-adjacency	Per source adjacency	The limits specified in this scope apply to all events received from the same adjacency. For example, you can restrict the total number of concurrent calls that can be initiated from any single adjacency by configuring limits in this scope.	Yes, except the adjacency scope
src-number	Per dialing number	The limits specified in this scope apply to all events that have the same source number. For example, you can restrict the total number of concurrent calls from every single source number by configuring limits in this scope.	Yes

Scope Option or Value of Scope Field	Scope	Description	Can Scope Be Combined?
sub-category	Per subscriber category	Note This is not supported in Cisco IOS XE Release 2.4. The limits specified in this scope apply to all events sent to or received from members of the same subscriber category. For example, you can restrict the total number of concurrent calls that can be sent to or received from the subscribers in a single subscriber category by configuring limits in this scope.	Yes, except the sub-category-pfx and subscriber scopes
sub-category-pfx	Per subscriber category prefix	Note This is not supported in Cisco IOS XE Release 2.4. The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix. For example, you can restrict the total number of concurrent calls that can be sent to or received from the subscribers in a single subscriber category prefix by configuring limits in this scope.	Yes, except the sub-category-pfx and subscriber scopes
subscriber	Per subscriber	NoteThis is not supported in Cisco IOS XE Release 2.4.The limits specified in this scope apply to all events sent to or received from individual subscribers. A subscriber is any device in the network that has registered with a Registrar server via SBC, or with an S-CSCF in an IP Multimedia Subsystem (IMS) network.This does not allow you to match on a specific subscriber.	Yes, except the sub-category-pfx and subscriber scopes

Table 7-4	Policy Scope	Definitions	(continued)
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If you are supporting Aggregate Registrations in a non-IMS network, all of the phones behind a device (such as a PBX) are counted as the same subscriber if you are using a per-subscriber scope.

Non-Subscriber Group

When a subscriber scope is enabled, the SBC includes an additional group of ALL "non-subscribers." The non-subscribers are counted within a special group of the subscriber scope. The non-subscriber group is matched if the call is from a non-subscriber. Limits set in the subscriber scope apply to this non-subscriber group.



A "subscriber" is identified using the Address-of-Record that is registered with the registrar. A "subscriber category" is based on the source IP address of the SIP message. When some subscribers sit behind a Network Address Translation (NAT) device and share the same IP address, they are in the same subscriber category. However, they differ among each other by their AOR.

Policy Set Tables and Limit Tables

Call admission control policies are configured using a combination of Policy Set and Limit tables.

A Policy Set table type is applied to all entries defined within the CAC table. Each entry within the table configures its own scope. Every entry in a Policy Set table automatically matches every event that reaches that table. Policy Set tables create multiple policies for each event.

A Limit table type selects the single best matching match value defined in a CAC entry. The scope for the limit table type is inherited from the limit table's parent table. The entries in a Limit table specify the values to match against and the limits to apply if a match is achieved.

The major difference between a Policy Set table and a Limit table is that the Policy Set table creates multiple policies for a given event, while a Limit table only defines one policy for a given event.

For information on table-types, match values, and when an event matches an entry for Limit Table, see Table 7-5. For information on scope name, scope definition, and whether a scope can be combined, see Table 7-4.

Limit Tables

Table 7-5 lists the types of Limit tables. For each table type, the corresponding Match value is listed, with the conditions under which a match is achieved. If a match is achieved, the corresponding policy is applied to the event.

Table Type	Match Value	Conditions Where an Event Matches an Entry
account	account name	Match value is the source and/or destination account name.
adj-group	adjacency group name	Match value is the source and/or destination adjacency group name.
adjacency	adjacency name	Match value is the source and/or destination adjacency name.
all	NA	All events match entry
call-priority	SBC priority	SBC priority is the event call-priority.
category	category name (assigned during number analysis)	Event has been assigned a category, and match value is the name of the category assigned.
dst-account	account name	Match value is the destination account name.
dst-adj-group	adjacency group name	Match value is the destination adjacency group name.
dst-adjacency	adjacency name	Match value is the destination adjacency name.

Table 7-5 Table Types for Limit Table

Table Type	Match Value	Conditions Where an Event Matches an Entry
dst-prefix	number prefix	Match value is the first digits of the number being called.
event-type	Type of event to which CAC policy is applied (new-call, call-update or endpoint-reg)	Match value is the event type.
src-account	account name	Match value is the source account name.
src-adj-group	adjacency group name	Match value is the source adjacency group name.
src-adjacency	adjacency name	Match value is the source adjacency name.
src-prefix	number prefix	Match value is the first digits of the calling number
sub-category	ipv4 { <i>ip-address</i> } [vrf <i>vrf</i>]	Match value is the IPv4 address.
		When the "sub-category" table type is defined for a CAC table, you must define the match-value within the entry. As an example, you would use the command: match-value ipv4 { <i>ip-address</i> } [vrf <i>vrf</i>]
sub-category-pfx	<pre>ipv4 {ip-address} {prefix-len}</pre>	Match value is the IPv4 address.
pfx-len	[vrf vrf]	When the "sub-category-pfx <i>pfx-len</i> " table type is defined for a CAC table, you must define the match-value and match-prefix-len within the entry. As an example, you would use the command: match-value ipv4 { <i>ip-address</i> } { <i>prefix-len</i> } [vrf vrf].

Table 7-5	Table Typ	es for Limit	Table ((continued)

CAC Table Entry Configuration Commands

Each CAC table consists of a collection of table entries, defined within the CAC table submode. For Policy Set table types, the CAC scope is defined within each entry. If unspecified, the scope defaults to global for that entry.

For Limit table types, the CAC entry specifies a value to match against. The semantics of this match-value are determined by the type of Limit table.

For both table types, the limits defined within the entry are calculated using per scope values. Some limits are not applicable at all scopes. Policy Set table types define the scope within the entry, thus both the limit and the scope are per entry. If you want per entry limits for a Limit table type, then configure the Limit table type to match the scope.

See the ?\$paranum>Configuring Call Admission Control Policy Sets, CAC Tables, and Global CAC Policy Sets? section on page 7-115 for detailed configuration step information.

Table 7-6 shows a list of various limits and options that can be configured on an entry in a CAC policy-set table. These configurable command options can be displayed with the following commands:

```
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-table 4
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# ?
```

<u>Note</u>

The cac-scope command option is only displayed for Policy Set table types. The match-value command option is only displayed for Limit table types.

Configurable Command Option	Description
cac-scope	Scope at which CAC limits are applied within each entry in a Policy Set table.
callee	Callee settings
callee-codec-list	List of codecs which the callee leg of a call is allowed to use
callee-hold-setting	The callee hold setting supported
callee-inbound-policy	Set callee inbound Session Description Protocol (SDP) policy table
callee-outbound-policy	Set callee outbound SDP policy table
callee-privacy	The level of privacy processing
callee-sig-qos-profile	QoS profile to use for callee signalling
callee-video-qos-profile	QoS profile to use for callee video media
callee-voice-qos-profile	QoS profile to use for callee voice media
caller	Caller settings
caller-codec-list	List of codecs which the caller leg of a call is allowed to use
caller-hold-setting	The caller hold setting supported
caller-inbound-policy	Set caller inbound sdp policy table
caller-outbound-policy	Set caller outbound sdp policy table
caller-privacy	the level of privacy processing
caller-sig-qos-profile	QoS profile to use for caller signalling
caller-video-qos-profile	QoS profile to use for caller video media
caller-voice-qos-profile	QoS profile to use for caller voice media
codec-restrict-to-list	Restrict to using codecs from a configured codec list
early-media-deny	Do not allow early-media
early-media-timeout	Duration for which to allow early media
early-media-type	Directions in which to allow early media
match-value	Match-value of an entry in a CAC Limit table
max-bandwidth	Maximum bandwidth
max-call-rate-per-scope	Maximum call rate
max-channels	Maximum number of channels
max-in-call-msg-rate	Configure maximum rate of in-call messages. See description of in-call messages in the ?\$paranum>CAC Rate Limiting? section on page 7-45.
max-num-calls	Maximum number of calls

 Table 7-6
 CAC Table Entry Configurable Command Options

Configurable Command Option	Description
max-out-call-msg-rate	Configure maximum rate of out-of-call messages
max-regs	Maximum subscriber registrations
max-regs-rate-per-scope	Maximum subscriber registrations rate
max-updates	Maximum updates to call media
media	Media Flag
transcode-deny	Sets transcoding to forbidden for the admission control entry
transport	Transport Protocol Parameters

Table 7-6	CAC Table Entry Configurable Command Options (continued)
-----------	--

Nonlimiting CAC Options

CAC allows you to configure policy for accepting or rejecting calls based on limit options such as max-num-calls and max-bandwidth. The CAC scope is used when policing limit options. CAC also allows you to apply a property to a call (rather than a limitation) with nonlimiting options, such as caller-inbound-policy. Scopes have no meaning for nonlimiting options.

You can configure multiple CAC policies that all apply to a given event (using a Policy Set table type). A nonlimiting option can be given contradictory values in each of these policies. CAC determines what its behavior towards that event is by examining the setting of the option in each applicable policy and applying a rule to produce a "derived value" for the field. If the option is not defined in any policy, then a default behavior is defined. When the SBC is deriving a value for a nonlimiting field, it should disregard all policies in which that field has not been defined by the user. The SBC derives that value based on the assigned behavior for the specific nonlimiting option. The behavior for the nonlimiting options takes one of the following values:

- Last non-default value used. Options of this type take the last non-default value as the derived value. For example, caller-inbound-policy uses the last found non-zero length sdp policy name as the derived value.
- Most restrictive value used. Options of this type take as the derived value the Policy Value that most restricts the behavior of the SBC.
- First non-default value used. Options of this type use the first non-default value as the derived value. For example, caller-voice-qos-profile uses the first non-zero length voice QoS profile name as the derived value.
- All found values combined. Options of this type perform a bitwise-OR to obtain a cumulative value as the derived value.

Nonlimiting Option in a CAC Entry	Behavior of Derived Value
branch bandwidth-field	Last nondefault value used
branch codec-list	Last nondefault value used
branch hold-setting	Last nondefault value used
branch inbound-policy	Last nondefault value used
branch media-description	All found values combined
branch media-type	Last nondefault value used

Table 7-7 Nonlimiting Options in CAC Entries

Nonlimiting Option in a CAC Entry	Behavior of Derived Value
branch outbound-policy	Last nondefault value used
branch privacy	Most restrictive value used
branch secure-media	All found values combined
branch sig-qos-profile	First nondefault value used
branch tel-event payload type	Last nondefault value used
branch video-qos-profile	First nondefault value used
branch voice-qos-profile	First nondefault value used
callee-bandwidth-field	Last nondefault value used
callee-codec-list	Last nondefault value used
callee-hold-setting	Last nondefault value used
callee-inbound-policy	Last nondefault value used
callee media-description, callee secure media	All found values combined
callee media-type	Last nondefault value used
callee-outbound-policy	Last nondefault value used
callee-privacy	Most restrictive value used
callee-sig-qos-profile	First nondefault value used
callee tel-event payload type	Last nondefault value used
callee-video-qos-profile	First nondefault value used
callee-voice-qos-profile	First nondefault value used
caller-bandwidth-field	Last nondefault value used
caller-codec-list	Last nondefault value used
caller-hold-setting	Last nondefault value used
caller-inbound-policy	Last nondefault value used
caller media-description, caller secure media	All found values combined
caller media-type	Last nondefault value used
caller-outbound-policy	Last nondefault value used
caller-privacy	Most restrictive value used
caller-sig-qos-profile	First nondefault value used
caller tel-event payload type	Last nondefault value used
caller-video-qos-profile	First nondefault value used
caller-voice-qos-profile	First nondefault value used
codec-restrict-to-list	Last nondefault value used
early-media-deny	Most restrictive value used
early-media-timeout	Most restrictive value used
early-media-type	Most restrictive value used

Table 7-7	Nonlimiting	Options	in CAC	Entries	(continued
Iadie /-/	Noniimiting	Options	In CAC	Entries	(continuea

Nonlimiting Option in a CAC Entry	Behavior of Derived Value
media address preserve, media bandwidth-field ignore, media tel-event interworking	All found values combined
sdp-media-profile	Last nondefault value used
transcode-deny	Most restrictive value used
transport srtp	Most restrictive value used

Table 7-7 Nonlimiting Options in CAC Entries (continued)

Configuring Directed Nonlimiting CAC Policies

In releases prior to Release 3.5.0, you can use the **caller** command and the **callee** command to configure the CAC policy entries that are applied when an adjacency, adjacency group, or account is either a caller or a callee in a call. However, this approach does not permit the configuration of certain directed nonlimiting CAC policy fields on specific adjacencies, adjacency groups, or accounts, in a way that is independent of whether the adjacencies, adjacency groups or accounts are the callees or the callers on the calls. The following example illustrates this limitation.

Suppose the following sequence of commands is part of the configuration of an entry in a CAC table:

```
cac-policy-set 3
first-cac-table cac-tb1
cac-table cac-tb1
table-type limit adjacency
entry 1
match-value adj1
caller port-range-tag adj-name
callee port-range-tag adj-name
action cac-complete
```

If there is a call from the adj1 adjacency to the adj2 adjacency, the settings specified for the caller in this example is applied to adj1. At the same time, the callee settings are applied to adj2 because that adjacency is the callee in this call. In a scenario such as this one, you might not want to apply any configuration to the other adjacency (the adj2 adjacency, in this example) involved in the call. The **branch** command helps overcome this limitation. This command has been introduced in Release 3.5.0.

In the preceding example, the **branch** command can be used to replace the **caller** command and the **callee** command as follows:

```
cac-policy-set 3
first-cac-table cac-tb1
cac-table cac-tb1
table-type limit adjacency
entry 1
match-value adj1
branch port-range-tag adj-name
action cac-complete
```

Note

The **branch** command is not a replacement for the **caller** command and the **callee** command pair in scenarios in which you want to apply settings to both the caller adjacency and the callee adjacency.

With this configuration, the settings specified in the **branch** command are applied to the adj1 adjacency. For a call from the adj2 adjacency to the adj1 adjacency, the same settings are applied to the adj1 adjacency. For this call, no settings are applied to adj2 or any other adjacency that calls or is called by adj1.

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The following are the features of the branch command:

- If a branch setting (that is, the **branch** command) and a caller-callee pair setting (that is, the **caller** and **callee** command pair) are configured in different policy entries, the setting in the last entry of the configuration takes precedence.
- If two branch settings, each in a different policy entry, are encountered, the setting in the last entry that is encountered takes precedence.
- If a branch setting and a caller-callee setting are in the same policy entry, the branch setting takes precedence over the caller-callee setting.

The following sample configuration illustrates how the branch command works:

```
cac-policy-set 3
    first-cac-table cac-tb1
   cac-table cac-tb1
     table-type limit adjacency
     entrv 1
     match-value phone2
     branch port-range-tag adj-name
     caller port-range-tag string tagB_cac
      callee port-range-tag string tagA_cac
      action cac-complete
     entry 2
      match-value phone1
      branch port-range-tag string tagA_cac
      caller port-range-tag adj-name
     callee port-range-tag adj-name
      action cac-complete
   complete
   cac-policy-set global 3
media-address ipv4 209.165.202.130
 port-range 10000 15000 any
 port-range 15002 15003 any tag phone1
 port-range 16002 16003 any tag phone2
 port-range 17002 17003 any tag tagA_cac
 port-range 18002 18003 any tag tagB_cac
```

In this example, the call goes from phone 1 to phone 2. The following sequence of events takes place during the call:

- 1. Matching is performed on the source adjacency, phone 1, which matches entry 2. Here, the branch entry refers to the caller side, so the caller entry is overridden. After this first policy match is performed, port-range-tag is set to tagA_cac on side A. In addition, the callee port tag is set to adj-name.
- 2. Matching is performed on the destination adjacency, phone2, which matches entry 1. Here, the branch entry refers to the callee side, so the callee entry is overridden. This entry sets the caller side port-range-tag to tagB_cac. In other words, adj_name is assigned as the callee side port-range-tag. These settings take precedence over the values assigned in the previously matched entry, entry 2, because these settings are assigned later.

The outcome is that the tagB_CAC port is used on side A, and an adj-name port, phone2, is used on side B.

Media Line Removal

Media line removal feature provides the ability to strip or pad disabled media descriptions (m-lines with zero port) when sending an offer or answer to interoperate with various non-compliant devices.

Where the SDP being forwarded represents an answer, the media line which was removed from the forwarded offer is identified and a dummy media line is inserted into the same location. This is required for the compliant partner to match appropriate media line requests and responses.

Where the SDP being forwarded is a future offer, it uses offer modification to effectively shuffle-up media lines allowing the "padding" dummy media lines to be added to the end of the forwarded SDP.

SBC's transmit behavior is independently configured for the caller and callee sides of the call using the following options:

- strip new on offer—removes disabled media streams in forwarded offers which are new or unknown to the recipient of the offer.
- strip all on offer—removes all disabled media streams from forwarded offers, whether known to the recipient of the offer or not.
- strip on answer—removes all disabled media streams from forwarded answers.
- do not pad on offer—stops SBC from padding forwarded offers with disabled media streams. This means that a forwarded offer may not comply because it may contain less media lines than previous offers.

Note

The "strip new on offer" and "strip all on offer" result in removal of m-lines from the forwarded offer. The missing lines are not "padded in" and there is no need to set the "do not pad on offer" option to achieve this. The "do not pad on offer" option only affects media lines that were missing from the received offer.

On selecting the appropriate option, the SDP to be forwarded is created with disabled media portions deleted, rather than the existing behavior of setting the port to zero.

Multiple SBC Media Bypass

The multiple SBC media bypass feature can send media packets directly from the answerer to the original offerer. When the SBC detects that the media packets are being looped back unnecessarily, as shown in Figure 7-5, the SBC removes itself from the loop so that the media packets can flow directly between the endpoints.



Partial Media Bypass

When at least one SBC from the network has to anchor the media because endpoints cannot communicate directly, the other SBC gets bypassed as shown in Figure 7-6. If the media bypass type is explicitly configured to be partial, only IP realm and VPN configuration on the adjacency can be used to determine whether media bypass is possible. Because media bypass tags are not used, the VPN names must be globally unique across all the SBCs for partial media bypass to work.





Figure 7-7 shows an example of media bypass across two or more SBC devices

In networks where direct media packets cannot pass, the feature creates an optimized media path through a group of SBCs, to avoid unnecessary media hops through the SBC network.

With the multiple SBC media bypass feature, the SBC can transmit an extra set of media addresses alongside an SDP offer. These are the original media addresses that the SBC itself received from the offerer. The original media addresses are placed in a separate multiple SBC media bypass feature information element. These addresses are associated with information about the media plane connectivity of the offerer. A downstream SBC uses the multiple SBC media bypass feature connectivity information to determine whether it can re-instate the original media addresses by rewriting the SDP offer to include them. This enables the media packets to directly pass between the answerer and the original offerer.

The multiple SBC media bypass feature information can also be used by a group of SBCs to optimize the media path and to avoid unnecessary media hops through the SBC network. The SDP answer is accompanied by an indication of whether the feature was successful or not. The SBC uses this indication to determine whether it has been bypassed or whether it is still in the media path. When many SBCs appear in the media path, they collectively build up a stack of alternative media addresses for each media streams in the offer, where each element of the stack has associated connectivity information.

The SBCs determines which endpoints and intermediate hops are connected to decide which intermediate entities can be bypassed. Such connectivity information is passed on by tags in the multiple SBC media bypass feature information elements. Two remote endpoints on an adjacency can be connected if they have one or more matching tags. Therefore, tags must be globally unique for the multiple SBC media bypass feature protocol to work.

Continuing Media Bypass After a Session Refresh

When a media bypass call is in progress, the SIP registrar does not process the media exchanged by the endpoints. Therefore, the registrar uses signaling to detect failures in the session. The registrar sends a session refresh request to check whether a session is alive. The session refresh request is in the form of an INVITE or UPDATE message containing a copy of the SDP forwarded by the registrar during the original call setup.

When the SBC receives the INVITE message from the SIP registrar, it does not correlate the SDP in the message with the SDP sent earlier in the call. The SBC processes the SDP in the INVITE message as normal and creates an SDP offer to send to either the caller endpoint or callee endpoint. From the perspective of the endpoint, the INVITE message is an attempt to renegotiate the media for the call. The endpoint processes the offer and creates an answer that is consistent with the offer. This answer is returned to the registrar through the SBC.

In the answer, the port number for each media stream in the call is different from the port number of the previous media stream. This mismatch in the port number could cause the registrar to send a late INVITE message to the endpoint. An endpoint that does not support the receipt of a late INVITE message for a renegotiation would reject the message. The call fails because the media is being sent from the endpoint to the SBC, from where the media is dropped. To circumvent this issue, renegotiation is enabled by default so that the same path is used to resume exchange of media packets between the endpoints. This feature ensures that media bypass calls continue to bypass the media after a session refresh.



You can disable or enable renegotiation.

Restrictions

The multiple SBC media bypass feature has the following restrictions:

- Media bypass is not supported for H.323 calls.
- Media services, such as provisioned transcoding, transrating, and DTMF Interworking preclude media, are sent directly between endpoints. For a given call, if the administrator has configured media bypass settings and if media bypass is possible, then it takes precedence over other media services. However, if lawful intercept (LI) is provisioned on the SBC, LI would take precedence over the multiple SBC media bypass feature.
- The SBC does not support the feature when one endpoint is IPv4 and the other endpoint is IPv6. Because the endpoints cannot understand the traffic they receive.
- The SBC does not support the feature when one endpoint is SIP and the other endpoint is H.323 if SIP-H.323 interworking is enabled. Because the endpoints cannot understand the traffic they receive.

Performance Impact

When the multiple SBC media bypass feature is enabled, it has the following performance impact on the Cisco ASR 1000 series routers:

- The SBCs signaling performance decreases by a small fraction, due to the increased parsing and message manipulation costs. However there is a corresponding gain on media resources for every call that successfully negotiates the feature.
- The transient occupancy of each call setup increases by a multiple of the size of the multiple SBC media bypass feature information that is encoded in the SIP message, plus a small amount of control information. For SDP sizes of 300 bytes, this is predicted to be around 1500 bytes in total. However,

the steady-state occupancy for calls that successfully negotiate the multiple SBC media bypass feature decreases as no media resources are required for those calls. This saves approximately 10000 bytes per call.

For more information on configuring the multiple SBC media bypass feature, see the ?\$paranum>Configuring Multiple SBC Media Bypass? section on page 7-136. For configuration examples of the feature, see the ?\$paranum>Example: Multiple SBC Media Bypass? section on page 7-161. For video of the example that explains how the SBC Media Bypass feature works, see http://www.cisco.com/en/US/docs/routers/asr1000/configuration/guide/SBCU3.5S/sbc_media_bypass. html.

Common IP Address Media Bypass

This section contains the following topics:

- Restrictions for Common IP Address Media Bypass, page 7-43
- Information About Common IP Address Media Bypass, page 7-43
- Features of Common IP Address Media Bypass, page 7-44

Restrictions for Common IP Address Media Bypass

The following are restrictions for the Common IP Address Media Bypass feature:

- This feature is not supported for H.323 adjacencies.
- This feature is not supported in a scenario in which endpoints are behind the same NAT device but are not registered with the SBC.
- This feature is not supported in a scenario in which the caller endpoint and callee endpoint are behind different NAT devices even when there is connectivity between the networks defined by each NAT device. The SBC always relays media between two such endpoints.
- This feature is not supported in a scenario in which only one of the endpoints is behind a NAT device. The SBC always relays media between two such endpoints.

Information About Common IP Address Media Bypass

When you enable the Multiple SBC Media Bypass feature, the SBC bypasses or relays media between the endpoints of an adjacency depending on the media bypass tags presented by the endpoints. If the tags match, the SBC determines that there is media connectivity between the endpoints and, therefore, bypasses itself from the media flow between the endpoints. In contrast, if the tags do not match, the SBC determines that there is no media connectivity between the endpoints and, therefore, relays media between the endpoints.

Note

For detailed information about the Multiple SBC Media Bypass feature, see the ?\$paranum>Multiple SBC Media Bypass? section on page 7-39.

An organization can use a hosted PBX solution that is owned and managed by a service provider. Typically, a hosted PBX solution can serve many organizations and, therefore, serve multiple NAT devices. There may be a scenario in which there are multiple NAT devices behind a single adjacency. In such a scenario, the SBC must bypass media for the endpoints behind the same NAT device and relay media for the endpoints that are behind different NAT devices. In releases prior to Release 3.6.0, the only way to achieve this is to configure an adjacency for each NAT device. This approach increases the overhead involved in managing the network.

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The Common IP Address Media Bypass feature is an enhancement to the Multiple SBC Media Bypass feature. It offers an alternative to the approach of creating an adjacency for each NAT device.

When you configure the Common IP Address Media Bypass feature, the SBC assigns each endpoint behind a NAT device a media bypass tag that is based on the corresponding endpoint's external, NAT IP address. These media bypass tags are used by the SBC to determine whether the caller endpoint and callee endpoint belong to the same NAT network. If the media bypass tag of the caller endpoint matches the media bypass tag of the callee endpoint, the SBC bypasses media. If the tags do not match, the SBC relays media.



The Common IP Address Media Bypass feature does not introduce any change in the mechanism by which the SBC compares the media bypass tags of the caller endpoint and callee endpoint. In other words, the SBC does not distinguish between the feature or method by which media bypass tags are created. The SBC only compares the tags and bypasses media when the tags match.

When the Common IP Address Media Bypass feature is not configured or is disabled, media-bypass decisions are taken by the SBC on the basis of the media bypass tags configured at the adjacency level. If media bypass tags are not configured, media-bypass decisions are taken on the basis of the autogenerated tags that are based on VPN IDs.

When the Common IP Address Media Bypass feature is configured and enabled:

- If the caller endpoint or callee endpoint is a registered subscriber that has been identified at registration time as being behind a NAT device, the SBC generates a media bypass tag and uses that media bypass tag for the call leg.
- If the caller endpoint or callee endpoint is not a registered subscriber or is not behind a NAT device, the SBC uses the tag that is created by the **media bypass tag** command, if such a tag is present, for the call leg.
- If the caller endpoint or callee endpoint is not a registered subscriber or is not behind a NAT device and if there are no configured tags, the SBC generates a media bypass tag based on the VPN ID of the endpoint, for the call leg.

During the call, if both the endpoints have media bypass tags that match, the SBC determines that both the endpoints are behind the same NAT device and it bypasses media for that call leg. Conversely, if the media bypass tags do not match, if either endpoint does not have a media bypass tag, or if either endpoint is not a registered subscriber, the SBC relays media for that call leg.

The following is the format of the media bypass tag generated by this feature:

nat-VPN-ID-IP-address

In this format, *IP-address* is the source IP address of the most recent non-fast-pathed, successful REGISTER request from the endpoint. The IP address can be in IPv4 format or IPv6 format.

The following are sample media bypass tags generated by this feature:

- nat-123-192.0.2.6
- nat-254-192.0.26.18
- nat-2233-2001:DB8::AC10:FE01

Features of Common IP Address Media Bypass

The following are additional points about how the Common IP Address Media Bypass feature works:

• After this feature is configured, the SBC can detect whether an endpoint is behind a NAT device by using the existing adjacency configuration features:

- If the **no nat** command is configured for an adjacency, an endpoint behind that adjacency is recognized as being behind a NAT if the IP address in the Via header of the SIP messages from that adjacency is different from the IP address from which the request was received.
- If the nat force-on command is configured, all endpoints are assumed to be behind a NAT.
- If the **nat force-on** command is configured and an endpoint is not behind a NAT, the SBC relays media for calls to and from such an endpoint. Note that if this feature is disabled, the SBC bypasses media. When you enable this feature, the SBC starts relaying media.
- If this feature is configured while an adjacency is active, only new calls that are processed by that adjacency are affected by this feature. Existing calls are not affected.
- This feature is independent of whether the initial INVITE message contains SDP content because the media bypass tag is not added to the SDP content.
- This feature is supported by all forms of media bypass:
 - Simple media bypass, in which the caller endpoint and callee endpoint are associated with local adjacencies.
 - Two-call media bypass, in which the SBC forwards the call to a softswitch or registrar, which then loops the call request back to the SBC.
 - N-call media bypass, in which a call is looped through the SBC multiple times.
 - Multi-SBC media bypass, in which a call is looped through multiple SBCs.

CAC Rate Limiting

You can limit the number or the rate of new calls accepted and the number of media renegotiations within a call. However, limits are not placed on the following:

- Media renegotiations which do not actually change the characteristics of the call.
- Any other in-call messages.

In-call messages include any message within the context of a call, including provisional responses during call setup and call renegotiation messages, but not including call setup or tear-down messages.

• Internally-generated messages



You cannot specify limits at the granularity of a specific SIP or H.323 message.

You can also limit the rate and number of registrations passing through the Cisco Unified Border Element (SP Edition). However, limits are not placed on any other out-of-call messages. (An out-of-call message is any messages which is not following within the context of a call and which does not form part of registration processing. These are always classified as either a request or a response.)

You can rate limit all in-call and out-of-call messages.

This includes in-call messages at all scopes, as normal. For example:

- Configuration at the "per-call" scope allows you to limit the rate at which an endpoint sends messages within a call.
- Configuration at the "dst-adjacency" scope allows you to limit the total rate of in-call messages sent out of an adjacency within all of the calls using that adjacency. (This could ensure that the load out of an adjacency never exceeds that which the attached network entity can cope with.)

The following messages are not rate-limited:

- SIP INVITE requests: 200 responses and ACK messages
- SIP PRACK messages and response
- SIP BYE messages and responses
- Any SIP message with non-duplicate SDP on
- For H.323 calls: Q.931 SETUP, Q.931 CONNECT and Q.931 RELEASE messages.

You can place restrictions on the rate at which out-of-call messages are processed. Configuration is permitted at all scopes except per-call scope (because this scope does not exist for out-of-call messages).

The Cisco Unified Border Element (SP Edition) will gracefully reject in-call messages when the rate exceeds that specified in the CAC. When an in-call message is not processed, the Cisco Unified Border Element (SP Edition) does the following:

- For SIP messages, Cisco Unified Border Element (SP Edition) rejects the message gracefully wherever possible. The rejection is sent back to the sending endpoint, so the call is likely to survive.
- For H.323 messages, Cisco Unified Border Element (SP Edition) drops the message because they usually cannot be gracefully rejected. This is likely to be disruptive for the call.

The Cisco Unified Border Element (SP Edition) gracefully rejects out-of-call messages when the rate exceeds that specified in CAC.

All rate limits must be protocol stack independent; limits must police SIP and H323messages.

In addition to configuring blacklists based on a number of CAC policy failures, you can now allow blacklists to be applied to endpoints that send in-call or out-of-call messages at a high rate.

Multiple CAC Averaging Periods

The user can apply different rate limits over a different averaging period by configuring a second set of rate-limiting CAC criteria. The user is able to do the following:

- Set the averaging period for the secondary rate calculation.
- Set the maximum number of new calls per minute for the secondary rate calculation, if a limit is required.
- Set the maximum number of endpoint registrations per minute for the secondary rate calculation, if a limit is required.
- Set the maximum number of in-call messages to be processed per minute for the secondary rate calculation, if a limit is required.
- Set the maximum number of out-of-call messages to be processed per minute for the secondary rate calculation, if a limit is required.

The user can configure two sets of SBC policies together that have rate-limiting criteria. The CAC rejects an event if it breaks any of the configured limits.

Subscriber Policy

A user can subscribe multiple endpoints to the network to allow them to make calls. A subscriber is one of those endpoints. In a particular network, you might want to limit each subscriber to no more than a specific number of simultaneous calls. The Subscriber Policy feature allows you to limit each subscriber to a specific number of simultaneous calls.

This feature provides the ability to configure the CAC limits. For example, you can configure the maximum number of concurrent calls, the maximum number of registrations, or the maximum call rate at different scopes, such as subscriber, subscriber category, and subscriber category prefix.

You can configure CAC tables:

- To associate a subscriber with a subscriber category. Call events between that subscriber and the core network are also associated with that same subscriber category.
- To match on a subscriber category or on a subscriber category prefix (the first n bits of the subscriber category), and then set limits when matched. The subscriber category prefix specifies the length of prefix to match. If specified, then only the first n bits of each of the call's subscriber categories is checked for a match.
- To set limits per subscriber category.
- To set limits per subscriber.

Note that when a subscriber scope is enabled, the SBC tracks an additional group of ALL "non-subscribers." The non-subscriber group is matched if the call is from a non-subscriber. Limits set in the subscriber scope apply to this non-subscriber group.

Privacy Service

The SBC provides the privacy service to ensure that requests for anonymity, as requested by a user during signaling, can be dynamically acted upon to ensure that the user's anonymity is maintained when the user leaves a trusted network. A user can request various levels of anonymity, with the privacy service removing the information that a user wants to withhold. The SBC can be configured such that individual adjacencies can be marked as trusted, untrusted, or configured in order to apply the privacy service. The privacy service is applied in a CAC policy set.

In addition to this, the SBC can edit—override or modify—a user's request for privacy when forwarding the privacy request. For example, a user can request identity of self to be withheld, but by editing the privacy request, the identity can be provided.

A user can also provide indications of anonymity in the display and presentation number. During number analysis, these calls can be detected and different analysis trees be used to progress the call.

The Privacy Service feature provides the following functions:

- Apply a privacy service based on information provided by a user when leaving a trusted domain.
- Edit a privacy service on request from a user and perform functions such as pass, strip, insert, and replace indications.
- Declare configurable trust boundaries.
- Detect calls in number analysis where the source is anonymous.
- Standard SIP header rewriting is performed by the SBC to cover the additional requirements specified in the SIP privacy header:
 - The Call ID, Server, and Contact headers are rewritten to hide the endpoint's identity.
 - Any Via headers are cached and replaced on the message with a single header identifying the SBC.

Both SIP and H323 adjacencies allow the configuration of the trusted and untrusted statuses.

For information about configuring the Privacy Service feature, see the ?\$paranum>Configuring Privacy Service? section on page 7-126.

Session Initiation Protocol

In the context of SIP, a user indicates the levels of privacy that should be applied using the Privacy header. If the SBC cannot recognize any of the tokens present in the header, the message is rejected with a 433 Anonymity Disallowed response. Similarly, a response containing a critical privacy request that cannot be met is converted to a 433 failure response for an in-call message. For an out-of-dialog message, the response is dropped to ensure that no private information gets leaked accidently.

If this is an in-call message that does not contain a privacy header, the privacy requirements are assumed to be the same as those specified in the last privacy header from the side of the call. However, if the SBC reroutes a call locally, for example, a SIP 3xx redirect response, it discards the previously learnt privacy requirements on the side of the call that has been rerouted.

The following events occur when privacy services are applied to a request or response:

• When the privacy service based on a user, *Privacy: user*, is applied to a request or response, the Reply-To, Call-Info, User-Agent, Organization, Subject, In-Reply-To, Warning, and Server headers are stripped from the message.

Also, when the privacy service based on a user, *Privacy: user*, is applied to a request, the URI in the From header is rewritten to anonymous@anonymous.invalid. The original URI is stored for replacement on responses. The display name in the From header is removed, and any further header manipulation rules that are configured as part of the user ID privacy are applied to the message.

- When the privacy service based on ID value, *Privacy: id*, is applied to a request or a response, the P-Preferred-ID, P-Asserted-Identity, and Remote-Party-Id headers are stripped from the message.
- When the privacy service based on session privacy, *Privacy: session*, is applied to a request or a response, media bypass is disallowed. However, if the session privacy is critical, and it is too late to disable media bypass, the call is torn down.
- When the privacy service based on header privacy, *Privacy:header*, is applied to a request or response, Record-Route or Route headers, if any, are removed and stored. They are restored on the responses within the dialog. If any further header manipulation rules are configured, they are applied to the message. The SBC strips the Privacy header from the ongoing message and removes the *privacy* option-tag, if any, from the Proxy-Require header.

Users can dynamically request for privacy service. This service can be applied by inserting *Privacy: header* based on RFC 3323 and RFC 3325.

Privacy Service on SIP Requests

Table 7-8 lists the behavior of the privacy service when it is applied on SIP requests, and *Privacy: header* is present to indicate the appropriate level of privacy to be applied.

Header Name	None	User	Header	ID
From		Set to anonymous value: anonymous@anonymou s.invalid		
Contact	—	—	Rewritten	—
Reply-to	—	Stripped	—	—
Via	Stripped	Stripped	Stripped	Stripped
Call-Info	—	Stripped	—	—

Table 7-8 Privacy Service on SIP Requests

User-Agent	—	Stripped		—
Organization	—	Stripped		—
Server	—	—		
Subject	—	Stripped		
Call-ID	Rewritten	Rewritten	Rewritten	Rewritten
In-Reply-To	—	Stripped		
Warning	—	—		
P-Asserted-Identity	—	—		Stripped
P-Preferred-Identity	—	—		Stripped
Remote-Party-ID	—	—		Stripped
Record-Route	—	—	Stripped	

Table 7-8 Privacy Service on SIP Requests (continued)

Privacy Service on SIP Responses

Table 7-9 lists the behavior of the privacy service when it is applied on SIP responses.

Header Name	None	User	Header	ID
From	_			—
Contact			Rewritten	—
Reply-to		Stripped		—
Via				—
Call-Info		Stripped		—
User-Agent		Stripped		—
Organization	—	Stripped	—	—
Server	—	Stripped	—	—
Subject	—	—	—	—
Call-ID	Rewritten	Rewritten	Rewritten	Rewritten
In-Reply-To	—	—	—	—
Warning	—	Stripped	—	—
P-Asserted-Identity	—	—	—	Stripped
P-Preferred-Identity	—	—	—	Stripped
Remote-Party-ID	—	—	—	Stripped
Record-Route			Stripped	_

Table 7-9 Privacy Service on SIP Responses

Privacy Service on H.323

The SBC treats the following H.323 protocol events as requests for the privacy service:

- On a Q.931 Setup, the caller address presentation restriction is requested if the Q.931 callingPartyNumber is present, and contains a presentationIndicator set to 3, presentation restricted, or, the H.225 presentationIndicator is present and set to presentationRestricted.
- On a Q.931 Connect, callee address presentation restriction is requested if the Q.931 connectedNumber is present, and contains a presentationIndicator set to 3, presentation restricted, or, the H.225 presentationIndicator is present and set to presentationRestricted.

When there is a conflict between the two presentationIndicators, the value in the Q.931 callingPartyNumber, the connectedNumber, takes precedence.

H.323 to SIP

A presentation restriction indication that is received for the callingPartyNumber or connectedNumber elements in an H.323 message is considered a request for *header;id;critical* privacy when being translated to a SIP privacy request.

If the presentation restriction is requested by the H.323 side, the URI in the From header is rewritten with anonymous@anonymous.invalid whether or not the SBC is acting as a privacy service.

When interworking with the SIP, the privacy service is always applied.

SIP to H.323

A SIP-signaled request for *id* or *header* privacy is translated into an H.323 presentation restriction on outgoing addresses, if any. All the other SIP privacy tokens are ignored.

Message, Policy, and Subscriber Statistics

From Cisco IOS XE Release 3.3S, enhancements have been made to the following statistics:

- Call Statistics, page 7-50
- CAC Statistics, page 7-55
- Subscriber Statistics, page 7-58

Call Statistics

The call-related statistics have been enhanced to include the following two features:

- Number of calls completed during a period—The summary periods of the call-related statistics includes the total number of calls that have been completed. A call is completed because of a signaling message received from an upstream or a downstream device. For SIP calls, a call is completed when a participating endpoint sends a BYE message. For H.323 calls, a call is completed when a participating endpoint sends a ReleaseComplete message with causeValue of 16, which indicates a *normal call clearing*.
- Running total for the calls statistics—The summary periods of the call-related statistics includes a *current-indefinite* time value. This time value provides the call-statistics for a period since the value has been last reset. Initially, the *current-indefinite* time value displays the statistics for the time

period since the router was booted. After the value has been reset, it displays the statistics for the time when the last reset was done. The time is in a UTC format where the year, month, day and time is displayed.

The following commands are used for displaying the call-related statistics and resetting the call-related statistics:

- The show sbc *sbc-name* sbe call-stats {all | global | per-adjacency *adjacency-name* | src-account *name* | dst-account *name* | src-adjacency *name* | dst-adjacency *name* } *period* command—Lists the statistics pertaining to all the calls on a SBE for a particular period, such as *currentindefinite*.
- The clear sbc sbc-name sbe call-stats [all | dst-account account-name | dst-adjacency adjacency-name | global | src-account account-name | src-adjacency adjacency-name | per-adjacency adjacency-name] [all | current-indefinite] command—Clears the call statistics on a SBE by the current-indefinite period.

The following example shows how the **show sbc sbe call-stats all adj1 currentindefinite** command displays statistic pertaining to all calls on the SBE for the current-indefinite period:

Router# show sbc SBC2 sbe call-stats all currentindefinite

```
statistics for the current indefinite for source adjacency phone1
Call count totals:
  Total call attempts =
                                                                       1
  Total active calls =
                                                                       0
  Total active IPv6 calls =
                                                                       0
  Total activating calls =
                                                                       0
  Total de-activating calls =
                                                                       0
  Total active emergency calls =
                                                                       0
  Total active e2 emergency calls =
                                                                       0
  Total IMS rx active calls =
                                                                       0
  Total IMS rx call renegotiation attempts =
                                                                       0
  Total SRTP-RTP interworked calls =
                                                                       0
  Total active calls not using SRTP =
                                                                       0
  Total active transcoded calls =
                                                                       0
  Total active transrated calls =
                                                                       0
  Total calls completed =
                                                                       1
General call failure counters:
  Total call setup failures =
                                                                       0
  Total active call failures =
                                                                       0
  Total failed call attempts =
                                                                       0
  Total failed calls due to update failure =
                                                                       0
  Total failed calls due to resource failure =
                                                                       0
  Total failed calls due to congestion =
                                                                       Ω
  Total failed calls due to media failure =
                                                                       0
  Total failed calls due to signaling failure =
                                                                       0
  Total failed calls due to IMS rx setup failure =
                                                                       0
  Total failed calls due to IMS rx renegotiation failure =
                                                                       0
  Total failed calls due to RTP disallowed on call leg =
                                                                       0
  Total failed calls due to SRTP disallowed on call leg =
                                                                       0
Policy control failures:
  Call setups failed due to NA =
                                                                       Λ
  Call setups failed due to RTG =
                                                                       0
  Call setups failed due to CAC =
                                                                       0
  CAC fails due to number of calls limit =
                                                                       0
  CAC fails due to call rate limit =
                                                                       0
  CAC fails due to bandwidth limit =
                                                                       0
  CAC fails due to number of media channels limit =
                                                                       0
  CAC fails due to number of media update limit =
                                                                       0
  CAC message drops due to mid call message rate limit =
                                                                       0
  CAC message drops due to out of call message rate limit =
                                                                       0
```

```
Stats Reset Timestamp:
 Timestamp when stats for this summary period were reset =
                                                                   2011/03/07 03:27:36
statistics for the current indefinite for destination adjacency phone2
Call count totals:
  Total call attempts =
                                                                       1
                                                                       0
 Total active calls =
                                                                       0
  Total active IPv6 calls =
  Total activating calls =
                                                                       0
  Total de-activating calls =
                                                                       0
  Total active emergency calls =
                                                                       0
 Total active e2 emergency calls =
  Total IMS rx active calls =
                                                                       0
  Total IMS rx call renegotiation attempts =
                                                                       0
  Total SRTP-RTP interworked calls =
                                                                       0
 Total active calls not using SRTP =
                                                                       0
                                                                       0
  Total active transcoded calls =
  Total active transrated calls =
                                                                       0
  Total calls completed =
                                                                       1
General call failure counters:
  Total call setup failures =
                                                                       0
  Total active call failures =
                                                                       0
  Total failed call attempts =
                                                                       0
  Total failed calls due to update failure =
                                                                       0
                                                                       0
  Total failed calls due to resource failure =
  Total failed calls due to congestion =
                                                                       0
  Total failed calls due to media failure =
                                                                       0
  Total failed calls due to signaling failure =
                                                                       0
  Total failed calls due to IMS rx setup failure =
                                                                       0
  Total failed calls due to IMS rx renegotiation failure =
                                                                       0
  Total failed calls due to RTP disallowed on call leg =
                                                                       0
  Total failed calls due to SRTP disallowed on call leg =
                                                                       0
Policy control failures:
  Call setups failed due to NA =
                                                                       0
  Call setups failed due to RTG =
                                                                       0
  Call setups failed due to CAC =
                                                                       0
  CAC fails due to number of calls limit =
                                                                       0
  CAC fails due to call rate limit =
                                                                       0
  CAC fails due to bandwidth limit =
                                                                       0
  CAC fails due to number of media channels limit =
                                                                       0
  CAC fails due to number of media update limit =
                                                                       0
  CAC message drops due to mid call message rate limit =
                                                                       0
  CAC message drops due to out of call message rate limit =
                                                                       0
Stats Reset Timestamp:
  Timestamp when stats for this summary period were reset =
                                                                   2011/03/07 03:27:36
statistics for the current indefinite for source account sourcel
Call count totals:
  Total call attempts =
                                                                       1
                                                                       0
  Total active calls =
 Total active IPv6 calls =
                                                                       0
 Total activating calls =
                                                                       0
  Total de-activating calls =
                                                                       0
  Total active emergency calls =
                                                                       0
  Total active e2 emergency calls =
                                                                       0
  Total IMS rx active calls =
                                                                       0
  Total IMS rx call renegotiation attempts =
                                                                       0
  Total SRTP-RTP interworked calls =
                                                                       0
  Total active calls not using SRTP =
                                                                       0
  Total active transcoded calls =
                                                                       0
  Total active transrated calls =
                                                                       0
                                                                       1
  Total calls completed =
```

```
General call failure counters:
  Total call setup failures =
                                                                       Λ
  Total active call failures =
                                                                       0
  Total failed call attempts =
                                                                       0
  Total failed calls due to update failure =
                                                                       Ω
  Total failed calls due to resource failure =
                                                                       0
  Total failed calls due to congestion =
                                                                       0
  Total failed calls due to media failure =
                                                                       0
  Total failed calls due to signaling failure =
                                                                       0
  Total failed calls due to IMS rx setup failure =
                                                                       0
  Total failed calls due to IMS rx renegotiation failure =
                                                                       0
  Total failed calls due to RTP disallowed on call leg =
                                                                       0
  Total failed calls due to SRTP disallowed on call leg =
                                                                       0
Policy control failures:
  Call setups failed due to NA =
                                                                       0
  Call setups failed due to RTG =
                                                                       0
  Call setups failed due to CAC =
                                                                       0
  CAC fails due to number of calls limit =
                                                                       0
  CAC fails due to call rate limit =
                                                                       0
  CAC fails due to bandwidth limit =
                                                                       0
  CAC fails due to number of media channels limit =
                                                                       0
  CAC fails due to number of media update limit =
                                                                       0
  CAC message drops due to mid call message rate limit =
                                                                       0
  CAC message drops due to out of call message rate limit =
                                                                       0
Stats Reset Timestamp:
  Timestamp when stats for this summary period were reset =
                                                                   2011/03/07 03:27:36
statistics for the current indefinite for destination account dest1
Call count totals:
  Total call attempts =
                                                                       1
  Total active calls =
                                                                       0
  Total active IPv6 calls =
                                                                       0
  Total activating calls =
                                                                       0
  Total de-activating calls =
                                                                       Ω
  Total active emergency calls =
                                                                       0
  Total active e2 emergency calls =
                                                                       0
  Total IMS rx active calls =
                                                                       0
  Total IMS rx call renegotiation attempts =
                                                                       0
  Total SRTP-RTP interworked calls =
                                                                       0
  Total active calls not using SRTP =
                                                                       0
  Total active transcoded calls =
                                                                       0
  Total active transrated calls =
                                                                       0
  Total calls completed =
                                                                       1
General call failure counters:
  Total call setup failures =
                                                                       0
  Total active call failures =
                                                                       0
  Total failed call attempts =
                                                                       0
  Total failed calls due to update failure =
                                                                       0
  Total failed calls due to resource failure =
                                                                       0
  Total failed calls due to congestion =
                                                                       0
  Total failed calls due to media failure =
                                                                       0
  Total failed calls due to signaling failure =
                                                                       0
  Total failed calls due to IMS rx setup failure =
                                                                       0
  Total failed calls due to IMS rx renegotiation failure =
                                                                       0
  Total failed calls due to RTP disallowed on call leg =
                                                                       0
  Total failed calls due to SRTP disallowed on call leg =
                                                                       0
Policy control failures:
  Call setups failed due to NA =
                                                                       0
  Call setups failed due to RTG =
                                                                       0
                                                                       0
  Call setups failed due to CAC =
```

```
CAC fails due to number of calls limit =
                                                                      0
  CAC fails due to call rate limit =
                                                                      0
  CAC fails due to bandwidth limit =
                                                                      0
  CAC fails due to number of media channels limit =
                                                                      0
  CAC fails due to number of media update limit =
                                                                      0
  CAC message drops due to mid call message rate limit =
                                                                      0
                                                                      0
  CAC message drops due to out of call message rate limit =
Stats Reset Timestamp:
  Timestamp when stats for this summary period were reset =
                                                                  2011/03/07 03:27:36
statistics for the current indefinite for global counters
Call count totals:
  Total call attempts =
                                                                      1
  Total active calls =
                                                                       0
 Total active IPv6 calls =
                                                                       0
                                                                       0
 Total activating calls =
 Total de-activating calls =
                                                                       0
  Total active emergency calls =
                                                                      0
  Total active e2 emergency calls =
                                                                       0
  Total IMS rx active calls =
                                                                      0
  Total IMS rx call renegotiation attempts =
                                                                      0
  Total SRTP-RTP interworked calls =
                                                                      0
  Total active calls not using SRTP =
                                                                      0
  Total active transcoded calls =
                                                                       0
  Total active transrated calls =
                                                                      0
                                                                      1
 Total calls completed =
General call failure counters:
  Total call setup failures =
                                                                       0
  Total active call failures =
                                                                       0
  Total failed call attempts =
                                                                       0
  Total failed calls due to update failure =
                                                                       0
  Total failed calls due to resource failure =
                                                                       0
  Total failed calls due to congestion =
                                                                      0
  Total failed calls due to media failure =
                                                                      0
  Total failed calls due to signaling failure =
                                                                      0
  Total failed calls due to IMS rx setup failure =
                                                                      0
  Total failed calls due to IMS rx renegotiation failure =
                                                                      0
  Total failed calls due to RTP disallowed on call leg =
                                                                      0
                                                                      0
  Total failed calls due to SRTP disallowed on call leg =
Policy control failures:
  Call setups failed due to NA =
                                                                       0
  Call setups failed due to RTG =
                                                                       0
  Call setups failed due to CAC =
                                                                       0
  CAC fails due to number of calls limit =
                                                                       0
  CAC fails due to call rate limit =
                                                                      0
  CAC fails due to bandwidth limit =
                                                                      0
  CAC fails due to number of media channels limit =
                                                                      0
  CAC fails due to number of media update limit =
                                                                      0
  CAC message drops due to mid call message rate limit =
                                                                      0
  CAC message drops due to out of call message rate limit =
                                                                      0
Stats Reset Timestamp:
                                                                  2011/03/07 03:27:36
 Timestamp when stats for this summary period were reset =
statistics for the current indefinite for adjacency phonel
Stats Reset Timestamp:
                                                                  2011/03/07 03:27:36
 Timestamp when stats for this summary period were reset =
Current count of Media Packets Lost =
                                                                      0
                                                                      0
Current count of Media Packets Dropped =
Current count of Media Packets Sent =
                                                                    236
Current count of Media Packets Received =
                                                                    236
Current count of RTCP Packets Sent =
                                                                      0
```

Current count of RTCP Packets Received =	0
Average Call Duration =	22004
Average of the Answer Seizure Ratio =	0
Average of the Round Trip Delay =	0 ms
Average of the locally calculated jitter =	0 ms
Average of the remotely calculated jitter =	0 ms
Average of the received media dropped per thousand pkts =	0
Average of the sent media lost per thousand pkts =	0
statistics for the current indefinite for adjacency phone2	
Stats Reset Timestamp:	
Timestamp when stats for this summary period were reset =	2011/03/07 03:27:36
Current count of Media Packets Lost =	0
Current count of Media Packets Dropped =	0
Current count of Media Packets Sent =	236
Current count of Media Packets Received =	236
Current count of RTCP Packets Sent =	0
Current count of RTCP Packets Received =	0
Average Call Duration =	22004
Average of the Answer Seizure Ratio =	1000
Average of the Round Trip Delay =	0 ms
Average of the locally calculated jitter =	0 ms
Average of the remotely calculated jitter =	0 ms
Average of the received media dropped per thousand pkts =	0
Average of the sent media lost per thousand pkts =	0

CAC Statistics

The CAC-related statistics have been enhanced to include the rejection counts for the CAC policies that have been implemented but failed.

A limiting field is configured in the CAC policy table entries and the CAC policy fails when there is a breach of that limit.

The following commands are used for displaying and clearing the CAC policy sets:

- The show sbc name sbe cac-policy-set [*id* [table name [entry *id*]] | global [table name [entry *id*]]] [detail] command—Lists information pertaining to the rejection counts for the failed CAC policies.
- The clear sbc *sbc-name* sbe cac-policy-set-stats [all | policy-set *cac-policy-number*] command—Clears all CAC policy statistics or clears statistics pertaining to a specified CAC policy set.

The following example shows how the **show sbc sbe cac-policy-set table entry** command displays statistic pertaining to the rejection counts for the failed CAC policies:

Router# show sbc SBC2 sbe cac-policy-set 1 table table entry 1

```
SBC Service "SBC2"
CAC Averaging period 1: 60 sec
CAC Averaging period 2: 0 sec
CAC Policy Set 1
Global policy set: Yes
Description:
First CAC table: table
First CAC scope: global
Table name: table
Description:
Table type: policy-set
Total call setup failures (due to non-media limits): 0
```

Entry 1		
CAC scope:		
CAC scope prefix length: 0		
Action: CAC complete		
Number of call setup failures (due to non-media	limits): O	
No. of registrations rejected (due to registrati	on limits): O	
Max calls per scope:	Unlimited	
No. of events rejected due to Max Call Limit:	0	
Max rog por goopo.	Imlimited	
No of events rejected due to Max Reg limit.	0	
No. of evenes rejected due to hax key fimit.	0	
Max channels per scope:	Unlimited	
Max updates per scope:	Unlimited	
Max bandwidth per scope:	Unlimited	
Ave	raging-period 1	Averaging-period
Ave	raging-period 1	Averaging-period
Ave	raging-period 1	Averaging-period
Ave Max call rate per scope:	unlimited	Averaging-period Unlimited
Ave Max call rate per scope: No. of events rejected due to Max call rate:	raging-period 1 Unlimited 0	Averaging-period Unlimited O
Ave Max call rate per scope: No. of events rejected due to Max call rate:	Unlimited	Averaging-period Unlimited O
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate:	Unlimited Unlimited	Averaging-period Unlimited 0 Unlimited
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate:	unlimited Unlimited Unlimited 0	Averaging-period Unlimited O Unlimited O
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate: Max in-call message rate:	Unlimited 0 Unlimited 0 Unlimited	Averaging-period Unlimited 0 Unlimited 0 Unlimited
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate: Max in-call message rate: No. of events rejected due to Max in-call rate:	Unlimited 0 Unlimited 0 Unlimited 0	Averaging-period Unlimited 0 Unlimited 0 Unlimited 0
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate: Max in-call message rate: No. of events rejected due to Max in-call rate:	unlimited 0 Unlimited 0 Unlimited 0 Unlimited 0	Averaging-period Unlimited 0 Unlimited 0 Unlimited 0
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate: Max in-call message rate: No. of events rejected due to Max in-call rate: Max out-call message rate:	unlimited 0 Unlimited 0 Unlimited 0 Unlimited 0 Unlimited	Averaging-period Unlimited 0 Unlimited 0 Unlimited 0 Unlimited
Ave Max call rate per scope: No. of events rejected due to Max call rate: Max reg. rate per scope: No. of events rejected due to Max reg rate: Max in-call message rate: No. of events rejected due to Max in-call rate: Max out-call message rate: No. of events rejected due to Max Out call rate:	unlimited 0 Unlimited 0 Unlimited 0 Unlimited 0 Unlimited 0	Averaging-period Unlimited 0 Unlimited 0 Unlimited 0 Unlimited 0

Timestamp when the rejection counts were last reset: 2011/03/07 04:38:24

Early media:	Allowed	Early media direction: Both
Early media timeout:	None	Transcoder per scope: Allowed
Callee Bandwidth-Field:	None	Caller Bandwidth-Field: None
Media bypass:	Allowed	Asymmetric Payload Type: Not Set
Renegotiate Strategy:		Delta
SRTP Transport:		Trusted-Only (by default)
Caller hold setting:		Standard
Callee hold setting:		Standard
Caller limited-privacy-se	ervice:	Never hide identity
Callee limited-privacy-se	ervice:	Never hide identity
Caller privacy-service:	Not set	
Callee privacy-service:	Not set	
Caller edit-privacy-reque	est: Not	t set
Callee edit-privacy-reque	est: Not	t set
Caller edit-privacy-reque	est sip s	strip: Not set
Callee edit-privacy-reque	est sip s	strip: Not set
Caller edit-privacy-reque	est sip :	insert: Not set
Callee edit-privacy-reque	est sip :	insert: Not set
Caller voice QoS profile	:	Default
Callee voice QoS profile	•	Default
Caller video QoS profile	:	Default
Callee video QoS profile	:	Default
Caller sig QoS profile:		Default
Callee sig QoS profile:		Default
Caller inbound SDP policy	<i>Y</i> :	None
Callee inbound SDP policy	/:	None
Caller outbound SDP polic	су:	None

Callee outbound SDP policy:	None
SDP Media Profile :	None
Caller Generic Stream :	Default
Callee Generic Stream :	Default
Caller media disabled:	None
Callee media disabled:	None
Caller unsignaled secure media:	Not Allowed
Callee unsignaled secure media:	Not Allowed
Caller response downgrade support	t: No
Callee response downgrade support	t: No
Caller retry rtp support:	No
Callee retry rtp support:	No
Resend sdp answer in 200ok:	No
Caller tel-event payload type:	Default
Callee tel-event pavload type:	Default
Media flag:	None
Restrict codecs to list:	Default
Restrict caller codecs to list:	Default
Restrict callee codecs to list:	Default
Codec preference list:	Default
Caller Codec profile:	None
Callee Codec profile:	None
Caller media cans list.	None
Callee media caps list:	None
TCC extra codec list.	None
Caller media-type:	Inherit (default)
Callee media-type.	Inherit (default)
Caller Media Purpage	Innerit (default)
Callee Media Bypass:	Innerit (default)
Modia Purpage Turpo.	Not got
Colleg logal transfor support.	Not Set
Manimum Call Dunabian	Inneric (deraurc)
Maximum Call Duration:	SU
Caller SRTP Support:	Innerit (default)
Callee SRTP support:	Innerit (default)
SRTP Interworking:	Inherit (default)
SRTP media interworking:	Innerit (default)
Ims rx preliminary-aar:	Disabled (default)
Ims media-service:	None(default)
media bandwidth policing:	Inherit(default)
Billing filter:	Inherit(default)
Caller ptime:	None (default)
Callee ptime:	None (default)
Caller codec variant conversion:	Disabled (default)
Callee codec variant conversion:	Disabled (default)
Caller inband DTMF mode:	Inherit(default)
Callee inband DTMF mode:	Inherit(default)
Caller Port Range Tag:	Inherit (default)
Callee Port Range Tag:	Inherit (default)
Session refresh renegotiation: In	nherit(default)

Subscriber Statistics

Subscriber-related statistics have been introduced to enable you to view and analyze information pertaining to the subscribers who are using the SBC. The **show sbc sbe** command has been enhanced to display these statistics.

The following are the features of subscriber-related statistics:

- The statistics include the subscribers who are currently registered with the SBC, that is, the subscribers stored in the SBC subscriber database. Therefore, only subscribers registered through a non-IMS Access adjacency, P-CSCF Access adjacency, or IPsec P-CSCF Access adjacency are included.
- The statistics include the individual access-side subscribers who register through the SBC. The significance of this feature is that if the SBC rewrites the register, the address of record forwarded to the registrar may be the same for two different access-side address of records. These two addresses of record are recorded as two subscribers in the statistics even though the registrar may consider this as a single subscriber. For example, if both sip:12345@192.0.2.22 and sip:12345@203.0.113.36 register through the SBC and the SBC rewrites them to sip:12345@example.registrar.com, the registrar treats this entry as a single subscriber. However, the SBC counts this entry as two subscribers.
- The statistics represent the number of registered subscribers and not the number of registered contacts. When a subscriber registers multiple contacts, that subscriber will be counted only against the source adjacency of the first contact who is registered. For example, if a subscriber registers Contact1 on Adjacency1 and then registers Contact2 on Adjacency2, the subscriber is counted only once in the global count of subscribers. The same subscriber is not included in the count of subscribers on Adjacency2. This holds true even if Contact1 expires and Contact2 remains active.
- The statistics include delegate registrations performed by the SBC.
- The statistics are available both globally and per adjacency.
- In the IMS mode, the SBC may store multiple addresses of record for a single subscriber, for example, if the registrar returns P-Associated-URIs on a REGISTER response. In this scenario, only a single subscriber is included in the statistics and not the additional addresses of record.

Restrictions for the Subscriber Statistics Feature

The following are the restrictions for the subscriber statistics feature:

- The statistics are lost in the event of a failover.
- The statistics do not include the number of fast-registered subscribers.
- If subscribers register through the SBC on an Interconnection Border Control Function (IBCF) or non-IMS adjacency, the SBC does not track these subscribers and they are, therefore, not included in the statistics.

The following command displays subscriber-related statistics:

show sbc sbc-name sbe subscriber-stats {all | dst-account name | dst-adjacency name | global |
src-account name | src-adjacency name } [current15mins | current5mins | currentday | currenthour
| currentindefinite | previous15mins | previous5mins | previousday | previoushour]

The following command resets all call-related statistics:

clear sbc *sbc-name* sbe call-stats [all | dst-account *name* | dst-adjacency *name* | global | src-account *name* | src-adjacency *name*] [all | currentindefinite]

The following example shows how the **show sbc sbe subscriber-stats** command displays statistics pertaining to subscribers:

Router# show sbc mySbc sbe subscriber-stats global currentindefinite

```
Subscribe count totals:

Active subscribers = 10

Subscriber high water mark = 15

Subscriber low water mark = 3

Stats Reset Timestamp:

Timestamp when stats for this summary period were reset = 2011/01/25 23:26:03
```

Administrative Domains

Each administrative domain represents administrative relationships with other peer entities, and can direct the SBC to use a particular number analysis and routing policy, and/or CAC policy for calls to and from the adjacency. The administrative domain is specified in admin-domain field for both SIP adjacencies and H.323 adjacencies. Any adjacencies without an administrative domain use globally configured policies.

An administrative domain has the following features:

- An administrative domain can identify policy trees that can be used for inbound or outbound number analysis, or taking a routing decision. These trees have all the attributes and capabilities of the existing number analysis and routing policy trees. The administrative domain can also identify zero or more CAC policy trees that have all the attributes and capabilities of the existing CAC policy tree.
- Users can create multiple separate policy trees for inbound number analysis, outbound number analysis, routing, and CAC. Each policy tree can be assigned to zero or more administrative domains. The user can bring each policy tree into and out of service independently from the others.
- An administrative domain can be identified by a text-based string that conveys the identity and scope of the domain.
- A signaling event can be assigned to a global administrative domain as its source or destination domain, if the classification system fails to assign it to any other source or destination administrative domain.
- Users can also assign a signaling message to multiple source and destination administrative domains. Each administrative domain is given a priority when it is assigned to an event. As per the priority given, the SBC uses the policy tree from the set of administrative domains.
- The user can assign a policy tree to administrative domain to take a routing decision for a signaling event. The user can also assign a policy tree to an administrative domain for outbound number analysis for the destination administrative domain. Changes to the policy trees can be made independently of each other. A routing decision is taken based on the policy tree chosen for the source administrative domain, but the outbound number analysis is based on the policy tree chosen for the destination administrative domain.
- All the source and destination administrative domains selected for a signaling event is provided to Billing Manager on detection points relating to that event. The XML format includes the names of the source and destination administrative domains in the billing record for a given call.

Asymmetric Payload Types

In Real-Time Transport Protocol (RTP) sessions, each codec is assigned an ID or payload type that is included in the RTP header. These payload types allow an RTP session to carry multiple formats, which may be different, concurrently. Different payload types can be assigned to the same codec in an RTP session.

If a session uses different payload types for the same codec, the session is said to be using asymmetric payload types.

SIP, H.323, and H.248 support asymmetric payload types. A SIP session negotiates the asymmetric payload types in RFC3264 Offer and Answer messages, while H.323 session negotiates the asymmetric payload types in the following messages:

- Fast Start request and response
- Open logical channel (OLC) and Open Logical Channel Acknowledgement
- Terminal Capabilities Set (applicable only to telephone-event codec)

The SBC enables seamless pass-through of asymmetric payload types in both signaling relay and media relay. Hence, an SBC can be used between two endpoints that use asymmetric payload types, without affecting the normal operations of the endpoints.

Asymmetric payload types are meant for only pass-through, and not for interworking. The SBC is not required to translate between asymmetric payload types on one leg of a call and symmetric payload types on the other leg of a call.

Prior to Cisco IOS XE Release 3.1.0S, if a SIP peer requested an asymmetric payload type, the SBC removed the codec that used the Asymmetric payload types. If no codecs were left, the entire call was torn down, as shown in Figure 7-8. From Cisco IOS XE Release 3.1.0S, the scenario illustrated in Figure 7-8 results in a successful call.



Figure 7-8 Asymmetric Payload Types – Call Teardown Scenarios

Figure 7-9 shows a scenario where two codecs are present in an Offer, but only one is present—but twice—in the Answer. In this scenario, a combination of Asymmetric payload types and a changed codec is present.



Figure 7-9 Asymmetric Payload Types – Two Codecs in Offer and One Codec in Answer

In the example illustrated in Figure 7-9, the SBC matches the Answer to the Offer, mapping the first codecs together as using Asymmetric payload types, and then discards the second set of codecs as an unsupported codec change.

Signaling

This section describes how the Asymmetric payload types feature works in the following scenarios:

- SIP-RFC3264 Offer-Answer
- H323-H245
- H.323-SIP Interworking
- Media Programming

SIP-RFC3264 Offer-Answer

This section provides details on how Asymmetric payload types works in a SIP-RFC3264 Offer-Answer scenario.

Asymmetric Payload Types:

- Communicate payload type reassignment from Answer onwards to the offerer.
- Communicate to the MEDIA asymmetric payload type bindings negotiated by RFC3264.
- Log an event when they detect an answer that changes corresponding payload type in a media relay call.

When signaling originates a codec in an offer—either the telephone-event codec in dual tone multifrequency (DTMF) interworking, or a transcoder codec in transcoding—signaling accepts a change of payload type on the answer.

Refer to RFC3264 for more details on how payload type bindings are assigned by SDP rtpmap line.

H323-H245

For H323 calls, asymmetric payload types support is available only for the telephone-event codec.

The Asymmetric Payload Type feature affects only the processing of H.245 Terminal Capability Set, in particular, the receiveRTPAudioTelephonyEventCapability, which signals the RFC2833 telephone-event payload type. This feature:

- Communicates to MEDIA the asymmetric telephone-event payload type bindings negotiated by the H.245 Terminal Capability Set.
- Generates logs when it detects a different telephone-event payload type in each direction.

H.323-SIP Interworking

For H.323-SIP interworking, asymmetric payload types support is available only for the telephone-event codec. The telephone-event payload type received in an H.245 Terminal Capability Set message is communicated onwards in an RFC3264 Offer or Answer message and vice versa.

Although H.323 does not support Asymmetric payload types for any codec other than telephone-event, the same restriction does not apply to a SIP. Hence, a SIP peer might attempt to change the payload type on a flow as part of a SIP-H.323 interworking call. If the payload type is changed, a high-severity Problem Determination log is created, and the call is discarded.

Media Programming

Signaling uses standard H.248 signaling to program asymmetric payload type streams. During the transitions between a Symmetric and Asymmetric payload type bindings, media addresses or ports are not reallocated.

Billing

The Asymmetric Payload Types feature provides the following information for billing:

- Asymmetric payload types that are in use for a given media relay call.
- Codecs that are bound to each payload type.

SIP-SIP Calls

SIP calls indicate Asymmetric payload types by indicating differing payload types in an answer to the previous offer. the SBC will then act upon these Asymmetric payload types.

See the ?\$paranum>Example: Allowing Asymmetric Payload Types? section on page 7-164 for examples of SIP/SIP configuration and Offer-Answer messages.

Configuring Asymmetric Payload Types

You can configure the SBC to allow or block Asymmetric payload types for each call. By default, asymmetric payload types are allowed on calls.

Use the **[no] payload-type asymmetric {allowed | disallowed}** command to specify whether to allow or disallow asymmetric payload types.

Performing ISSU for Asymmetric Payload Types

When performing ISSU to upgrade to Cisco IOS XE Release 3.1.0S, a call requesting Asymmetric payload types from an active SBC with a release prior to Cisco IOS XE Release 3.1.0S is replicated to a standby with Cisco IOS XE Release 3.1.0S that supports Asymmetric payload types as if Symmetric Payload Types are being used. The media may not flow correctly on the primary or the backup after the failover.

If a call is currently using Symmetric payload types on an active SBC that does not support Asymmetric payload types, during attempts to renegotiate using Asymmetric payload types, one of the following occurs:

- If the Media, media forwarding component, or the Media Gateway detect that the active SBC does not support Asymmetric payload types, then the change to the corresponding call may be rejected and the call will remain unchanged.
- If the Media, media forwarding component, or Media Gateway does not detect that the active SBC does not support Asymmetric payload types, the corresponding call may continue as if using Symmetric payload types, and this may result in media not flowing correctly.

If a call changes the payload type from Symmetric to Asymmetric, or vice versa:

- After a gate is defined as Asymmetric, it remains Asymmetric even if it ceases to use Asymmetric payload types as a result of a renegotiation.
- If a Symmetric gate is marked as Asymmetric, and the partner does not support Asymmetric payload types, the gate is no longer replicated. The gate is deleted from the backup partner.
How to Implement Policies

How to Implement Policies

Cisco Unified Border Element (SP Edition) policies are configured and activated as described in the following sections:

- Configuring Number Analysis Tables
- Configuring Administrative Domain
- Configuring Default Call Policy Set
- Configuring Routing Tables
- Configuring Number Manipulation
- Configuring Hunting
- Configuring H.323 MultiARQ Hunting
- Configuring Call Admission Control Policy Sets, CAC Tables, and Global CAC Policy Sets
- Configuring Privacy Service
- Configuring Multiple SBC Media Bypass
- Configuring Common IP Address Media Bypass
- Activating a CAC Policy Set

Configuring Number Analysis Tables

This task configures a number analysis table. The types of number analysis configuration are described in the following sections:

- Configuring Number Validation
- Configuring Number Categorization
- Configuring Text Address Validation and Source Address Manipulation

Configuring Number Validation

This task configures number validation for a number analysis table.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. first-inbound-na-table table-name
- 6. na-dst-prefix-table table-name
- 7. entry entry-id
- 8. match-prefix key
- 9. action [next-table goto-table-name | accept | reject]
- **10.** category category-name

- 11. entry entry-id
- **12.** edit [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- **13.** edit-cic [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- 14. match-prefix key
- **15.** action [next-table goto-table-name | accept | reject]
- **16.** category category-name
- 17. entry entry-id
- **18.** match-prefix key
- **19.** action [next-table goto-table-name | accept | reject]
- **20.** category category-name
- 21. exit
- 22. exit
- 23. end
- 24. show

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# sbc mySbc Router(config-sbc)#	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe Router(config-sbc-sbe)#	
Step 4	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# call-policy-set 1 Router(config-sbc-sbe-rtgpolicy)#	
Step 5	first-inbound-na-table table-name	Configures the name of the first policy table to process when performing the number analysis stage of policy.
	Example: Router(config-sbc-sbe-rtgpolicy)# first-inbound-na-table hotel_table	

	Command or Action	Purpose
Step 6	na-dst-prefix-table table-name	Enters the mode for configuring a number analysis table whose entries match the prefix (the first several digits) of the dialed number within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-rtgpolicy)# na-dst-prefix-table hotel_table	Commands for other number analysis tables:
		• na-carrier-id-table —This table requires additional commands match-cic and edit-cic (see below)
		• na-dst-address-table
		• na-src-address-table
		• na-src-prefix-table
		• na-src-account-table
		• na-src-adjacency-table
		• na-carrier-id-table
Step 7	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy-natable)# entry 1	
Step 8	match-prefix key / match-cic cic	Configures the match value of an entry in the number analysis table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# match-prefix XXX</pre>	• The match-prefix <i>key</i> argument is a string used to match the prefix (the starting part) of the dialed number.
		• The match-cic <i>cic</i> argument is used with the na-carrier-id-table command and configures the match carrier ID code in a table whose entries match a carrier ID.
Step 9	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)#	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 10	category category-name	Configures the category of an entry in the number analysis table.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# category external	

	Command or Action	Purpose
Step 11	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# entry 2</pre>	
Step 12	<pre>edit [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]</pre>	Configures a dial-string manipulation action in a number analysis table. You are not allowed to do this if the table is part of the active policy set.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# odit_dol_profix_1</pre>	The no version of the command deletes the edit action of the given entry in the routing table.
	euro del pretix i	The edit command can be set to the following values:
		• del-prefix <i>pd</i> —Delete prefix <i>pd</i> , where <i>pd</i> is a positive integer specifying a number of digits to delete from the front of the dialed string.
		• del-suffix <i>sd</i> —Delete suffix <i>sd</i> , where <i>sd</i> is a positive integer specifying a number of digits to delete from the end of the dialed string.
		• add-prefix <i>pa</i> —Add prefix <i>pa</i> , where <i>pa</i> is a string of digits to add to the front of the dialed string.
		• replace <i>ds</i> —Replace <i>ds</i> , where <i>ds</i> is a string of digits that replaces the dialed string.
		In the example to the left, the edit command sets entry 2 to delete 1 digit from the beginning of the dialed string in the number analysis table.

	Command or Action	Purpose
Step 13	edit-cic [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]	Configures a carrier identification code (CIC) manipulation action in a number analysis table.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# edit-cic del-prefix 1	You are not allowed to do this if the table is part of the active policy set.
		• del-prefix <i>pd</i> : A positive integer specifying a number of digits to delete from the front of the carrier ID string.
		• del-suffix <i>sd</i> : A positive integer specifying a number of digits to delete from the end of the carrier ID string.
		• add-prefix <i>pa</i> : A string of digits to add to the front of the carrier ID string.
		• replace <i>ds</i> : A string of digits to replace the carrier ID string with.
		The "edit-cic del-prefix 1" command sets entry 2 to delete the first digit of the carrier ID in the current number analysis table.
		You can remove the CIC or carrier ID from outbound messages by specifying a replacement string of 0000 or by specifying a prefix deletion length of 4.
		For example:
		edit-cic del-prefix 4 OR edit-cic replace 0000
Step 14	match-prefix key	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match the start of the dialed number.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# match-prefix 9XXX</pre>	The no version of the command destroys the match value.
Step 15	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# action_accept</pre>	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 16	category category-name	Configures the category of an entry in the number analysis table.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# category external	

	Command or Action	Purpose
Step 17	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# entry 3</pre>	
Step 18	<pre>match-prefix key Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# matrix find only the set find only the se</pre>	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match the start of the dialed number.
Step 19	action [next-table goto-table-name accept reject]	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)#</pre>	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
	action accept	• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 20	category category-name	Configures the category of an entry in the number analysis table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# category bar</pre>	
Step 21	exit	Exits from the entry mode to the natable mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# exit</pre>	
Step 22	exit	Exits from the natable mode to the callpolicy mode.
	Example: Router(config-sbc-sbe-rtgpolicy-natable)# exit	
Step 23	end	Exits the callpolicy mode to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-rtgpolicy-natable)# end	
Step 24	show	Displays the current configuration information.
	Example: Router(config-sbc-sbe-rtgpolicy)# show	

Configuring Number Categorization

This task configures number categorization for a number analysis table.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. first-inbound-na-table table-name
- 6. na-src-account-table table-name
- 7. entry entry-id
- 8. match-account key
- 9. action [next-table goto-table-name | accept | reject]
- **10.** entry entry-id
- **11. match-account** *key*
- **12.** action [next-table goto-table-name | accept | reject]
- **13.** entry *entry-id*
- 14. match-account key
- **15.** action [next-table goto-table-name | accept | reject]
- **16. na-dst-prefix-table** *table-name*
- 17. entry entry-id
- **18.** match-prefix key
- **19.** category category-name
- **20.** action [next-table goto-table-name | accept | reject]
- **21.** entry *entry-id*
- 22. match-prefix key
- **23.** category category-name
- 24. action [next-table goto-table-name | accept | reject]
- 25. end
- 26. show

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mySbc Router(config-sbc)#	• <i>sbc-name</i> —Name of the SBC.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe Router(config-sbc-sbe)#	
Step 4	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# call-policy-set 1 Router(config-sbc-sbe-rtgpolicy)#	
Step 5	first-inbound-na-table table-name	Configures the name of the first policy table to process when performing the number analysis stage of policy.
	Example: Router(config-sbc-sbe-rtgpolicy)# first-inbound-na-table check_account	
Step 6	na-src-account-table table-name	Enters the mode for configuring a number analysis table within the context of an SBE policy set with the entries of
	<pre>Example: Router(config-sbc-sbe-rtgpolicy)# na-src-account-table check_account Router(config-sbc-sbe-rtgpolicy- natable)#</pre>	the table matching the source account.
Step 7	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy- natable)# entry 1 Router(config-sbc-sbe-rtgpolicy- natable-entry)#	
Step 8	<pre>match-account key Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# match-account hotel_foo</pre>	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match the source account.

	Command or Action	Purpose
Step 9	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy-	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
	hotel_dialing_plan	• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 10	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# entry 2</pre>	
Step 11	<pre>match-account key Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# match-account hotel bar</pre>	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match the source account.
Step 12	action [next-table goto-table-name accept reject]	Configures the action of an entry in a number analysis table. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy-	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
	hotel_dialing_plan	• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 13	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# entry 3</pre>	
Step 14	match-account key	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match
	<pre>Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# match-account internal</pre>	

	Command or Action	Purpose
Step 15	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# action accent	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 16	na-dst-prefix-table table-name	Enters the mode for configuring a number analysis table within the context of an SBE policy set with the entries of the table matching the start of the dialed number.
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# na-dst-prefix-table hotel_dialing_plan	
Step 17	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# entry 1</pre>	
Step 18	match-prefix key	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match the start of the dialed number.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# match-prefix XXX</pre>	
Step 19	category category-name	Specifies the category of an entry in a number analysis table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# category internal_call</pre>	
Step 20	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# action accept</pre>	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 21	entry entry-id	Enters the mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# entry 2</pre>	

	Command or Action	Purpose
Step 22	match-prefix key	Configures the match value of an entry in the number analysis table. The <i>key</i> argument is a string used to match
	Example:	the start of the dialed number.
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# match-prefix 9XXX	
Step 23	category category-name	Specifies the category of an entry in a number analysis table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# category external_call</pre>	
Step 24	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# action_accont</pre>	• Configure the name of the next number analysis table to process if the event matches this entry using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table using the reject keyword.
Step 25	end	Exits from the entry mode and returns to Privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# end	
Step 26	show	Displays the current configuration information.
	Example:	
	Router(config-sbc-sbe-rtgpolicy)# show	

Configuring Text Address Validation and Source Address Manipulation

This task shows how to configure text address validation and source address manipulation for a number analysis table.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. first-inbound-na-table table-name
- 6. na-dst-address-table table-name
- 7. entry entry-id
- 8. action [next-table goto-table-name | accept | reject]

- **9.** edit-src [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- **10.** match-address *key* [regex | digits]
- 11. entry entry-id
- **12.** action [next-table goto-table-name | accept | reject]
- **13.** edit-src [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- 14. match-address key [regex | digits]
- 15. exit
- 16. exit
- 17. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mySbc Router(config-sbc)#	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe Router(config-sbc-sbe)#	
Step 4	call-policy-set policy-set-id	Enters the routing policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# call-policy-set 1 Router(config-sbc-sbe-rtgpolicy)#	
Step 5	first-inbound-na-table table-name	Configures the name of the first policy table to be processed when performing the number analysis stage of the policy.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy)# first-inbound-na-table hotel_table</pre>	

	Command or Action	Purpose
Step 6	<pre> na-dst-address-table table-name Example: Router(config-shc-she-rtgpolicy)# </pre>	Enters the number analysis table mode for configuring a number analysis table whose entries match the prefix (the first few digits) of the dialed number within the context of an SBE policy set.
	na-dst-address-table room_table	The commands for other number analysis tables are:
		• na-carrier-id-table (This table requires additional commands— match-cic and edit-cic)
		• na-dst-address-table
		• na-src-address-table
		• na-src-prefix-table
		• na-src-account-table
		na-src-adjacency-table
		• na-carrier-id-table
Step 7	entry entry-id Example:	Enters the number analysis table entry mode for configuring an entry in a number analysis table, creating the entry, if necessary.
	Router(config-sbc-sbe-rtgpolicy-natable)# entry 1	
Step 8	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# action accept</pre>	• Configure the name of the next number analysis table to be processed if the event matches this entry, using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table, using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table, using the reject keyword.
Step 9	edit-src [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]	Configures the source address manipulation action in the NA table.
	Freedo	This cannot be done if a table is part of the active policy set.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)#	The no version of the command removes the match value.
	edit-src del-prefix 3	• del-prefix <i>pd</i> : A positive integer specifying the number of digits to be delete from the front of the carrier ID string.
		• del-suffix <i>sd</i> : A positive integer specifying the number of digits to be deleted from the end of the carrier ID string.
		• add-prefix <i>pa</i> : A string of digits to be added to the front of the carrier ID string.
		• replace <i>ds</i> : A string of digits to replace the carrier ID string with.

	Command or Action	Purpose
Step 10	<pre>match-address key [regex digits]</pre>	Configures the match value of an entry in an NA table.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)#	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table, and include a textual value in the match-address field.
	match-address 123456 digits	The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if the address contains any character other than 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the X in the routing tables is treated not as a wildcard character, but as a literal X. For example, the match value of X matches the username X, but not A.
		Note A direct string comparison is not done by NA. To compare a fixed string, a regex without any regex meta-characters can be used.
Step 11	entry entry-id	Enters the number analysis table entry mode for configuring an entry in a number analysis table, creating the entry, if
	Example:	necessary.
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# entry 2	
Step 12	<pre>action [next-table goto-table-name accept reject]</pre>	Configures the action of an entry in a number analysis table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# action accept</pre>	• Configure the name of the next number analysis table to be processed if the event matches this entry, using the next-table keyword and the <i>goto-table-name</i> argument.
		• Configure the call to be accepted if it matches the entry in the table, using the accept keyword.
		• Configure the call to be rejected if it matches the entry in the table, using the reject keyword.

	Command or Action	Purpose
Step 13	edit-src [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]	Configures the source address manipulation action in the NA table.
	Example:	This cannot be done if the table is a part of the active policy set.
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# edit-src del-suffix 1	The no version of the command destroys the match value.
		• del-prefix <i>pd</i> : A positive integer specifying the number of digits to be deleted from the front of the carrier ID string.
		• del-suffix <i>sd</i> : A positive integer specifying the number of digits to be deleted from the end of the carrier ID string.
		• add-prefix <i>pa</i> : A string of digits to be added to the front of the carrier ID string.
		• replace <i>ds</i> : A string of digits to be replaced the carrier ID string with.
Step 14	<pre>match-address key [regex digits]</pre>	Configures the match value of an entry in an NA table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# match-address ^.* regex</pre>	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table, and include a textual value in the match-address field.
		The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if the address contains any character other than 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the X in the routing tables is treated not as a wildcard character, but as a literal X . For example, the match value of X matches the username X , but not A .
		Note A direct string comparison is not done by NA. To compare a fixed string, a regex without any regex meta-characters can be used.
Step 15	exit	Exits from the entry mode and enters the natable mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# exit</pre>	
Step 16	exit	Exits from the natable mode and enters the call policy mode.
	Example: Router(config-sbc-sbe-rtgpolicy-natable)# exit	
Step 17	end	Exits the call policy mode and enters the Privileged EXEC mode.
	Example: Router(config-sbc-sbe-rtgpolicy)# end	

Configuring Administrative Domain



This task configures an administrative domain.

The policy sets must be in a complete state before they are assigned to an administrative domain. A default call-policy-set must be configured before the administrative domain mode is entered. If an inbound NA set, a routing set, or an outbound NA set is undefined, the administrative domain uses the values defined within the default call-policy-set.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. admin-domain name
- 5. description [line]
- 6. call-policy-set {inbound-na number | outbound-na number | rtg number } [priority priority-value]
- 7. cac-policy-set number
- 8. exit
- 9. adjacency sip | h323 adjacency-name
- 10. admin-domain name
- 11. end
- 12. show sbc *sbc-name* sbe admin-domain [adjacency]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
		• <i>sbc-name</i> —Defines the name of the SBC service.
	Example:	
	Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example:	
	Router(config-sbc)# sbe	
Step 4	admin-domain name	Enters the mode of an administrative domain.
		• <i>name</i> —Defines the administrative domain name that
	Example:	can be of 30 characters maximum.
	Router(config-sbc-sbe)# admin-domain Domain1	

	Command or Action	Purpose
Step 5	description [line]	Assigns a text description to the administrative domain.
	Example: Router(config-sbc-sbe-ad)# description This is	• <i>line</i> —Describes the administrative domain.
Step 6	a description of DOMAINI call-policy-set {inbound-na number outbound-na number rtg number} [priority priority-value] Example:	Configures a single call-policy-set or separate call-policy-sets for routing, inbound number analysis, and outbound number analysis. The policy sets must be in a complete state before they can be assigned to the policy set of an administrative domain.
	Router(config-sbc-sbe-ad)# call-policy-set rtg 2 priority 1 Router(config-sbc-sbe-ad)# call-policy-set inbound-na 2 priority 1 Router(config-sbc-sbe-ad)# call-policy-set outbound-na 2 priority 1	Note Specifying an inbound NA, a routing, or an outbound NA policy set is optional. If the policy sets are undefined, the admin-domain uses the values defined within the default call policy set.
		• inbound-na —Specifies the inbound number analysis policy
		• outbound-na —Specifies the outbound number analysis policy
		• rtg —Specifies the routing policy
		• priority —Specifies the priority of a policy-set.
		• <i>number</i> —An unique identifier for the policy set. The value can range from 1 to 2147483647.
		• <i>priority-value</i> —The priority value ranging from 1 to 10 where 10 indicates the highest priority. By default, the priority is set to 10.
		Note Priority is required because more than one administrative domain can be specified on an adjacency. The SBC uses the policy-set with the highest priority.
Step 7	cac-policy-set number	Configures the cac-policy-set in an administrative domain. Only one cac-policy-set can be specified.
	Example: Router(config-sbc-sbe-ad)# cac-policy-set 2	The policy sets must be in a complete state before they can be assigned to the policy set of an administrative domain.
		• <i>number</i> —An unique identifier for the policy set. The value can range from 1 to 2147483647.
Step 8	exit	Exits the administrative domain mode and enters the SBE mode.
	<pre>Example: Router(config-sbc-sbe-ad)# exit</pre>	
Step 9	adjacency sip h323 adjacency-name	Enters the mode of an SBE SIP or H.323 adjacency.
	Example:	• adjacency-name—Defines the name of the SIP or H.323 adjacency.
	Router(config-sbc-sbe)# adjacency sip sipadj	Note The H323 adjacency must be unattached to add, delete, or modify the admin-domain command.

	Command or Action	Purpose
Step 10	admin-domain name	Configures an administrative domain on an adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# admin-domain Domain1	• <i>name</i> —Defines the administrative domain name that can be of 30 characters maximum.
Step 11	end	Exits the SBE SIP or H.323 adjacency mode and enters the Privilege exec mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 12	show sbc <i>sbc-name</i> sbe admin-domain [adjacency]	Displays details of administrative domains configured on the SBC.
	Example:	• <i>sbc-name</i> —Defines the name of the SBC service.
	Router# show sbc MySBC sbe admin-domain	• adjacency —Lists the administrative domains per adjacency.

The following example shows the output of the show sbc sbe admin-domain command:

```
Router# show sbc mySBC sbe admin-domain

SBC Service "mySBC"

Global cac-policy-set: 2

Default call-policy-set/priority: 1/6

cac call-policy-set/priority

Administrative Domain policy-set inbound-na routing outbound-na

DOMAIN1 2 2/1 2/1 2/1
```

The following example shows the output of the **show sbc sbe admin-domain adjacency** command: Router# **show sbc mySBC sbe admin-domain adjacency** SBC Service "mySBC"

Adjacency Name	Туре	State	Admin-domain
SIPP1A	SIP	Attached	DOMAIN1

Configuring Default Call Policy Set

This task configures a call-policy-set and sets a priority for the SBC to determine the default policy set to use when the administrative domain is not present.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. first-inbound-na-table word
- 6. first-outbound-na-table word
- 7. complete

- 8. exit
- 9. call-policy-set default *policy-set-id* [priority *priority*]
- 10. end
- 11. show sbc *sbc-name* sbe call-policy-set [default]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc_mySbc	• <i>sbc-name</i> —The name of the SBC service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set policy-set-id	Creates a new call policy set and enters SBE routing policy configuration mode.
	Example: Router(config-sbc-sbe)# call-policy-set 25	• policy-set-id—The call policy set number that can range from 1 to 2147483647.
Step 5	first-inbound-na-table word	Specifies the first inbound number analysis table.
	Example: Router(config-sbc-sbe-rtgpolicy)# first-inbound-na-table InTable	• <i>word</i> —Inbound number analysis table name. The table length can be of 30 characters maximum.
Step 6	first-outbound-na-table word	Specifies the first outbound number analysis table.
	Example: Router(config-sbc-sbe-rtgpolicy)# first-outbound-na-table InTable	• <i>word</i> —Outbound number analysis table name. The table length can be of 30 characters maximum.
Step 7	complete	Completes the call-policy set after committing the full set.
	Example: Router(config-sbc-sbe-rtgpolicy)# complete	
Step 8	exit	Exits the SBE routing policy mode.
	Example: Router(config-sbc-sbe-rtgpolicy)# exit	

	Command or Action	Purpose	
Step 9	call-policy-set default <i>policy-set-id</i> [priority <i>priority</i>]	Assigns the default call-policy-set id when an administrative domain is not specified on the adjacency or the specified administrative domain does not exist.	
	<pre>Example: Router(config-sbc-sbe)# call-policy-set default 25 priority 1</pre>	• <i>policy-set-id</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.	
		• priority —Specifies the priority to determine which active call-policy-set to use. The SBC uses the policy set with the highest priority.	
		• <i>priority</i> — The priority value ranging from 1 to 10 with 10 indicating highest priority. By default, priority is set to 6.	
		Note A default call-policy-set must be configured before the user enters the administrative domain mode. If an inbound NA set, a routing set, or an outbound NA set is undefined, the administrative domain uses the values defined within the default call-policy-set.	
Step 10	end	Exits the SBE mode and enters the Privilege exec mode.	
	Example: Router(config-sbc-sbe)# end		
Step 11	<pre>show sbc sbc-name sbe call-policy-set [default]</pre>	Displays details of the call policy sets configured on the SBC.	
	Example:	• <i>sbc-name</i> —Defines the name of the SBC service.	
	Router# show sbc mySBC sbe call-policy-set	• default —Lists the information pertaining to the default call policy set.	

The following example shows the output of the show sbc sbe call-policy-set command:

Router# show sbc mySBC sbe call-policy-set

SBC Service "mySBC"		
Policy set 1 Default policy set First inbound NA table First call routing table First reg routing table First outbound NA table	: Yes (priority 6) : : TAB1 : TAB2 :	
Table Name Class Table type Total Call-policy Failures Entry Match Value	: TAB1 : Routing : rtg-src-adj : 0 (0 *) Destination Adjacency	Action
1 SIPP1A 2 SIPP1B	SIPP1B SIPP1A	Routing complete Routing complete
Table Name Class	: TAB2 : Routing	

```
Table type
                            : rtg-src-adj
   Total Call-policy Failures : 0 (0 *)
   Entry Match Value Destination Adjacency Action
   ____
             _____
                                 ----- -----
            SIPP1A
                               Registrar Routing complete
Registrar Routing complete
   1
   2
            SIPP1B
Policy set 2
 First call routing table : TAB1
First outbound NA table : TAB2
First call routing table : TAB2
First outbound NA table :
 Table Name
                            : TAB1
   Class
                : rtg-src-adj
                           : Routing
   Table type
   Total Call-policy Failures : 0 (0 *)
   Entry Match Value Destination Adjacency Action
            SIPP1A
SIPD1B
   ____
                                 _____
                                                       ____
                                SIPP1B
   1
                                                      Routing complete
   2
                                SIPP1A
            SIPP1B
                                                       Routing complete
  Table Name
                           : TAB2
                         : Routing
: rtg-src-adj
   Class
   Table type
   Total Call-policy Failures : 0 (0 *)
   Entry Match Value Destination Adjacency Action
             _____
   ____
                                 _____
                                                       ____
                       Registrar
Registrar
        SIPP1A
SIPP1B
                                                     Routing complete
   1
   2
                                                      Routing complete
Policy set 25
 Default policy set : No
First inbound NA table : ADMINTable
First call routing table :
 First reg routing table
                            :
 First outbound NA table
                            : OutTable
```

* Numbers in brackets refer to a call being rejected by a routing or number analysis table because there were no matching entries in the table. This is also included in the total figure.

The following example shows the output of the show sbc sbe call-policy-set default command:

```
Router# show sbc mySBC sbe call-policy-set default
```

SBC Service "mySBC"		
Policy set 1		
Default policy set	: Yes (priority 6)	
First inbound NA table	:	
First call routing table	: TAB1	
First reg routing table	: TAB2	
First outbound NA table	:	
Table Name	: TAB1	
Class	: Routing	
Table type	: rtg-src-adj	
Total Call-policy Failures	: 0 (0 *)	
Entry Match Value	Destination Adjacency	Action
1 SIPP1A	SIPP1B	Routing complete

```
2
          SIPP1B
                             SIPP1A
                                                Routing complete
                        : TAB2
Table Name
 Class
                       : Routing
 Table type
                       : rtg-src-adj
 Total Call-policy Failures : 0 (0 *)
 Entry Match Value Destination Adjacency Action
          _____
                            ----- -----
 ____
 1
          SIPP1A
                            Registrar
                                                Routing complete
 2
          SIPP1B
                            Registrar
                                               Routing complete
```

* Numbers in brackets refer to a call being rejected by a routing or number analysis table because there were no matching entries in the table. This is also included in the total figure.

Configuring Routing Tables

See the following sections:

- Configuring a Destination Address Table, page 7-86
- Configuring the Destination, Source Domain, and Carrier ID Tables, page 7-92
- Configuring Number Manipulation, page 7-104
- Configuring the Least Cost Table, page 7-97
- Configuring Time-Based Tables, page 7-99
- Configuring Trunk-Group ID Tables, page 7-101

Configuring a Destination Address Table

This task configures a dst-address routing table.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. first-call-routing-table table-name
- 6. rtg-dst-address-table table-name
- 7. entry entry-id
- 8. match-address key [regex | string | digits]
- 9. prefix
- 10. dst-adjacency target-adjacency
- 11. action [next-table goto-table-name | complete | reject]
- 12. exit
- 13. entry entry-id
- 14. match-address key [regex | string | digits]

- 15. prefix
- **16. dst-adjacency** *target-adjacency*
- **17.** action [next-table goto-table-name | complete | reject]
- 18. exit
- 19. entry entry-id
- 20. match-address key [regex | string | digits]
- 21. prefix
- 22. dst-adjacency target-adjacency
- 23. action [next-table goto-table-name | complete | reject]
- 24. exit
- 25. entry entry-id
- 26. match-address key [regex | string | digits]
- 27. prefix
- **28.** dst-adjacency target-adjacency
- **29.** action [next-table goto-table-name | complete | reject]
- 30. exit
- 31. complete name
- 32. end
- 33. show

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	

	Command or Action	Purpose
Step 5	<pre>first-call-routing-table table-name Example: Router(config-sbc-sbe-rtgpolicy)# first-call-routing-table ROUTE-ON-DEST-NUM</pre>	Configures the name of the first policy table to process when performing the routing stage of policy for new-call events.
Step 6	<pre>rtg-dst-address-table table-name Example: Router(config-sbc-sbe-rtgpolicy)# rtg-dst-address-table MyRtgTable</pre>	Enters the configuration mode of a routing table within the context of an SBE policy set with the entries of the table matching the dialed number (after number analysis).
Step 7	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	Example:	
Step 8	match-address key [regex string digits]	Configures the match value of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 334	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table and put a textual value in the match-address field.
		The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if it contains any character other than: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the "X" in the routing tables is treated not as a wildcard character, but as a literal "X". For example, the match value of "X" matches the username "X", but not "A".
Step 9	prefix	Configures the match-address of this entry to match the start of the destination address.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# prefix</pre>	
Step 10	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW2	

	Command or Action	Purpose
Step 11	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 12	exit	Exits the entry mode to the rtgtable mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit</pre>	
Step 13	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2</pre>	
Step 14	<pre>match-address key [regex string digits]</pre>	Configures the match value of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 434	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table and put a textual value in the match-address field.
		The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if it contains any character other than: 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the "X" in the routing tables is treated not as a wildcard character, but as a literal "X". For example, the match value of "X" matches the username "X", but not "A".
Step 15	prefix	Configures the match-address of this entry to match the start of the destination address.
	Example:	
	Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# prefix	
Step 16	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	Example:	
	Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW1	

	Command or Action	Purpose
Step 17	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 18	exit	Exits the entry mode to the rtgtable mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit</pre>	
Step 19	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 3</pre>	
Step 20	<pre>match-address key [regex string digits]</pre>	Configures the match value of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 354	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table and put a textual value in the match-address field.
		The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if it contains any character other than: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the "X" in the routing tables is treated not as a wildcard character, but as a literal "X". For example, the match value of "X" matches the username "X", but not "A".
Step 21	prefix	Configures the match-address of this entry to match the start of the destination address.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgpolicy-rtgtable -entry)# prefix</pre>	
Step 22	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency H323-AS540-PSTN-GW2</pre>	

	Command or Action	Purpose
Step 23	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete</pre>	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 24	exit	Exits the entry mode to the rtgtable mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit</pre>	
Step 25	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 4</pre>	
Step 26	<pre>match-address key [regex string digits]</pre>	Configures the match value of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 454	To create a routing table that routes on user name, use the existing rtg-dst-address-table or rtg-src-address-table and put a textual value in the match-address field.
		The SBC skips number analysis and performs only routing when the SIP message contains a user name. The SBC decides that an address is a user name (as opposed to a phone number) if it contains any character other than: 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, plus, hyphen, period, open-round-bracket, and close-round-bracket.
		When the SBC has decided that an address is a user name, the "X" in the routing tables is treated not as a wildcard character, but as a literal "X". For example, the match value of "X" matches the username "X", but not "A".
Step 27	prefix	Configures the match-address of this entry to match the start of the destination address.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# prefix</pre>	
Step 28	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)#	

	Command or Action	Purpose
Step 29	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete</pre>	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
		• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 30	exit	Exits the entry mode to the rtgtable mode.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit	
Step 31	complete name	Completes the full routing policy set when you have committed the full set.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# complete	
Step 32	end	Exits rtgtable mode and enters Privileged Exec mode.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# end	
Step 33	show	Displays the current configuration information.
	Example: Router# show	

Configuring the Destination, Source Domain, and Carrier ID Tables

This task configures dst-domain and src-domain and carrier ID routing tables.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. **rtg-src-domain-table** *table-name* | **rtg-dst-domain-table** *table-name* | **rtg-carrier-id-table** *table-name*
- 6. entry entry-id
- 7. match-domain key [regex] | match-cic cic
- 8. edit action

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- **9.** edit-cic [del-prefix *pd*] | [del-suffix *sd*] | [add-prefix *pa*] | [replace *ds*]
- **10.** action [next-table goto-table-name | complete | reject]
- **11. dst-adjacency** *target-adjacency*
- 12. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
Stor 4	Example: Router(config-sbc)# sbe	
Slep 4	call-policy-set policy-set-la	an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	
Step 5	<pre>rtg-src-domain-table table-name rtg-dst-domain-table table-name rtg-carrier-id-table table-name</pre>	Enters the configuration mode of a routing table (creating a new table if necessary) whose entries match the source or destination domains, or carrier ID respectively.
	Example:	You are not allowed to enter the submode of routing table configuration in the context of the active policy set.
	rtg-src-domain-table MyRtgTable	The no version of the command destroys the routing table. A routing table may not be destroyed if it is in the context of the active policy set.
Step 6	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable)# entry 1	<i>entry-id</i> is a number that uniquely identifies an entry in the newly created routing table.
Step 7	<pre>match-domain key [regex] / match-cic cic</pre>	Creates or modifies the matching domain or carrier id code (CIC) of an entry in a routing table.
	Example:	• <i>key</i> is regular expression, not just a string.
	Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# match-domain ^cisco.com\$	• <i>cic</i> is the carrier ID that matches the entry in a routing table.

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	Command or Action	Purpose
Step 8	edit action	Configures a dial-string manipulation action in the routing table. You are not allowed to do this if the table is part of the active policy set.
	Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# edit del-prefix 1	The no version of the command deletes the edit action of the given entry in the routing table.
		The edit command can be set to the following values:
		• del-prefix <i>pd</i> —Delete prefix <i>pd</i> , where <i>pd</i> is a positive integer specifying a number of digits to delete from the front of the dialed digit string.
		• del-suffix <i>sd</i> —Delete suffix <i>sd</i> , where <i>sd</i> is a positive integer specifying a number of digits to delete from the end of the dialed digit string.
		• add-prefix <i>pa</i> —Add prefix <i>pa</i> , where <i>pa</i> is a string of digits to add to the front of the dialed string.
		• replace <i>ds</i> —Replace <i>ds</i> , where <i>ds</i> is a string of digits that replaces the dialed string.
		In the example to the left, the edit command sets entry 1 to delete 1 digit from the beginning of the dialed string in the routing table "MyRtgTable".
Step 9	<pre>edit-cic [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]</pre>	Configures a carrier identification code (CIC) manipulation action in any routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy- natable-entry)# edit-cic del-prefix 1</pre>	You are not allowed to do this if the table is part of the active policy set.
		• del-prefix <i>pd</i> : A positive integer specifying a number of digits to delete from the front of the carrier ID string.
		• del-suffix <i>sd</i> : A positive integer specifying a number of digits to delete from the end of the carrier ID string.
		• add-prefix <i>pa</i> : A string of digits to add to the front of the carrier ID string.
		• replace <i>ds</i> : A string of digits to replace the carrier ID string with.
		The following command sets entry 2 to delete the first digit of the carrier ID in the current routing table.
		If you wish to remove the carrier ID entirely from outgoing messages, you should specify a replacement string of 0000 or a prefix deletion length of 4. For example,
		edit-cic del-prefix 4 OR edit-cic replace 0000

	Command or Action	Purpose
Step 10	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# action complete	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
	regraphe energy# accion complete	• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
Step 11	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	Example:	
	Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW2	
Step 12	exit	Exits the current mode of the configuration.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# exit	

Configuring the Category Table

This task configures dst-domain and src-domain and carrier ID routing tables.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. rtg-category-table table-name
- 6. entry entry-id
- 7. match-category word
- 8. action [next-table goto-table-name | complete | reject]
- 9. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set <i>policy-set-id</i>	Enters the mode of routing policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	
Step 5	rtg-category-table table-name	Enters the submode of configuration of a routing table whose entries match on the category within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-rtgpolicy)# rtg-category-table MyRtgTable	
Step 6	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable)# entry 1	<i>entry-id</i> is a number that uniquely identifies an entry in the newly created routing table.
Step 7	match-category word	Configures the match value of an entry in a routing table matching on the category.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-category emergency\$</pre>	
Step 8	<pre>action [next-table goto-table-name complete reject]</pre>	If any calls match the criterion, they are rejected.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action reject	
Step 9	exit	Exits the current mode of the configuration.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit</pre>	

Configuring the Least Cost Table

This task configures a Least Cost routing table.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. rtg-least-cost-table table-name
- 6. entry entry-id
- 7. cost cost
- 8. dst-adjacency
- 9. action complete
- 10. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	
Step 5	<pre>rtg-least-cost-table table-name</pre>	Enters the submode of configuration of a routing table whose entries match on the least cost within the context of
	Example: Router(config-sbc-sbe-rtgpolicy)# rtg-least-cost-table MyRtgTable	an SBE policy set.

	Command or Action	Purpose
Step 6	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable)# entry 1	<i>entry-id</i> is a number that uniquely identifies an entry in the newly created routing table.
Step 7	cost cost	Assigns a cost to the route.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # cost 50\$	
Step 8	dst-adjacency target-adjacency	Configures the destination adjacency of an entry in a routing table.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW2	
Step 9	action complete	Specifies that routing is complete when an entry matches this policy
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# action complete	
Step 10	exit	Exits the current mode of the configuration.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# exit	

Configuring Time-Based Tables

This task configures dst-domain and src-domain and carrier ID routing tables.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. rtg-time-table table-name
- 6. entry entry-id
- 7. match-time {[date yr year_low year_high mon month_low month_high day date_low date_high] [dow DoW_low DoW_high] [tod hr hour_low hour_high min minute_low minute_high]}
- 8. precedence precedence
- 9. dst-adjacency dst_adj
- **10.** action [next-table goto-table-name | complete | reject]
- 11. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	
Step 5	<pre>rtg-time-table table-name</pre>	Enters the submode of configuration of a routing table whose entries match on the time within the context of an
	Example:	SBE policy set.
	Router(config-sbc-sbe-rtgpolicy)# rtg-time-table MyRtgTable	

	Command or Action	Purpose
Step 6	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable)# entry 1	<i>entry-id</i> is a number that uniquely identifies an entry in the newly created routing table.
Step 7	<pre>match-time {[date yr year_low year_high mon month_low month_high day date_low date_high] [dow DoW_low DoW_high] [tod hr hour_low hour_high min minute_low minute_high]) Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time date yr 2006 2020 mon 1 12 day 1 31\$ </pre>	 Configures the match time of an entry. A string used to match the time and can include one or more of the following specifiers: date_low - date_high—the inclusive range of dates (1-31). date—date day—date DoW_low - DoW_high—the inclusive range of days (Sun-Mon). dow—day of the week hr—hour hour_low - hour_high—the inclusive range of hours (0-23). minute_low - minute_high—the inclusive range of minutes (0-59). min—minute mon—month month_low - month_high—the inclusive range of months (1-12). tod—time of day yr—year year_low - year_high—the inclusive range of years.
Step 8	precedence precedence	to the low values. Configures the precedence of the routing entry.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # precedence 0	
Step 9	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	Example: Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# action complete	• Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument.
	regraphe-entry)# action complete	• Complete the action using the complete keyword.
		• Reject the indicated action using the reject keyword.
	Command or Action	Purpose
---------	--	--
Step 10	dst-adjacency dst_adj	Configures the destination adjacency of an entry in a routing table.
	Example:	
	Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW2	
Step 11	exit	Exits the current mode of the configuration.
	Example:	
	Router(config-sbc-sbe-rtgpolicy- rtgtable-entry)# exit	

Configuring Trunk-Group ID Tables

This task configures src-trunk-group-id and dst-trunk-group-id routing tables.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. tgid-routing
- 6. exit
- 7. call-policy-set policy-set-id
- 8. rtg-src-trunk-group-id-table table-name | rtg-dst-trunk-group-id-table table-name
- 9. entry entry-id
- **10.** action {next-table *goto-table-name* | complete | reject}
- **11. dst-adjacency** *dst_adj*
- 12. match-type {none | any | context | tgid}
- 13. tgid-context tgid-context-name {tgid tgid-name}
- 14. exit

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal Router(config)#	
Step 2	sbc sbc-name	Enters the mode of an SBC service. The <i>sbc-name</i> argument defines the name of the service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe Router(config-sbc-sbe)#	
Step 4	adjacency sip adjacency-name	Enters adjacency SIP configuration submode.
	<pre>Example: Router(config-sbc-sbe)# adjacency sip adj1 Router(config-sbc-sbe-adj-sip)#</pre>	
Step 5	tgid-routing	Enables parsing the trunk group identifier for call routing.
	Example: Router(config-sbc-sbe-adj-sip)# tgid-routing Router(config-sbc-sbe-adj-sip)#	
Step 6	exit	
	<pre>Example: Router(config-sbc-sbe-adj-sip)# exit Router(config-sbc-sbe)#</pre>	
Step 7	call-policy-set policy-set-id	Enters the mode of routing policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1 Router(config-sbc-sbe-rtgpolicy)#	
Step 8	rtg-src-trunk-group-id-table table-name	Enters the submode of configuration of a routing table whose entries match on the TGID or TGID context
	<pre>Example: Router(config-sbc-sbe-rtgpolicy)# rtg-src-trunk-group-id-table MyRtgTable Router(config-sbc-sbe-rtgpolicy-rtgtable)#</pre>	parameters of an SBE policy set.

	Command or Action	Purpose
Step 9	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry, if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) #</pre>	<i>entry-id</i> is a number that uniquely identifies an entry in the newly created routing table.
Step 10	<pre>action [next-table goto-table-name complete reject]</pre>	Configures the action to take if this routing entry is chosen. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action complete Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) #</pre>	 Set the name of the next routing table to process if the event matches this entry. This is done using the next-table keyword and the <i>goto-table-name</i> argument. Complete the action using the complete keyword. Reject the indicated action using the reject keyword.
Step 11	dst-adjacency dst_adj	Configures the destination adjacency of an entry in a routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency SIP-AS540-PSTN-GW2 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) #</pre>	
Step 12	<pre>match-type {none any context tgid}</pre>	Matches the entries of the routing table with the source TGID or TGID context parameter. Possible match types are:
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-type tgid Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)	 none: Matches an entry if no TGID information is present. any: Matches an entry if any TGID information is
	#	present.
		 context: Matches an entry on the TGID context. tgid: Matches an entry on both the TGID and TGID context.
Step 13	<pre>tgid-context tgid-context-name {tgid tgid-name}</pre>	Defines trunk-group ID context and trunk-group ID to match the entries of the routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # tgid-context example-domain tgid trunkgroup1 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) #</pre>	
Step 14	exit	Exits the current mode of the configuration.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit Router(config-sbc-sbe-rtgpolicy-rtgtable)#</pre>	

Configuring Number Manipulation

This task enables you to specify various number manipulations that can be performed on a dialed number after a destination adjacency has been selected.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. rtg-src-address-table table-id
- 6. rtg-src-adjacency-table table-id
- 7. rtg-src-account-table table-id
- 8. rtg-round-robin-table table-id
- 9. rtg-carrier-id-table table-id
- 10. rtg-dst-address-table table-id
- **11.** entry entry-id
- **12**. **edit** *action*
- **13.** edit-cic [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- **14.** edit-src [del-prefix pd] | [del-suffix sd] | [add-prefix pa] | [replace ds]
- 15. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set policy-set-id	Enters the mode of the routing policy set configuration in the SBE mode, creating a new policy set if necessary
	Example: Router(config-sbc-sbe)# call-policy-set 1	

	Command or Action	Purpose
Step 5	rtg-src-address-table table-id	Enters the configuration mode of a routing table (creating one if necessary) whose entries match the dialer's number or SIP user name within the context of an SBE policy set.
	Router(config-sbc-sbe-rtgpolicy)# rtg-src-address-table MySrcAddressTable	You are not allowed to enter the submode of routing table configuration in the context of the active policy set.
		The no version of the command destroys the routing table. A routing table may not be destroyed if it is in the context of the active policy set.
Step 6	rtg-src-adjacency-table table-id	Enters the configuration mode of a routing table (creating one if necessary) within the context of an SBE policy set whose entries match the source adjacency.
	Router(config-sbc-sbe-rtgpolicy)# rtg-src-adjacency-table MySrcAdjTable	
Step 7	<pre>rtg-src-account-table table-id Example: Router(config-sbc-sbe-rtgpolicy)#</pre>	Enters the configuration mode of a routing table (creating one if necessary) whose entries match the source account within the context of an SBE policy set.
Cton 0	rtg-src-account-table MySrcAccTable	
Steh o	Example: Router(config-sbc-sbe-rtgpolicy)# rtg-round-robin-table MyRobinTable	events do not have any match-value parameters, nor next-table actions. Its actions are restricted to configuring number manipulation, as well as setting the destination adjacency. A group of adjacencies are chosen for an event if an entry in a routing table matches that event and points to a round-robin adjacency table in the next-table action.
Step 9	rtg-carrier-id-table table-id	Enters the configuration mode of a routing table (creating one if necessary) within the context of an SBE policy set whose entries match the carrier ID.
	Router(config-sbc-sbe-rtgpolicy)# rtg-carrier-id-table MyCarrierIdTable	You are not allowed to enter the mode of the routing table configuration in the context of the active policy set.
		The no version of the command destroys the routing table. A routing table may not be destroyed if it is in the context of the active policy set.
Step 10	<pre>rtg-dst-address-table table-id Example: Router(config_shc_shc_shc_rtgpolicy)#</pre>	Enters the configuration mode of a routing table (creating one if necessary) within the context of an SBE policy set whose entries match the dialed number (after number analysis) or SIP user name.
	rtg-dst-address-table MyRtgTable	You are not allowed to enter the submode of routing table configuration in the context of the active policy set.
		The no version of the command destroys the routing table. A routing table may not be destroyed if it is in the context of the active policy set.

	Command or Action	Purpose
Step 11	entry entry-id	Enters the mode for configuring an entry in a routing table, creating the entry if necessary.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1</pre>	
Step 12	edit action	Configures a dial-string manipulation action in the routing table. You are not allowed to do this if the table is part of the active policy set.
	Example. Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # edit del-prefix 1	The no version of the command deletes the edit action of the given entry in the routing table.
		The edit command can be set to the following values:
		• del-prefix <i>pd</i> —Delete prefix <i>pd</i> , where <i>pd</i> is a positive integer specifying a number of digits to delete from the front of the dialed digit string.
		• del-suffix <i>sd</i> —Delete suffix <i>sd</i> , where <i>sd</i> is a positive integer specifying a number of digits to delete from the end of the dialed digit string.
		• add-prefix <i>pa</i> —Add prefix <i>pa</i> , where <i>pa</i> is a string of digits to add to the front of the dialed string.
		• replace <i>ds</i> —Replace <i>ds</i> , where <i>ds</i> is a string of digits that replaces the dialed string.
		In the example to the left, the edit command sets entry 1 to delete 1 digit from the beginning of the dialed string in the routing table "MyRtgTable".
Step 13	edit-cic [del-prefix pd] [del-suffix sd]	Configures a CIC manipulation action in any routing table.
		You are not allowed to do this if the table is part of the active policy set.
	Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# edit-cic del-prefix 1	• del-prefix <i>pd</i> : A positive integer specifying a number of digits to delete from the front of the carrier ID string.
		• del-suffix <i>sd</i> : A positive integer specifying a number of digits to delete from the end of the carrier ID string.
		• add-prefix <i>pa</i> : A string of digits to add to the front of the carrier ID string.
		• replace <i>ds</i> : A string of digits to replace the carrier ID string with.
		The following command sets entry 2 to delete the first digit of the carrier ID in the current routing table.
		If you wish to remove the carrier ID entirely from outgoing messages, you should specify a replacement string of 0000 or a prefix deletion length of 4. For example,
		edit-cic del-prefix 4 OR edit-cic replace 0000

	Command or Action	Purpose	
Step 14	edit-src [del-prefix pd] [del-suffix sd] [add-prefix pa] [replace ds]	Configures a source number manipulation action in the routing table.	
	Example:	You are not allowed to do this if the table is part of the active policy set.	
	Router(config-sbc-sbe-rtgpolicy-natable-entry)# edit-src del-prefix 1	The no version of the command destroys the match value.	
		• del-prefix <i>pd</i> : A positive integer specifying a number of digits to delete from the front of the carrier ID string.	
		• del-suffix <i>sd</i> : A positive integer specifying a number of digits to delete from the end of the carrier ID string.	
		• add-prefix <i>pa</i> : A string of digits to add to the front of the carrier ID string.	
		• replace <i>ds</i> : A string of digits to replace the carrier ID string with.	
Step 15	exit	Exits the entry mode of the configuration.	
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit</pre>		

Configuring Hunting

This task enables Cisco Unified Border Element (SP Edition) to hunt for other routes or destination adjacencies in case of a failure.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name or adjacency h323 adjacency-name
- 5. hunting-trigger error-codes or hunting-trigger error-codes
- 6. exit
- 7. h323
- 8. hunting-mode [altEndps | multiARQ]
- 9. end
- **10.** show sbc *sbc-name* sbe *h323* | *sip* hunting-trigger
- **11.** show sbc *sbc-name* sbe *h323* | *sip* hunting-mode

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency sip adjacency-name or adjacency h323 adjacency-name</pre>	Configures a destination SIP or H.323adjacency for the SBC service, and enters into adjacency sipor adjacency h323 configuration mode.
	<pre>Example: Router(config-sbc-sbe)# adjacency sip test Router(config-sbc-sbe)# adjacency h323 test</pre>	adjacency sip —A destination SIP adjacency where the configured failure return codes cause hunting to occur. This command overrides any globally configured retry error codes.
		adjacency h323 —A destination H.323 adjacency where the configured failure return codes cause hunting to occur. This command overrides any globally configured retry error codes.

	Comm	and or Action	Purpose)	
Step 5	<pre>hunting-trigger error-codes or hunting-trigger error-codes</pre>		Configu occur, in	res which failure return codes cause hunting to none of the following four modes:	
			• sip con	(global SIP scope)—use the sip hunting-trigger mand.	
	Examp	le:			
	Router(config-sbc-sbe-adj-sip)# hunting-trigger 415 480		Note	Exit (config-sbc-sbe-adj-sip) or (config-sbc-sbe-adj-h323) mode first and enter	
	(This SIP ac	(This command configures the hunting trigger for a SIP adjacency in Adjacency SIP configuration mode.)		into (config-sbc-sbe) mode to configure in the global SIP scope level.	
	or Boute	r(config-sbc-sbe-adi-b323)#	• h32	3 (global H.323 scope)—use the hunting-trigger	
	hunti: Route	ng-trigger noBandwidth r(config-sbc-sbe-adj-h323)#	• adja	acency sip (destination SIP adjacency)—use the	
	hunti	ng-trigger unreachableDestination	hunting-trigger command		
	(These commands configure the hunting trigger for an H.323 adjacency in Adjacency H.323 configuration		• adja the	acency h323 (destination H.323 adjacency)—use hunting-trigger command	
	mode.)	error-co	odes can have the following values:	
	Note	If both adjacency level and SBE level hunting triggers are configured, the adjacency level takes priority.	In t repr cod rou "48 are for	he sip and adjacency sip modes, <i>error-codes</i> resent a space-separated list of SIP numeric error es. The examples to the left configures SIP to retry ting if it receives a "415" (media unsupported) or 0" (temporarily unavailable) error. Both error codes set as hunting triggers. See Table 7-2 on page 7-22 a list of SIP error codes.	
			• In t are list	he h323 and adjacency h323 modes, <i>error-codes</i> entered in separate commands. The following is a of H.323 textual error codes:	
			-	noBandwidth—The bandwidth is taken away or the ARQ is denied.	
			-	unreachableDestination—The terminal cannot reach the gatekeeper for ARQ.	
			-	destinationRejection—The code has been rejected at destination.	
			-	noPermission—The callee's gatekeeper rejects the code.	
			-	gatewayResources—The gateway resources are exhausted.	
			-	badFormatAddress—The address field in the H.225 message is not understood.	
			-	securityDenied—The security settings are incompatible.	

	Command or Action	Purpose
		 the internally-defined value "connectFailed"—Either a releaseComplete response was received that gave no cause or any reason code for the release, or there was no response from the remote endpoint.
		Note These textual error codes apply to H.323 only.
		If you type no sip hunting-trigger or no hunting-trigger , then all error codes are cleared out. If you type no sip hunting-trigger x y , then just the codes x and y are removed from the configured list.
		Note In the case of the adjacency h323 mode, enter the noRetry value to specify that routing should never be retried for this adjacency no matter what failure return code is received.
Step 6	exit	Exits the Adjacency H.323 configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-h323)# exit	
Step 7	h323	The h323 command enters into the H.323 configuration mode.
	Example: Router(config-sbc-sbe)# h323	
Step 8	hunting-mode [altEndps/multiARQ]	Configures the form of H.323 hunting to perform if H.323 hunting is triggered.
	Example:	• altEndps—alternateEndpoints
	Router(config-sbc-sbe-h323)# hunting-mode multiARQ	• multiARQ—uses a nonstandard H.323 mechanism based on issuing multiple ARQs to a Gatekeeper for a single call.
		The no version of this command restores the hunting mode to the default of alternateEndpoints. It does not disable hunting completely. If the hunting mode is not defined, the default is alternateEndpoints.
Step 9	end	Exits the current mode of the configuration and enters into Privileged EXEC mode.
	Example: Router(config-sbc-sbe-h323)# end	

	Command or Action	Purpose
Step 10	show sbc sbc-name sbe h323/sip hunting-trigger	Shows the H.323 or SIP hunting triggers.
	Example:	
	Router# show sbc mysbc sbe h323 hunting-trigger	
Step 11	show sbc sbc-name sbe h323 sip hunting-mode	Shows the H.323 hunting mode.
	Example:	
	Router# show sbc mysbc sbe h323 hunting-mode	

Activating a Routing Policy Set

This task activates a number analysis and routing policy set.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set default *policy-set-id* [priority *priority-value*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.Use the <i>sbc-name</i> argument to define the name of the
	Example: Router(config)# sbc mysbc	service.

	Command or Action	Purpose
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	call-policy-set default <i>policy-set-id</i> [priority <i>priority-value</i>]	Assigns the default call-policy-set id when an administrative domain is not specified on the adjacency or the specified administrative domain does not exist.
	Example: Router(config-sbc-sbe)# call-policy-set default 1	• <i>policy-set-id</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
		• priority —Specifies the priority to determine which active call-policy-set to use. The SBC uses the policy set with the highest priority.
		• <i>priority</i> — The priority value ranging from 1 to 10 with 10 indicating highest priority. By default, priority is set to 6.

Configuring H.323 MultiARQ Hunting

This task configures Cisco Unified Border Element (SP Edition) to hunt for other H.323 routes or destination adjacencies in case of a failure.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. hunting-trigger error-codes
- 6. hunting-mode mode
- 7. exit
- 8. show sbc *sbc-name* sbe h323 hunting-mode

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Configures a destination H.323adjacency for the SBC service, and enters into adjacency h323 configuration mode.
	Example: Router(config-sbc-sbe)# adjacency h323 test	A destination H.323 adjacency is where the configured failure return codes cause hunting to occur. This command overrides any globally configured retry error codes.

	Command or Action	Purpose
Step 5	hunting-trigger error-codes	Configures which failure return codes cause hunting to occur, in one of the following configuration modes:
	Example:	• h323 (global H.323 scope)
	Router(config-sbc-sbe-h323)# hunting-trigger noBandwidth	• adjacency h323 (destination H.323 adjacency)
	Router(config-sbc-sbe-h323)# hunting-trigger securityDenied	The example to the left configures H.323 to retry routing if it receives a "noBandwidth" or "securityDenied" error codes.
		In the h323 and adjacency h323 configuration modes, <i>error-codes</i> are entered in separate commands. The following is a list of H.323 textual error codes:
		– noBandwidth
		- unreachableDestination
		- destinationRejection
		– noPermission
		- gatewayResources
		 badFormatAddress
		- securityDenied
		- the internally-defined value "connectFailed"
		If you type no hunting-trigger , all error codes are cleared out.
		Note In the case of the adjacency h323 mode, enter the noRetry value to specify that routing should never be retried for this adjacency no matter what failure return code is received.
Step 6	hunting-mode [altEndps multiARQ]	Configures the form of hunting to perform if hunting is triggered.
	Example:	• altEndps—alternateEndpoints
	Router(config-sbc-sbe-h323)# hunting-mode multiARQ	• multiARQ—uses a nonstandard H.323 mechanism based on issuing multiple ARQs to a Gatekeeper for a single call.
		The no version of this command restores the hunting mode to the default of alternateEndpoints. It does not disable hunting completely. If the hunting mode is not defined, the default is alternateEndpoints.
Step 7	exit	Exits the current mode of the configuration and enters into Privileged EXEC mode.
	Example: Router(config-sbc-sbe-h323)# exit	
Step 8	show sbc sbc-name sbe h323 hunting-mode	Shows the H.323 hunting mode.
	Example: Router# show sbc mysbc sbe h323 hunting-mode	

Configuring Call Admission Control Policy Sets, CAC Tables, and Global CAC Policy Sets

This optional task configures Call Admission Control policy sets, CAC tables, and assigns a global CAC policy set.



If you are performing this procedure to modify an active CAC policy set, see the ?\$paranum>Modifying Active CAC Policy Sets? section on page 7-8 prior to performing the procedure.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set averaging-period avg-number avg-period
- 5. cac-policy-set policy-set-id
- 6. first-cac-scope scope-name
- 7. first-cac-table table-name
- 8. cac-table table-name
- 9. table-type {policy-set | limit {list of limit tables}}
- **10.** entry *entry-id*
- **11.** cac-scope {*list of scope options*}
- **12.** match-value key
- **13.** max-num-calls *mnc*
- 14. max-call-rate-per-scope limit [averaging-period period-num]
- 15. max-in-call-msg-rate limit [averaging-period period-num]
- 16. max-out-call-msg-rate *limit* [averaging-period *period-num*]
- 17. max-bandwidth *mbw bwsize*
- 18. callee-privacy callee-priv-setting
- **19. action** [next-table *goto-table-name* | cac-complete]
- 20. exit
- 21. entry entry-id
- **22**. match-value *key*
- **23**. **max-num-calls** *mnc*
- 24. max-call-rate-per-scope limit [averaging-period period-num]
- 25. max-bandwidth *mbw bwsize*
- 26. transcode-deny
- 27. max-regs-rate-per-scope limit [averaging-period period-num]
- **28.** action [next-table *goto-table-name* | cac-complete]

- 29. exit
- 30. exit
- 31. complete
- 32. exit
- 33. cac-policy-set global cac-policy-num
- 34. end
- 35. show sbc *sbc-name* sbe cac-policy-set [global]

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set averaging-period avg-number	Specifies the averaging period for rate calculations.
	Example:	 <i>avg-number</i>—The averaging period number, can be 1 or 2. <i>avg-period</i>—The averaging period used by CAC in rate
	averaging-period 1 100 Router(config-sbc-sbe)# cac-policy-set averaging-period 2 175	calculations in seconds, can range from 1 to 3600 seconds. By default, 60 seconds is configured.
Step 5	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.

	Command or Action	Purpose
Step 6	first-cac-scope scope-name	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope global	Note The first-cac-scope definition is only relevant if the table type configured by the first-cac-table command is a Limit table. In that case, the scope of the first-cac-table is determined by first-cac-scope. If the first-cac-table is a Policy Set table, the first-cac-scope is ignored and defaults to global.
		The <i>scope-name</i> argument configures the scope at which limits should be initially defined. Possible values are:
		• adj-group
		• call
		• category
		• dst-account
		• dst-adj-group
		• dst-adjacency
		• dst-number
		• global
		• src-account
		• src-adj-group
		• src-adjacency
		• src-number
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies.
Step 7	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table StandardListByAccount	
Step 8	<pre>cac-table table-name</pre>	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	SDE poncy set.

Command or Action	Purpose
<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)#</pre>	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
table-type policy-set	• <i>account</i> —Compare the name of the account.
	• <i>adj-group</i> —Compare the name of the adjacency group.
	• <i>adjacency</i> —Compare the name of the adjacency.
	• <i>all</i> —No comparison type. All events match this type.
	• <i>call-priority</i> —Compare with call priority.
	• <i>category</i> —Compare the number analysis assigned category.
	• <i>dst-account</i> —Compare the name of the destination account.
	• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
	• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
	• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
	• <i>event-type</i> —Compare with CAC policy event types.
	• <i>src-account</i> —Compare the name of the source account.
	• <i>src-adj-group</i> —Compare the name of the source adjacency group.
	• <i>src-adjacency</i> —Compare the name of the source adjacency.
	• <i>src-prefix</i> —Compare the beginning of the calling number string.
	Note For Limit tables, the event or message or call matches only a single entry.
	Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
	When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
	Note For Policy Set tables, the event or call or message is applied to all entries in this table.

	Command or Action	Purpose
Step 10	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy- cactable)# entry 1	
Step 11	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each of the entries at which limits are applied in a policy set table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # cac-scope category</pre>	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
		- account —Events that are from the same account.
		 adjacency—Events that are from the same adjacency.
		 adj-group—Events that are from members of the same adjacency group.
		- call —Scope limits are per single call.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		- dst-number —Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server).

	Command or Action	Purpose
Step 12	match-value key	Configures the match-value of an entry in a CAC Limit table. It is only relevant for Limit table types.
	Example: Router(config-sbc-sbe-cacpolicy- cactable-entry)# match-value SIP-CUSTOMER-1	The <i>key</i> argument is a string or a keyword based on the table type. The format of the key is determined by the Limit table type (for example, Limit event-type tables or Limit call-priority tables).
		For Limit event-type tables (table-type limit <i>event-type</i>), the match value string options are the following:
		• <i>call-update</i> —Compare the beginning of the calling number string.
		• <i>endpoint-reg</i> —Compare the name of the destination adjacency.
		• <i>new-call</i> —Compare the beginning of the dialed digit string.
		For Limit call-priority tables (table-type limit <i>call-priority</i>), the match value string options are the following:
		• <i>critical</i> —Match calls with resource priority 'critical'.
		• <i>flash</i> —Match calls with resource priority 'flash'.
		• <i>flash-override</i> —Match calls with resource priority 'flash-override'.
		• <i>immediate</i> —Match calls with resource priority 'immediate'.
		• <i>priority</i> —Match calls with resource priority 'priority'.
		• <i>routine</i> —Match calls with resource priority 'routine'.
		For all other Limit tables, enter a name or digit string
		• <i>WORD</i> —Name or digit string to match. (Max Size 255).
Step 13	max-num-calls mnc	Configures the maximum number of calls of an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy- cactable-entry)# max-num-calls 100	
Step 14	<pre>max-call-rate-per-scope limit [averaging-period period-num]</pre>	Configures the maximum call rate for an entry in an admission control table.
	Example:	• <i>limit</i> —The limit for the number of new calls per minute. The value can range from 0 to 2147483647.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-call-rate-per-scope 1000 averaging-period 2	• averaging-period —Specifies the averaging-period to use in the rate calculation. By default, 1 is selected.
		• <i>period-num</i> —Calculates rate based on specified averaging period, ranging from 1 to 2.

	Command or Action	Purpose
Step 15	<pre>max-in-call-msg-rate limit [averaging-period period-num]</pre>	Configures the maximum in call rate for an entry in an admission control table.
	Example:	• <i>limit</i> —The limit for the number of in-call messages per minute. The value can range from 0 to 2147483647.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-in-call-msg-rate 500 averaging-period 2	• averaging-period —Specifies the averaging-period to use in the rate calculation. By default, 1 is selected.
		• <i>period-num</i> —Calculates rate-based on specified averaging period, ranging from 1 to 2.
Step 16	<pre>max-out-call-msg-rate limit [averaging-period period-num]</pre>	Configures the maximum out call rate for an entry in an admission control table.
	Example:	• <i>limit</i> —The limit for the number of new calls per minute. The value can range from 0 to 2147483647.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-out-call-msg-rate 500 averaging-period 2	• averaging-period —Specifies the averaging-period to use in the rate calculation. By default, 1 is selected.
		• <i>period-num</i> —Calculates rate-based on specified averaging period, ranging from 1 to 2.
Step 17	<pre>max-bandwidth mbw bwsize Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-bandwidth 1000000 bps</pre>	Configures the maximum bidirectional bandwidth for an entry in an admission control table. For example, if a max-bandwidth value is configured, the SBC allows half of this value in each direction.
		The <i>mbw</i> argument is a positive integer specifying the total maximum rate at which call media should be admitted in both directions (in bytes per second).
		The bwsize argument specifies the transfer size to which mbw refers. Possible values are:
		• bps
		• Kbps
		• Mbps
		• Gbps
Step 18	callee-privacy [callee-priv-setting]	Configures the level of privacy processing to perform on messages sent from callee to caller.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	The <i>callee_priv_setting</i> argument indicates the specific callee privacy setting. Possible values are:
	# callee-privacy never	• never—Indicates to never hide identity.
		• account-boundary—Indicates to hide identity only if caller is different account from callee.
		• always—Indicates to always hide identity.

	Command or Action	Purpose
Step 19	action [next-table goto-table-name cac-complete]	Configures the action to perform after this entry in an admission control table. Possible actions are:
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name argument</i> .
	-	• Stop processing for this scope using the cac-complete keyword.
Step 20	exit	Exits from entry to cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 21	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 2</pre>	
Step 22	match-value key	Configures the match-value of an entry in a CAC Limit table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value SIP-CUSTOMER-2</pre>	The <i>key</i> argument is a string used to match events. The format of the key is determined by the Limit table type (for example, Limit event-type tables or Limit call-priority tables). See the match-value command page for more details.
Step 23	max-num-calls mnc	Configures the maximum number of calls of an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-num-calls 110</pre>	
Step 24	<pre>max-call-rate-per-scope limit [averaging-period period-num]</pre>	Configures the maximum call rate for an entry in an admission control table.
	Example:	• <i>limit</i> —The limit for the number of new calls per minute. The value can range from 0 to 2147483647.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-call-rate-per-scope 1000 averaging-period 2	• averaging-period —Specifies the averaging-period to use in the rate calculation. By default, 1 is selected.
		• <i>period-num</i> —Calculates rate-based on specified averaging period, ranging from 1 to 2.

	Command or Action	Purpose
Step 25	<pre>Step 25 max-bandwidth mbw bwsize Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-bandwidth 2000000 bps</pre>	Configures the maximum bidirectional bandwidth for an entry in an admission control table. For example, if a max-bandwidth value is configured, the SBC allows half of this value in each direction.
		The <i>mbw</i> argument is a positive integer specifying the total maximum rate at which call media should be admitted in both directions (in bytes per second).
		The bwsize argument specifies the transfer size to which mbw refers. Possible values are:
		• bps
		• Kbps
		• Mbps
		• Gbps
Step 26	transcode-deny	Forbids transcoding for this entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # transcode-deny</pre>	
Step 27	<pre>max-regs-rate-per-scope limit [averaging-period period-num]</pre>	Configures the maximum call number of subscriber registrations for an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-regs-rate-per-scope 300 averaging-period 2</pre>	 <i>limit</i>—The limit for the number of new calls per minute. The value can range from 0 to 2147483647. averaging-period—Specifies the averaging-period to use in the rate calculation. By default, 1 is selected. <i>period-num</i>—Calculates rate-based on specified averaging period, ranging from 1 to 2.
Step 28	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to perform after this entry in an admission control table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	 Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name</i> argument. Stop processing for this scope using the cac-complete keyword.
Step 29	exit	Exits from entry to cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 30	exit	Exits from cactable to cacpolicy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	

	Command or Action	Purpose
Step 31	complete	Completes the CAC policy set when you have committed the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 32	exit	Exits the SBE CAC policy mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# exit</pre>	
Step 33	cac-policy-set global policy-num	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Router(config-sbc-sbe)# cac-policy-set global 23	• <i>policy-num</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 34	end	Exits the SBE mode to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 35	<pre>show sbc sbc-name sbe cac-policy-set [global]</pre>	Displays details of the CAC policy sets configured on the SBC.
	Example: Router# show sbc mySBC sbe cac-policy-set	 <i>sbc-name</i>—Defines the name of the SBC service. global—Lists the information pertaining to the global CAC policy set.

The following example shows the output of the show sbc sbe cac-policy-set command:

```
Router# show sbc mySBC sbe cac-policy-set
SBC Service "mySBC"
CAC Averaging period 1: 100 sec
CAC Averaging period 2: 1500 sec
CAC Policy Set 2
 Global policy set: Yes
  First CAC table: 1
 First CAC scope: src-adjacency
  Table name: 1
    Table type: limit adjacency
    Total call setup failures (due to non-media limits): \ensuremath{\texttt{0}}
    Entry Match value
                                                 Action
                                                                         Failures
    ____
           _____
                                                 ____
                                                                         _____
    1
           SIPP1A
                                                 Complete
                                                                                0
    2
            SIPP1B
                                                 Complete
                                                                                0
CAC Policy Set 12
 Global policy set: No
 First CAC table: 1
 First CAC scope: global
  Table name: 1
   Table type: limit adjacency
    Total call setup failures (due to non-media limits): 0
```

	Entry	Match value	Action	Failures
	2	SIPP1B	Complete	0
CAC C	Policy Global po First CAC	Set 21 licy set: No table: 1		
F	'irst CAC	scope: src-adjacency		
T	able nam Table t Total c	e: 1 ype: limit adjacency all setup failures (due to	non-media limits): 0	
CAC	2 Policy	Set 22		
C F F	lobal po irst CAC irst CAC	licy set: No table: scope: global		
Ţ	able nam Table t	e: table1 ype: limit adjacency		
	Total c Entry	all setup failures (due to Match value	non-media limits): 0 Action	Failures
	1		Complete	0
CAC	Policy	Set 25		
G F	lobal po first CAC	licy set: No table: TBL2		
г	IISU CAC	. scope: global		
Г	able nam' Table t	ne: Table2 Sype: limit adjacency		
	Total c Entry	all setup failures (due to Match value	non-media limits): 0 Action	Failures
	1	SIPP	Complete	0

The following example shows the output of the show sbc sbe cac-policy-set global command:

```
Router# show sbc mySBC sbe cac-policy-set global
SBC Service "mySBC"
CAC Averaging period 1: 100 sec
CAC Averaging period 2: 1500 sec
CAC Policy Set 2
 Global policy set: Yes
 First CAC table: 1
 First CAC scope: src-adjacency
 Table name: 1
   Table type: limit adjacency
   Total call setup failures (due to non-media limits): 0
   Entry Match value
                                              Action
                                                                      Failures
    ____
           -----
                                               ____
                                                                      _____
   1
           SIPP1A
                                              Complete
                                                                             0
   2
           SIPP1B
                                              Complete
                                                                             0
```

Configuring Privacy Service

This section describes the tasks to configure the privacy service on a CAC policy set, adjacencies, and number analysis table:

- Configuring Privacy Service on a CAC Policy Set, page 7-126
- Configuring Privacy Service on Adjacencies, page 7-132
- Configuring a Number Analysis Table, page 7-134

Configuring Privacy Service on a CAC Policy Set

This task shows how to configure the privacy service on a CAC policy set.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type {policy-set | limit {list of limit tables}}
- 7. entry entry-id
- 8. caller-privacy edit-privacy-request {pass | strip | insert | replace | sip {strip {all | critical | header | id | none | session | token word | user} | insert {critical | header | id | none | session | token word | user}}}
- 9. callee-privacy edit-privacy-request {pass | strip | insert | replace | sip {strip {all | critical | header | id | none | session | token word | user} | insert {critical | header | id | none | session | token word | user}}
- 10. caller-privacy privacy-service {adj-trust-boundary | always | never}
- 11. callee-privacy privacy-service {adj-trust-boundary | always | never}
- 12. end

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.
Step 5	cac-table table-name	Enters the admission control table configuration mode (creating one, if necessary) within the context of an SBE
	Example:	policy set.
	Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	

	Command or Action	Purpose	
Step 6	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.	
	<pre>Example: Acuter(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set</pre>	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. <i>list of limit tables</i> values are:	
		• <i>account</i> —Compares the name of the account.	
		• <i>adj-group</i> —Compares the name of the adjacency group.	
		• <i>adjacency</i> —Compares the name of the adjacency.	
		• <i>all</i> —No comparison type. All the events match this type.	
		• <i>call-priority</i> —Compares with call priority.	
		• <i>category</i> —Compares the number analysis-assigned category.	
		• <i>dst-account</i> —Compares the name of the destination account.	
		• <i>dst-adj-group</i> —Compares the name of the destination adjacency group.	
		• <i>dst-adjacency</i> —Compares the name of the destination adjacency.	
		• <i>dst-prefix</i> —Compares the beginning of the dialed digit string.	
		• event-type—Compares with CAC policy event types.	
		• <i>src-account</i> —Compares the name of the source account.	
		• <i>src-adj-group</i> —Compares the name of the source adjacency group.	
		• <i>src-adjacency</i> —Compares the name of the source adjacency.	
		• <i>src-prefix</i> —Compares the beginning of the calling number string.	
		Note For Limit tables, the event, message, or call matches only a single entry.	
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacency. The adj-group table type matches on either the source adjacency group or the destination adjacency group.	
		After the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry in which limits are applied in a CAC policy set table.	
		Note In Policy Set tables, the event, call, or message is applied to all the entries.	

	Command or Action	Purpose
Step 7	entry entry-id	Enters the CAC table entry mode to create or modify an
		entry in an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 8	caller-privacy edit-privacy-request {pass strip insert replace sip {strip {all critical header id none session token word user} insert {critical header id	Edits and updates the privacy indications provided by the user: • insert—Inserts privacy restrictions:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-privacy edit-privacy-request strip</pre>	- SIP—Inserts Privacy:header;session;user;id;critical if the header is not present already.
		 H323—Sets presentation indicator from allowed to restricted.
		• pass —Passes on the privacy header or the presentation indicators.
		• replace —Replaces privacy restrictions:
		 SIP—Replaces Privacy:header;session;user;id;critical, except when none has been requested.
		 H323—Sets the presentation indicator to restricted.
		• strip —Removes all the privacy restrictions:
		- SIP —Removes the Privacy header.
		- H323—Sets the presentation indicator to allowed.
		• sip —Specifies the following SIP settings. This allows greater control and overrides all generic actions:
		 insert—Inserts privacy tokens into the Privacy header.
		 strip—Removes privacy tokens from the Privacy header.
		• critical —Specifies the call to be discontinued if privacy cannot be achieved in the SIP Privacy header.
		• header —Obscures all the header information, which is related to the user, from the SIP Privacy header.
		• id —Removes ID headers from the SIP Privacy header.
		• none —Privacy is not applied to the call.
		• session —Specifies media privacy for the session in the SIP Privacy header. No media bypass is performed.
		• token —Specifies the nonstandard user-defined privacy token in the SIP Privacy header.
		• <i>word</i> —Specifies the user-defined privacy token.
		• user —Removes all nonessential header information, which is related to the user, from the SIP Privacy header.
		By default, the privacy setting value is set to pass .

	Command or Action	Purpose
Step 9	callee-privacy edit-privacy-request {pass	Edits and updates privacy indications provided by the user:
	<pre>strip insert replace sip {strip {all critical header id none session token word user} insert {critical header id none session token word user}}} Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee-privacy edit-privacy-request strip</pre>	• insert —Inserts privacy restrictions:
		 SIP—Inserts Privacy:header;session;user;id;critical if the header is not present already.
		- H323—Sets presentation indicator from allowed to restricted.
		• pass —Passes on the privacy header or the presentation indicators.
		• replace —Replaces privacy restrictions:
		 SIP—Replaces Privacy:header;session;user;id;critical, except when none has been requested.
		 H323—Sets the presentation indicator to restricted.
		• strip —Removes all the privacy restrictions:
		- SIP —Removes the Privacy header.
		- H323—Sets the presentation indicator to allowed.
		• sip —Specifies the following SIP settings. This allows greater control and overrides all generic actions:
		 insert—Inserts privacy tokens into the Privacy header.
		 strip—Removes privacy tokens from the Privacy header.
		• critical —Specifies the call to be discontinued if privacy cannot be achieved in the SIP Privacy header.
		• header —Obscures all the header information, which is related to the user, from the SIP Privacy header.
		• id —Removes ID headers from the SIP Privacy header.
		• none —Privacy is not applied to the call.
		• session —Specifies media privacy for the session in the SIP Privacy header. No media bypass is performed.
		• token —Specifies the nonstandard user-defined privacy token in the SIP Privacy header.
		• word—Specifies the user-defined privacy token.
		• user —Removes all nonessential header information, which is related to the user, from the SIP Privacy header.
		By default, the privacy setting value is set to pass .

	Command or Action	Purpose
Step 10	<pre>caller-privacy privacy-service {adj-trust-boundary always never} Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-privacy privacy-service always</pre>	 Configures privacy settings according to RFC3323, RFC3325, and/or setting of the H.323 presentation restriction settings in a given entry in the admission control table: adj-trust-boundary—Specifies the adjacency privacy trust level to determine if the privacy service is
		 required. always—Provides privacy service always, if requested by the user.
		• never —Never provides privacy service even if requested by the user.
		By default, the privacy setting value is set to adj-trust-boundary .
Step 11	callee-privacy privacy-service {adj-trust-boundary always never}	Configures privacy settings according to RFC3323, RFC3325, and/or setting of H.323 presentation restriction settings in a given entry in the admission control table:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee-privacy privacy-service edit trust boundary;</pre>	• adj-trust-boundary —Specifies the adjacency privacy trust level to determine if the privacy service is required.
	adj-trust-boundary	• always —Provides privacy service always, if requested by the user.
		• never —Never provides privacy service even if requested by the user.
		By default, the privacy setting value is set to adj-trust-boundary .
Step 12	end	Exits from the CAC table entry configuration mode and enters the Privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Configuring Privacy Service on Adjacencies

This task shows how to configure the privacy service on the SIP and H323 adjacencies.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. privacy [inherit-profile | trusted | untrusted]
- 6. exit
- 7. adjacency h323 adjacency-name

- 8. allow private info
- 9. end

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode. Use the <i>sbc-name</i> argument to define the name of the service.
	Example:	
	Router(config)# sbc mysbc	
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the SBE SIP adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency sip SIPP	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	<pre>privacy [inherit-profile trusted untrusted]</pre>	Configures the trust level for determining whether the privacy service should be applied:
	<pre>Example: Router(config-sbe-adj-sip)# privacy trusted</pre>	• inherit-profile —Specifies that the trust level for determining whether privacy services are required is derived from the adjacencies inherit-profile.
		• trusted —Specifies that the adjacency is trusted and privacy services do not have to be applied.
		• untrusted —Specifies that the adjacency is not trusted and requires privacy services to be applied.
		By default, the trust level is set to inherit-profile.
Step 6	exit	Exits the SIP adjacency mode and enters the SBE mode.
	Example: Router(config-sbe-adj-sip)# exit	
Step 7	adjacency h323 adjacency-name	Configures a destination H.323 adjacency for the SBC service, and enters into H. 323 adjacency configuration mode.
	Example: Router(config-sbc-sbe)# adjacency h323 test	A destination H.323 adjacency is where the configured failure return codes cause hunting to occur. This command overrides any globally configured retry error codes.

	Command or Action	Purpose
Step 8	allow private info	Configures the H.323 adjacency to allow private information to be sent.
	Example: Router(config-sbe-adj-h323)# allow private info	By default, the H.323 adjacency does not send the private information of a user.
Step 9	end	Exits from a H.323 adjacency configuration mode and entry the Privileged EXEC mode.
	Example: Router(config-sbe-adj-h323)# end	

Configuring a Number Analysis Table

This task shows how to configure a number analysis table to detect anonymity.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. call-policy-set policy-set-id
- 5. na-src-name-anonymous-table table-name
- 6. entry entry-id
- 7. match-anonymous [false | true]
- 8. end

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
Step 4	Example: Router(config-sbc)# sbe call-policy-set policy-set-id	Enters the routing policy set configuration mode within an
		SBE entity.
	Example: Router(config-sbc-sbe)# call-policy-set 1	
Step 5	na-src-name-anonymous-table table-name	Enters the configuration mode of a number analysis table to determine whether the display name or presentation number
	Example: Router(config-sbc-sbe-rtgpolicy)# na-src-name-anonymous-table NameTable	is anonymous.
Step 6	entry entry-id	Enters the number analysis table entry mode for configuring an entry in a number analysis table, creating the entry, if
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable)# entry 1</pre>	necessary.
Step 7	match-anonymous [false true]	Matches the display name or presentation number to Anonymous in the na-src-name-anonymous-table number
	Example:	 false—Specifies the display name or presentation
	match-anonymous false	number as not anonymous.
		• true—Specifies the display name or presentation number as anonymous.
Step 8	end	Exits the number analysis table entry mode and enters the Privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-natable-entry)# end</pre>	

Configuring Multiple SBC Media Bypass

This task shows how to configure the Multiple SBC Media Bypass feature. The steps to configure the renegotiation of media bypass after a session refreshes are also included in this task.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. media bypass {max-data-len data-length | tag sequence-number tag-name}
- 6. exit
- 7. cac-policy-set policy-set-id
- 8. cac-table table-name
- 9. table-type {policy-set | limit {list of limit tables}}
- 10. entry entry-id
- 11. match-value key
- 12. media bypass type [all | none | full [hairpin partial] | hairpin [full partial] | partial [full hairpin]
- 13. caller media bypass {enable | disable}
- 14. callee media bypass {enable | disable}
- **15.** action [next-table goto-table-name | cac-complete]
- 16. exit
- 17. entry entry-id
- **18**. session-refresh renegotiation {allow | suppress}
- 19. end
- 20. show sbc sbc-name sbe cac-policy-set id table name entry entry
- 21. show sbc sbc-name sbe adjacencies adjacency-name detail
DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the SBE SIP adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency sip access	• Use the adjacency-name argument to define the name of the service.
Step 5	<pre>media bypass {max-data-len data-length tag sequence-number tag-name}</pre>	Configures the multiple SBC media bypass feature on a SIP adjacency:
	Example: Router(config-sbc-sbe-adj-sip)# media bypass	• max-data-len —Specifies the maximum length of the multiple SBC media bypass data that can be transmitted on outbound signaling messages on an adjacency.
	cay i iAGi	• tag —Specifies the tag that can be used to control groups to which endpoints on the adjacency belong to the multiple SBC media bypass feature.
		• <i>data-length</i> —Specifies the maximum multiple SBC media bypass data length in bytes that can range from 100 to 2048. By default, <i>data-length</i> is set to 1000 bytes.
		• <i>sequence-number</i> —Specifies the sequence number for a media bypass tag in the tag list. The tag list is formed from the set of tags ordered according to their sequence number. The sequence number can range from 1 to 20.
		• <i>tag-name</i> —Specifies the name of the multiple SBC media bypass tag. The total length of all tags in an adjacency cannot exceed 255 characters. Each tag must consist of alphabets, numerals, and special characters. All printable characters other than comma, semi-colon & space.
		Note Media bypass is not supported for H.323 calls.

	Command or Action	Purpose
Step 6	exit	Exits the adjacency SIP mode and enters the SBE entity mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 7	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.
Step 8	cac-table table-name	Enters the admission control table configuration mode (creating one if necessary) within the context of an SBE
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table MyTable	policy set.
Step 9	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the limit of the table types to be matched by the match-value command. For the multiple SBC media bypass feature, use the following table type:
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type src-adjacency	• <i>src-adjacency</i> —Compare the name of the source adjacency.
Step 10	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 11	match-value key	Configures the match-value of an entry in a CAC Limit Table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value access</pre>	• <i>key</i> —Specifies the keyword used to match events. The format of the key is determined by the table-type limit.

	Command or Action	Purpose
Step 12	media bypass type [all none full [hairpin partial] hairpin [full partial] partial [full hairpin]	Configures the multiple SBC media bypass feature for CAC policy set.
	Evample:	• all —Enables all, such as partial, hairpin, and full types of media bypass for the CAC table entry.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # media bypass type full hairpin	• none —Disables all types of media bypass for the CAC table entry.
		• full —Enables media bypass on the SBC if adjacent and non-adjacent downstream and upstream hops have direct media connectivity, common tags in bypass tag list or with same VPN.
		• hairpin —Enables media bypass for the hairpin calls.
		• partial —Enables media bypass if the SBC is a member of a group of SBCs that share the same IP realm and if even one SBC within that group is on the media path.
		Note If the media bypass type is explicitly configured to be partial, only IP realm and VPN configuration on the adjacency can be used to determine whether media bypass is possible. Because media bypass tags are not used, the VPN names must be globally unique across all the SBCs for partial media bypass to work.
Step 13	caller media bypass {enable disable}	Enables or disables the multiple SBC media bypass feature on the caller side.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller media bypass enable	• enable —Enables the multiple SBC media bypass feature on the caller side.
		• disable —Disables the multiple SBC media bypass feature on the caller side.
Step 14	callee media bypass {enable disable}	Enables or disables the multiple SBC media bypass feature on the callee side.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee media bypass enable	• enable —Enables the multiple SBC media bypass feature on the callee side.
		• disable —Disables the multiple SBC media bypass feature on the callee side.
Step 15	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to be performed after this entry, in an admission control table. Possible actions are:
	Example:	• Identify the CAC table to be processed next using the next-table keyword and the <i>goto-table-name</i> argument.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• Stop the processing action for this scope using the cac-complete keyword.

	Command or Action	Purpose
Step 16	exit	Configures the action to be performed after this entry, in an admission control table. Possible actions are:
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• Identify the CAC table to be processed next using the next-table keyword and the <i>goto-table-name</i> argument.
	# action cac-complete	• Stop the processing action for this scope using the cac-complete keyword.
Step 17	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 2</pre>	
Step 18	<pre>session-refresh renegotiation {allow suppress}</pre>	Depending on the option that you select, one of the following actions is configured:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # session-refresh renegotiation suppress</pre>	• allow —Specifies that an offer that contains duplicate SDP must be processed using the normal offer-answer rules. Media reservations can change, and interworking functions can be renegotiated.
		• suppress —Specifies that an offer that contains duplicate SDP must be processed using the session refresh variant of the offer-answer rules. Media reservations are not changed, and interworking functions are not renegotiated. The SBC forwards the last sent offer or answer regardless of the offer or answer that was received.
		The default is that the session refresh strategy for the call is not affected by this CAC policy entry.
Step 19	end	Exits from the CAC table entry configuration mode and enters the Privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	
Step 20	show sbc <i>sbc-name</i> sbe cac-policy-set <i>id</i> table <i>name</i> entry <i>entry</i>	Displays detailed information about a specific entry in a CAC policy table.
	Example: Router# show sbc mysbc sbe cac-policy-set 1 table MyTable entry 1	
Step 21	show sbc sbc-name sbe adjacencies adjacency-name detail	Displays all the detailed field outputs for the specified SIP adjacency.
	Example: Router# show sbc sbe mySBC sbe adjacencies access detail	

Configuring Common IP Address Media Bypass

This procedure shows how to configure the Common IP Address Media Bypass feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. media bypass auto-nat-tag-gen
- 6. end
- 7. show sbc sbc-name sbe adjacencies adjacency-name detail

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service. Use the <i>sbc-name</i> argument to define the name of the service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip access-side-1	• <i>adjacency-name</i> —Name of the adjacency.
Step 5	media bypass auto-nat-tag-gen	Configures the Common IP Address Media Bypass feature to generate a media bypass tag for the registered endpoints
	Example: Router(config-sbc-sbe-adj-sip)# media bypass auto-nat-tag-gen	that are benind a NAT device associated with this adjacency.

	Command or Action	Purpose
Step 6	end	Exits the SBE SIP adjacency mode, and enters the privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-adj-sip)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adj-name detail</pre>	Shows the configuration details of the specified adjacency.
	Example:	
	access-side-1 detail	

Activating a CAC Policy Set

This task activates a global CAC policy set.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set global policy-set-id

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service. Use the <i>sbc-name</i> argument to define the name of the service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set global policy-set-id	Activates the global CAC policy set within an SBE entity.
	Example: Router(config-sbc-sbe)# cac-policy-set global 1	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.

Configuring Asymmetric Payload Types

This task configures SBC to allow asymmetric payload types.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table table-name
- 7. table-type policy-set
- 8. entry entry-id
- 9. action cac-complete
- 10. payload-type asymmetric allowed
- 11. complete
- 12. cac-policy-set global policy-set-id
- 13. end
- 14. show sbc sbc-name sbe cac-policy-set

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Enables entry into the mode of an SBC service.
		Use the <i>sbc-name</i> argument to define the name of an SBC.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enables entry into the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enables entry into the mode of the CAC policy.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	

	Command or Action	Purpose
Step 5	first-cac-table table-name	Configures the name of the first policy table to be processed when performing the admission control stage of the CAC policy.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table</pre>	
Step 6	cac-table table-name	Enables entry into the mode for configuring an admission control table (or creating one, if necessary) within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table	
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC Policy table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	
Step 8	entry entry-id	Enables entry into the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 9	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to be performed after this entry, in an admission control table. Possible actions are:
	Fuerrales	• Identify the next CAC table to be processed using the next-table keyword and the <i>goto-table-name</i> argument.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• Stop the processing for this scope using the cac-complete keyword.
Step 10	payload-type asymmetric allowed	Configures SBC to allow asymmetric payload types.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # payload-type asymmetric allowed</pre>	
Step 11	complete	Completes the CAC policy.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 12	cac-policy-set global policy-set-id	Activates the global CAC policy set within an SBE entity.
	Example: Router (config-sbc-sbe)# cac-policy-set global 1	

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	Command or Action	Purpose
Step 13	end	Enables exit from the CAC policy set configuration mode and entry into the Privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# end	
Step 14	show sbc <i>sbc-name</i> sbe cac-policy-set <i>id</i> table <i>name</i> entry <i>entry</i>	Displays detailed information for a specific entry in a CAC policy table, including any restricted codecs.
	Example:	
	Router# show sbc mysbc sbe cac-policy-set 1 table standard_policy_list entry 1	

Limiting Resource Usage

New router features, such as transcoding, transrating, and inband DTMF interworking, have been introduced in earlier releases. If no limits are set on the number of calls that use the resources provided by these features, overload conditions may occur and the router may stop responding. You can configure limits on resource usage to prevent the occurrence of overload conditions. This is one of the areas in which Cisco Unified Border Element (SP Edition) policies can be applied.

Note

The Limiting Resource Usage feature has been introduced in Release 3.4S.

You can configure media policies to specify maximum levels of usage for the following:

- Number of audio streams using transcoding
- Number of audio streams using transrating
- Number of video streams using transcoding
- · Number of audio streams using inband DTMF interworking
- Number of streams using SRTP encryption and decryption
- Number of registered subscribers using IPSec encryption and decryption on the signaling link to the SBC
- Number of calls made by subscribers who are using IPSec-protected signaling
- Total number of video and audio streams using transcoding, transrating, inband DTMF interworking, and SRTP encryption and decryption—weighted by the costs assigned to each of these resources.

Table 7-10 lists the default resource costs. You can modify these default resource costs.

Table 7-10 Default Resource Costs

Resource	Default Resource Cost
Audio transcoding	10
Audio transrating	6
Video transcoding	50

Resource	Default Resource Cost
Inband DTMF interworking	4
SRTP encryption and decryption	15

Table 7-10Default Resource Costs

At run time, the total resource usage value is calculated for each incoming call using the resource costs configured for the resources requested by the call. This calculated value is then compared with the maximum total resource usage value that you have configured. If the calculated value is more than the configured value, the media policy rejects the call. This means that the call either fails or is directed to a different message gateway or signaling route.

After you define a media policy, you can apply it in one of the following ways:

• As a CAC policy

For example, call-scoped policies restrict resource usage for a particular call. In contrast, adjacency-scoped policies restrict resource usage at the adjacency level.

• As a media gateway policy

Media policies applied at the media gateway level restrict resource usage for the media gateway.

After you apply a media policy, you can view the resource usage of each resource for which you have specified a limit in the media policy. For example, you can view the number of media streams that are being video-transcoded by the message gateway on which you have applied the media policy.

The following sections describe the procedures for limiting resource usage:

- Configuring Resource Costs for Transcoding, Transrating, Inband DTMF Interworking, and SRTP Encryption and Decryption, page 7-146
- Configuring Usage Limits for Transcoding, Transrating, Inband DTMF Interworking, and SRTP Encryption and Decryption, page 7-149
- Configuring Usage Limits for IPSec Encryption and Decryption and IPSec-Protected Signaling, page 7-153
- Example: Limiting Resource Usage, page 7-167

Configuring Resource Costs for Transcoding, Transrating, Inband DTMF Interworking, and SRTP Encryption and Decryption

The resource costs that you have configured are used to calculate and compare the total weighted resource usage against the maximum total usage that you have configured. Table 7-10 shows the default resource costs. You can modify these resource costs to suit the requirements of your operating environment.

This task explains how to configure resource costs for transcoding, transrating, inband DTMF interworking, and SRTP encryption and decryption.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe

- 4. media-gateway policy type {default | local | {remote {ipv4 | ipv6} *ip-address* [port *port-number*]}}
- 5. transcode audio cost number
- 6. transcode video cost number
- 7. transrate audio cost *number*
- 8. interwork inband-dtmf cost number
- 9. interwork srtp cost number
- 10. end
- 11. show sbc *sbc-name* sbe media-gateway-policy
- 12. show sbc *sbc-name* sbe media-gateway-policy [stats | type {default | local | remote {ipv4 | ipv6} *ip-address* [port *port-number*]}]

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example:	
	Router(config)# sbc mySbc	
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose		
Step 4	media-gateway policy type {default local	Configures a media gateway policy.		
	<pre>{remote {ipv4 ipv6} ip-address [port port-number]}} Example: Router(config-sbc-sbe)# media-gateway policy type remote ipv4 192.0.2.26 6886</pre>	• default —Specifies that the media gateway policy must be applied to all media gateways configured on the SBC. A default media gateway policy is applied on a media gateway (local or remote) when no other media policy is applied on the media gateway.		
		• local —Specifies that the media gateway policy must be applied to the media gateway that is locally configured on the SBC.		
		• remote —Specifies that the media gateway policy must be applied to a remote media gateway.		
		• ipv4 —Specifies that the remote media gateway has an IPv4 IP address.		
		• ipv6 —Specifies that the remote media gateway has an IPv6 IP address.		
		• <i>ip-address</i> — IP address of the remote media gateway. The IP address can be in the IPv4 format or IPv6 format.		
		• port —Specifies the port number of the remote media gateway.		
		• <i>port-number</i> —Port number of the remote media gateway.		
		Enters the media policy configuration mode.		
Step 5	transcode audio cost number	Specifies the resource cost for transcoding an audio stream.		
	Example: Router(config-sbc-sbe-media-pol)# transcode audio cost 10	• <i>number</i> —Resource cost. The range is from 1 to 4294967295. As mentioned in Table 7-10, the default cost is 10.		
Step 6	transcode video cost number	Specifies the resource cost for transcoding a video stream.		
	Example: Router(config-sbc-sbe-media-pol)# transcode video cost 55	• <i>number</i> —Resource cost. The range is from 1 to 4294967295. As mentioned in Table 7-10, the default cost is 50.		
Step 7	transrate audio cost number	Specifies the resource cost for transrating an audio stream.		
	Example: Router(config-sbc-sbe-media-pol)# transrate audio cost 10	• <i>number</i> —Resource cost. The range is from 1 to 4294967295. As mentioned in Table 7-10, the default cost is 6.		
Step 8	interwork inband-dtmf cost number	Specifies the resource cost for an audio stream using inband DTMF interworking.		
	Example: Router(config-sbc-sbe-media-pol)# interwork inband-dtmf cost 6	• <i>number</i> —Resource cost. The range is from 1 to 4294967295. As mentioned in Table 7-10, the default cost is 4.		

	Command or Action	Purpose		
Step 9	interwork srtp cost number	Specifies the resource cost for an audio or video stream using SRTP encryption and decryption.		
	Example: Router(config-sbc-sbe-media-pol)# interwork srtp cost 15	• <i>number</i> —Resource cost. The range is from 1 to 4294967295. As mentioned in Table 7-10, the default cost is 15.		
Step 10	end	Exits the media policy configuration mode, and enters the privileged EXEC mode.		
	Example: Router(config-sbc-sbe-media-pol)# end			
Step 11	show sbc sbc-name sbe media-gateway-policy	Displays the details of all media gateway policies.		
		• <i>sbc-name</i> —Name of the SBC.		
	Example: Router# show sbc mySbc sbe media-gateway-policy			
Step 12	show sbc sbc-name sbe media-gateway-policy	Displays the details of the specified media gateway policy.		
	[stats type {default local remote {ipv4 ipv6} ip-address [port port-number]}]	• <i>sbc-name</i> —Name of the SBC.		
	Example: Router# show sbc mySbc sbe media-gateway-policy	• <i>ip-address</i> — IP address of the remote media gateway. The IP address can be in the IPv4 format or IPv6 format.		
	cype remote ipv4 i92.0.2.20 port 6886	• <i>port-number</i> —Port number of the remote media gateway.		

The following example shows the output of the **show sbc sbe media-gateway-policy type** command for a specified media gateway policy:

Router# show sbc mySbc sbe media-gateway-policy type remote ipv4 192.0.2.26 port 6886

Gateway Policy Type	=	REMOTE
Remote vpn	=	0
Remote address type	=	IPV4
Remote address	=	192.0.2.26
Remote Port	=	6886
Media Limit Table	=	
Transcode Audio Cost	=	10
Transrate Audio Cost	=	6

Configuring Usage Limits for Transcoding, Transrating, Inband DTMF Interworking, and SRTP Encryption and Decryption

This task describes how to configure usage limits for transcoding, transrating, inband DTMF interworking, and SRTP encryption and decryption.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. media-policy policy-name
- 5. type {cac-policy | gateway}
- 6. transcode audio maximum number
- 7. transcode video maximum number
- 8. transrate audio maximum number
- 9. interwork inband-dtmf maximum number
- 10. interwork srtp maximum number
- 11. total resource maximum number
- 12. exit
- **13.** media-gateway policy type {default | local | {remote {ipv4 | ipv6} *ip-address* [port *port-number*]}}
- 14. media limits policy-name
- 15. end
- 16. show sbc *sbc-name* sbe media-policy
- **17. show sbc** *sbc-name* **sbe media-policy** *policy-name*

	Command or Action	Purpose			
Step 1	configure terminal	Enables the global configuration mode.			
	Example:				
	Router# configure terminal				
Step 2	sbc sbc-name	Enters the SBC service mode.			
		• <i>sbc-name</i> —Name of the SBC.			
	Example:				
	Router(config)# sbc mySbc				
Step 3	sbe	Enters the SBE configuration mode.			
	Example:				
	Router(config-sbc)# sbe				
Step 4	media-policy policy-name	Specifies the media policy to be created.			
		• <i>policy-name</i> —Name of the media policy.			
	Example:	Enters the media policy configuration mode			
	Router(config-sbc-sbe)# media-policy media_policy2	Eners the media poncy configuration mode.			

	Command or Action	Purpose		
Step 5	type {cac-policy gateway}	Specifies the type of media policy table to be configured. You can specify one of the following media policy types:		
	Example: Router(config-sbc-sbe-media-pol)# type gateway	• cac-policy —Specifies that a media policy table must be configured for a CAC-policy type policy.		
		• gateway —Specifies that a media policy table must be configured for a gateway type policy.		
Step 6	transcode audio maximum number	Specifies the maximum number of media streams that can be audio transcoded at any point of time.		
	Example: Router(config-sbc-sbe-media-pol)# transcode audio maximum 20000	• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 7	transcode video maximum number	Specifies the maximum number of media streams that can be video transcoded at any point of time.		
	Example: Router(config-sbc-sbe-media-pol)# transcode video maximum 20000	• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 8	transrate audio maximum number	Specifies the maximum number of media streams that can be audio transrated at any point of time.		
	Example: Router(config-sbc-sbe-media-pol)# transrate audio maximum 6000	• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 9	interwork inband-dtmf maximum number	Specifies the maximum number of media streams that can use the inband DTMF interworking resource at any point of time.		
	Example: Router(config-sbc-sbe-media-pol)# interwork inband-dtmf maximum 2000	• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 10	interwork srtp maximum number	Specifies the maximum number of media streams that can use the SRTP interworking resource at any point of time.		
	Example: Router(config-sbc-sbe-media-pol)# interwork srtp maximum 500	• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 11	total resource maximum number Example:	Specifies the total number of video and audio streams that can use transcoding, transrating, inband DTMF interworking, and SRTP encryption and		
	Router(config-sbc-sbe-media-pol)# total resource maximum 35000	decryption—weighted by the costs assigned to each of these resources.		
		• <i>number</i> —Number of media streams. The range is from 1 to 4294967295. The default is 4294967295.		
Step 12	exit	Exits the SBE media policy configuration mode, and enters the SBE configuration mode.		
	Example: Router(config-sbc-sbe-media-pol)# exit			

	Command or Action	Purpose		
Step 13	media-gateway policy type {default local	Configures a media gateway policy.		
	<pre>{remote {ipv4 ipv6} ip-address [port port-number]}} Example: Router(config-sbc-sbe)# media-gateway policy type default</pre>	• default —Specifies that the media gateway policy must be applied to all media gateways configured on the SBC. A default media gateway policy is applied on a media gateway (local or remote) when no other media policy is applied on the media gateway.		
		• local —Specifies that the media gateway policy must be applied to the media gateway that is locally configured on the SBC.		
		• remote —Specifies that the media gateway policy must be applied to a remote media gateway.		
		• ipv4 —Specifies that the remote media gateway has an IPv4 IP address.		
		• ipv6 —Specifies that the remote media gateway has an IPv6 IP address.		
		• <i>ip-address</i> —IP address of the remote media gateway. The IP address can be in the IPv4 format or IPv6 format.		
		• port —Specifies the port number of the remote media gateway.		
		• <i>port-number</i> —Port number of the remote media gateway.		
		Enters the media policy configuration mode.		
Step 14	media limits policy-name	Specifies the media policy to be associated with the CAC policy table entry or applied on the media gateway.		
	Example: Router(config-sbc-sbe-mg-pol)# media limits media_policy2	• <i>policy-name</i> —Name of the media policy.		
Step 15	end	Exits the media policy configuration mode, and enters the privileged EXEC mode.		
	Example:			
0	Router(config-sbc-sbe-media-pol)# end			
Step 16	show sbc sbc-name sbe media-policy	Displays details of all media policies. These details inclu the resource usage limits that you have configured.		
	Example: Router# show sbc mySbc sbe media-policy	• <i>sbc-name</i> —Name of the SBC service.		
Step 17	show sbc sbc-name sbe media-policy policy-name	Displays details of all media policies. These details include the resource usage limits that you have configured.		
	Example:	• <i>sbc-name</i> —Name of the SBC service.		
	Router# show sbc mySbc sbe media-policy	• <i>policy-name</i> —Name of the media policy.		

The following example shows the output of the **show sbc sbe media-policy** command for a specified media policy:

Router# show sbc mySbc sbe media-policy my_media_policy

Policy Name: my_media_policy

Гуре		=	gateway
Audio transcode	limit	=	30
Audio transrate	limit	=	30
/ideo transcode	limit	=	30
Inband-dtmf-iw	limit	=	10
SRTP-iw	limit	=	20
Total resource	limit	=	40

Configuring Usage Limits for IPSec Encryption and Decryption and IPSec-Protected Signaling

This task explains how to configure usage limits for IPSec encryption and decryption and IPSec-protected signaling.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set {policy-set-id | copy {source policy-set-id destination policy-set-id} | swap {source policy-set-id destination policy-set-id} | averaging-period {average-number average-period}
- 5. cac-table table-name
- 6. entry entry-id
- 7. ipsec maximum registers number
- 8. ipsec maximum calls number
- 9. end
- 10. show sbc sbc-name sbe cac-policy-set policy-set-id detail
- 11. show sbc sbc-name sbe cac-policy-set policy-set-id table table-name detail
- 12. show sbc sbc-name sbe cac-policy-set policy-set-id table table-name entry entry-id

DETAILED STEPS

	Command or Action	Purpose		
Step 1	configure terminal	Enables the global configuration mode.		
	Example: Router# configure terminal			
Step 2	sbc <i>sbc</i> - <i>name</i>	Enters the SBC service mode.		
	Example: Router(config)# sbc mySbc	• <i>sbc-name</i> —Name of the SBC.		
Step 3	sbe	Enters the SBE configuration mode.		
	Example: Router(config-sbc)# sbe			
Step 4	<pre>cac-policy-set {policy-set-id copy {source policy-set-id destination policy-set-id} swap</pre>	Enters the CAC policy set configuration mode within an		
	<pre>{source policy-set-id destination policy-set-id} averaging-period {average-number average-period}</pre>	 <i>policy-set-id</i>—CAC policy set number. The range is from 1 to 2147483647. 		
	Example: Router(config-sbc-sbe)# cac-policy-set 1	Note The keywords and arguments of the cac-policy-set command that are not relevant to this section have not been described here. See <i>Cisco Unified Border Element (SP Edition) Command Reference: Unified Model</i> for information about these keywords and arguments.		
Step 5	<pre>cac-table table-name</pre>	Enters the CAC table configuration mode within an SBE policy set. If the CAC table does not exist, it is created.		
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table t1	• <i>table-name</i> —Name of the CAC table.		
Step 6	entry entry-id	Enters the mode for configuring an entry in a CAC table. If the entry does not exist, it is created.		
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	• <i>entry-id</i> —ID of the CAC table entry.		
Step 7	ipsec maximum registers number	Specifies the maximum number of endpoint registrations that can use IPsec encryption and decryption on their signaling link to the SBC.		
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # ipsec maximum registers 10	• <i>number</i> —Number of endpoint registrations. The range is from 1 to 4294967295. The default is 4294967295.		
		Note This configuration is not used when determining the call scope. In addition, this configuration is not used when the SBC performs the Interconnection Border Control Function (IBCF) because all registrations are stateless and the SBC cannot determine whether a registration is new.		

	Command or Action	Purpose		
Step 8	ipsec maximum calls number	Specifies the maximum number of calls that can use IPsec-protected signaling.		
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # ipsec maximum calls 5</pre>	• <i>number</i> —Number of calls. The range is from 1 to 4294967295. The default is 4294967295.		
Step 9	end	Exits the SBE mode, and returns to the privileged EXEC mode.		
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>			
Step 10	<pre>show sbc sbc-name sbe cac-policy-set policy-set-id detail</pre>	 Shows detailed information pertaining to a CAC policy set. <i>sbc-name</i>—Name of the SBC. 		
	Example: Router(config)# show sbc mySbc sbe cac-policy-set 1 detail	• <i>policy-set-id</i> —ID of the policy set.		
Step 11	show sbc sbc-name sbe cac-policy-set policy-set-id table table-name detail	Shows detailed information pertaining to a table in a CAC policy set.		
		• <i>sbc-name</i> —Name of the SBC.		
	Example: Router(config)# show sbc mySbc sbe cac-policy-set 1 table t1 detail	• <i>policy-set-id</i> —ID of the policy set.		
		• <i>table-name</i> —Name of the table.		
Step 12	<pre>show sbc sbc-name sbe cac-policy-set policy-set-id table table-name entry entry-id</pre>	Shows detailed information pertaining to an entry in a table in a CAC policy set.		
	Example: Router(config)# show sbc mySbc sbe cac-policy-set 1 table t1 entry 1	 sbc-name—Name of the SBC. policy-set-id—ID of the policy set. table-name—Name of the table. 		
		• <i>entry-ta</i> —ID of the CAC table entry.		

The following example shows the output of the show sbc sbe cac-policy-set table entry command:

```
Router# show sbc mySbc sbe cac-policy-set 1 table t1 entry 1
SBC Service "mySbc"
CAC Averaging period 1: 60 sec
CAC Averaging period 2: 0 sec
CAC Policy Set 1
 Active policy set: No
  Description:
  First CAC table:
  First CAC scope: global
  Table name: t1
   Description:
   Table type: policy-set
   Total call setup failures (due to non-media limits): 0
   Entry 1
   CAC scope:
   CAC scope prefix length: 0
   Action: Not set
```

```
Number of call setup failures (due to non-media limits): 0
media bandwidth policing:
                                   Degrade
Caller ptime:
                                 None (default)
Callee ptime:
                                 None (default)
Caller inband DTMF mode: Inherit(default)
Callee inband DTMF mode: Inherit(default)
Callee inband DTMF mode:
                                   Inherit(default)
Media policy limit table name:
                                   mp1
IPsec maximum registers:
                                   10
                                   5
IPsec maximum calls:
```

Configuration Examples for Implementing Policies

This section provides the following configuration examples:

- Example: Implementing Number Analysis, page 7-156
- Example: Configuring Administrative Domain, page 7-157
- Example: Implementing Call Admission Control Policy Sets and CAC Tables, page 7-159
- Example: Multiple SBC Media Bypass, page 7-161
- Example: Configuring Hunting, page 7-163
- Example: Allowing Asymmetric Payload Types, page 7-164
- Example: Common IP Address Media Bypass, page 7-166
- Example: Limiting Resource Usage, page 7-167
- Example: Configuration the CAC Threshold, page 7-168

Example: Implementing Number Analysis

The following example shows call processing handled with number analysis working with a category routing table in the following manner: 1) shows number analysis, based on number categorization, of a set of dialed digits to determine which is a valid telephone number, 2) shows how the categorized calls are handled with a call routing policy based on category, and 3) shows source address manipulation.

This task configures text address validation and source address manipulation for a number analysis table.

Under 1) for any new call, the SBC inspects the first few digits of the called number that is determined by the "match-prefix" and categorizes the call, based on the category configured under the "na-dst-prefix-table Determine-Category" entry. For example, calls with a prefix of 911 in the destination number are categorized as EMERGENCY calls; calls with a prefix of 919 are Legit_Call, and calls with a prefix of 900 are Blocked_Number calls.

Under 2) routing policy is defined based on category as specified by the "rtg-category-table Category_Routing" table that allows EMERGENCY calls and Legit_Call and rejects all Blocked_Number calls.

```
call-policy-set 1
first-inbound-na-table Determine-Category
first-call-routing-table Category_Routing
rtg-src-adjacency-table Routing-Table-2
entry 1
action complete
dst-adjacency Adj-502
```

```
match-adjacency Adj-503
  entry 2
   action complete
   dst-adjacency Adj-503
   match-adjacency Adj-502
rtg-category-table Category_Routing ==> 2) categorized calls handled with routing policy
  entry 1
   action next-table Routing-Table-1
   match-category EMERGENCY
  entry 2
   action next-table Routing-Table-2
   match-category Legit_Call
  entrv 3
   action complete
   match-category Blocked_Number
rtg-src-adjacency-table Routing-Table-1
 entry 1
   action complete
   dst-adjacency Adj-502
   match-adjacency Adj-501
  entry 2
   action complete
   dst-adjacency Adj-501
   match-adjacency Adj-502
na-dst-prefix-table Determine-Category ====> 1) number analysis based on categorization
  entry 1
   action accept
   category EMERGENCY
   match-prefix 911
  entry 2
   action accept
   category Legit Call
   match-prefix 919
  entry 3
   action reject
   category Blocked_Number
   match-prefix 900
 na-dst-address-table mytable
     entry 1
       action accept
        edit-src del-prefix 3 =====> 3) source address manipulation
       match-address 123456 digits
      entry 2
       action accept
        edit-src del-suffix 1
       match-address ^.* regex
```

Example: Configuring Administrative Domain

The following example shows how to configure the administrative domains:

```
adjacency sip SIPP1A
  admin-domain SIPP1A
  inherit profile preset-access
  signaling-address ipv4 10.10.100.140
  statistics method summary
  signaling-port 7065
  remote-address ipv4 10.10.100.11 255.255.255.255
  signaling-peer 10.10.100.11
  signaling-peer-port 7065
  registration rewrite-register
  attach
```

```
adjacency sip SIPP1B
 admin-domain SIPP1B
 inherit profile preset-access
signaling-address ipv4 10.10.100.140
 statistics method summary
 signaling-port 7066
 remote-address ipv4 10.10.100.12 255.255.255.255
 signaling-peer 10.10.100.12
 signaling-peer-port 7066
 registration rewrite-register
 attach
adjacency sip Registrar
inherit profile preset-core
signaling-address ipv4 10.10.100.140
 statistics method summary
 signaling-port 7020 7029
 remote-address ipv4 10.10.100.12 255.255.255.255
 signaling-peer 10.10.100.12
 signaling-peer-port 7068
 registration contact username passthrough
 registration target address 10.10.100.12
registration target port 7069
 attach
cac-policy-set averaging-period 1 120
cac-policy-set averaging-period 2 40
cac-policy-set 10
 first-cac-table TAB1
 first-cac-scope src-adjacency
 cac-table TAB1
  table-type limit adjacency
  entry 1
  match-value SIPP1A
   action cac-complete
 complete
cac-policy-set 20
 first-cac-table TAB1
 cac-table TAB1
  table-type policy-set
  entry 1
   max-call-rate-per-scope 600 averaging-period 1
   action cac-complete
  complete
cac-policy-set global 20
call-policy-set 10
 first-call-routing-table RTG_TBL
 first-reg-routing-table REG_TBL
rtg-src-adjacency-table RTG_TBL
  entry 1
  match-adjacency SIPP1A
   dst-adjacency SIPP1B
   action complete
 rtg-src-adjacency-table REG_TBL
  entry 1
   match-adjacency SIPP1A
   dst-adjacency Registrar
   action complete
 complete
call-policy-set 20
 first-call-routing-table RTG_TBL
 first-reg-routing-table REG_TBL
 rtg-src-adjacency-table RTG_TBL
```

```
entry 1
     match-adjacency SIPP1A
     dst-adjacency SIPP1B
     action complete
    entry 2
     match-adjacency SIPP1B
     dst-adjacency SIPP1A
     action complete
   rtg-src-adjacency-table REG_TBL
    entrv 1
     match-adjacency SIPP1A
     dst-adjacency Registrar
     action complete
    entry 2
     match-adjacency SIPP1B
     dst-adjacency Registrar
     action complete
   complete
  call-policy-set default 20
  admin-domain SIPP1A
   cac-policy-set 10
   call-policy-set 10
! no admin-domain for SIPP1B defaults to default call-policy
```

Example: Implementing Call Admission Control Policy Sets and CAC Tables

The following example shows how to configure call admission control policy sets and CAC tables:

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router (config-sbc-sbe-cacpolicy) # first-cac-table STANDARD-LIST-BY-ACCOUNT
Router (config-sbc-sbe-cacpolicy) # cac-table STANDARD-LIST-BY-ACCOUNT
Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit dst-account
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIP-CUSTOMER-1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 100
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-call-rate-per-scope 20
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 1000000 bps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable) # entry 2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIP-CUSTOMER-2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 100
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-call-rate-per-scope 20
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 1000000 bps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# transcode deny
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy) # complete
```

The following example limits the total number of concurrent calls per SBC (global limit) to 2000 and the number of concurrent calls per source adjacency to 5. If an adjacency has 5 calls that are active, it is not allowed to make the sixth call even if the total number of active calls on the SBC is less than 2000. Also, if the total number of active calls on the SBC is 2000, an adjacency is not allowed to make a call even if it has no active calls.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe) # cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router(config-sbc-sbe-cacpolicy) # first-cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 5
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable) # entry 2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope global
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 2000
Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy)# exit
Router(config-sbc-sbe) # cac-policy-set global 1
```

The following example limits the number of concurrent calls per subscriber to 5 with no global limit:

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy-cactable) # table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 5
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy) # exit
Router(config-sbc-sbe) # cac-policy-set global 1
```

You could also achieve this with the following configuration:

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 3
Router(config-sbc-sbe-cacpolicy) # first-cac-table 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope src-adjacency
Router(config-sbc-sbe-cacpolicy) # cac-table 1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit all
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 5
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy) # exit
Router(config-sbc-sbe) # cac-policy-set global 1
```

Both of the above configurations will limit the number of concurrent calls per subscriber to 5. There is no global limit.

In the following example, if the bandwidth used by an adjacency whose source IP address is 1.1.1.1 is less than 1 Mbps, then the call is admitted. Also adjacencies with a source IP address of 2.2.2.2 that use less than 2 Mbps of bandwidth will have their calls admitted.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value 1.1.1.1
Router (config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 1 Mbps
Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit
Router(config-sbc-sbe-cacpolicy-cactable)# entry 2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value 2.2.2.2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 2 Mbps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy) # exit
Router(config-sbc-sbe)# cac-policy-set global 1
```

This example allows 10 calls, 100 updates, a max-in-call-msg-rate and a max-out-call-msg-rate of 5000 msg/min for any source adjacency:

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope sub-category
Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable) # entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-num-calls 10
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-updates 100
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-in-call-msg-rate 5000
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-out-call-msg-rate 5000
Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy)# complete
Router(config-sbc-sbe-cacpolicy)# exit
Router(config-sbc-sbe)# cac-policy-set global 1
```

Example: Multiple SBC Media Bypass

The following example shows how to configure a media bypass across two or more SBCs as shown in Figure 7-7, when making calls from endpoint 1 to endpoint 2. In the example, the adjacencies configured on each SBC is named *access*, for endpoint facing adjacency, and *core* for proxy facing adjacency. To achieve media bypass for calls from endpoint 2 to endpoint 1, two CAC entries with match-value as *access* and *core* must be configured with the same settings in the CAC table.

SBC 1:

```
sbc SBC1
   sbe
      adjacency sip access
       .
        •
        media bypass tag 1 enterprise1
        .
      adjacency sip core
       .
        .
      cac-policy-set 1
         cac-table MyTable
           table-type limit src-adjacency
             entry 1
            .
            .
               match-value access
              media bypass type full hairpin
              caller media bypass enable
              callee media bypass enable
              action cac-complete
           entry 2
               session-refresh renegotiation suppress
            .
            .
```

SBC 2:

```
sbc SBC2
   sbe
      adjacency sip access
       .
       .
       media bypass tag 1 enterprise1
      adjacency sip core
      cac-policy-set 1
         cac-table MyTable
           table-type limit src-adjacency
             entry 1
      match-value core
              media bypass type full hairpin
               caller media bypass enable
               callee media bypass enable
               action cac-complete
```

The following example shows the output of the show sbc sbe adjacencies detail command:

```
Router# show sbc SBC1 sbe adjacencies access detail
```

```
SBC Service SBC1
Adjacency access (SIP)
Media Bypass Tag List:
Tag 1: tag1
Tag 2: tag2
Media Bypass Max Out Data Length: 1024
```

The following example shows the output of the **show sbc sbe cac-policy-set table entry detail** command:

```
Router# show sbc SBC1 sbe cac-policy-set 1 table MyTable entry 1 detail
```

```
SBC Service "SBC1"
```

```
CAC Policy Set 1
Active policy set: No
Description:
Averaging period: 60 sec
First CAC table:
First CAC scope: global
```

Table name: MyTable Description: Table type: policy-set

```
Entry 1
Action: CAC Complete
...
Media Bypass Type: Full Partial
Caller Media Bypass: Enabled
Callee Media Bypass: Enabled
```

Example: Configuring Hunting

The following example shows how to hunt for other routes or destination adjacencies in case of a failure in a SIP mode:

```
Router# configure terminal
Router(config) # sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # first-call-routing-table SAMPLE
Router(config-sbc-sbe-rtgpolicy)# first-reg-routing-table SAMPLE
Router(config-sbc-sbe-rtgpolicy) # rtg-src-adjacency-table SAMPLE
Router(config-sbc-sbe-rtgpolicy-rtgtable) # entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency TA1
Router(config-sbc-sbe-rtgpolicy-entry) # match-adjacency Hunted
Router(config-sbc-sbe-rtgpolicy-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency TA2
Router(config-sbc-sbe-rtgpolicy-entry)# match-adjacency Hunted
Router(config-sbc-sbe-rtgpolicy-entry) # exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

```
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency TA3
Router (config-sbc-sbe-rtgpolicy-entry)# match-adjacency Hunted
Router (config-sbc-sbe-rtgpolicy-entry)# exit
Router (config-sbc-sbe-rtgpolicy-rtgtable)# entry 4
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency TA4
Router (config-sbc-sbe-rtgpolicy-entry)# match-adjacency Hunted
Router (config-sbc-sbe-rtgpolicy-entry)# match-adjacency Hunted
Router (config-sbc-sbe-rtgpolicy-entry)# exit
Router (config-sbc-sbe-rtgpolicy)# complete
Router (config-sbc-sbe-rtgpolicy)# exit
Router (config-sbc-sbe-rtgpolicy)# exit
Router (config-sbc-sbe-rtgpolicy)# exit
Router (config-sbc-sbe)# adjacency sip Hunted
Router (config-sbc-sbe-adj-sip)# hunting-trigger 403 415 503 604
Router (config-sbc-sbe-adj-sip)# exit
```

The following example shows how to configure Cisco Unified Border Element (SP Edition) to hunt for other H.323 routes or destination adjacencies in case of a failure: Router# configure terminal Router(config)# sbc mySBC Router(config-sbc)# sbe Router(config-sbc-sbe)# adjacency h323 adjacency-name Router(config-sbc-sbe)# adjacency h323 adjacency-name Router(config-sbc-sbe)# adjacency h323 adjacency-name Router(config-sbc-sbe)# hunting-trigger noBandwidth Router(config-sbc-sbe-h323)# hunting-trigger unreachableDestination Router(config-sbc-sbe-h323)# hunting-trigger unreachableDestination Router(config-sbc-sbe-h323)# hunting-mode altEndps Router(config-sbc-sbe-h323)# exit

Example: Allowing Asymmetric Payload Types

The following example shows how to configure the SBC to specify support for Asymmetric payload types on the mySBC SBC:

```
Router# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-table my_table
Router(config-sbc-sbe-cacpolicy)# cac-table TAB1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# payload-type asymmetric allowed
Router(config-sbc-sbe)# cac-policy-set global 1
Router(config-sbc-sbe)# end
Router#
```

The following example shows a SIP/SIP call with a single CAC Policy Set table allowing Asymmetric payload types:

Configuration:

```
cac-policy-set 1
first-cac-table TAB1
cac-table TAB1
table-type policy-set
entry 1
payload-type asymmetric allowed
action cac-complete
complete
cac-policy-set global 1
```

Call succeeds with the following invite, and 200 messages exchanged:

Invite Sent:

2010-01-12 16:28:35 UDP message sent:

```
INVITE sip:service@2.0.0.5:5078 SIP/2.0
Via: SIP/2.0/UDP 2.0.0.3:5078;branch=z9hG4bK-32567-1-0
From: sipp ;tag=32567SIPpTag091
To: sut
Call-ID: 1-32567@2.0.0.3
CSeq: 1 INVITE
Contact: sip:sipp@2.0.0.3:5078
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 127
```

v=0

```
o=user1 53655765 2353687637 IN IP4 2.0.0.3
s=-
c=IN IP4 2.0.0.3
t=0 0
m=audio 6000 RTP/AVP 18
a=rtpmap:18 G729/8000
```

200 Received:

2010-01-12 16:28:35 UDP message received [485] bytes :

```
SIP/2.0 200 OK
Call-ID: 1-32567@2.0.0.3
CSeq: 1 INVITE
From: sipp ;tag=32567SIPpTag091
To: sut ;tag=sip+1+1060000+47e93fd7
Via: SIP/2.0/UDP 2.0.0.3:5078;branch=z9hG4bK-32567-1-0
Server: CISCO-SBC/2.x
Content-Length: 146
Contact:
Content-Type: application/sdp
```

```
v=0
o=user1 5338645241744 5338645241744 IN IP4 10.10.20.20
s=-
c=IN IP4 10.10.20.20
t=0 0
m=audio 16384 RTP/AVP 118
a=rtpmap:118 G729/8000
```

The following example shows a SIP/SIP call with a single CAC policy set table disallowing Asymmetric payload types:

Configuration:

```
cac-policy-set 1
first-cac-table TAB1
cac-table TAB1
table-type policy-set
entry 1
payload-type asymmetric disallowed
action cac-complete
```

```
complete
cac-policy-set global 1
```

Call fails with the following invite and error messages:

Invite Sent:

2010-01-12 16:39:09 UDP message sent:

```
INVITE sip:service@2.0.0.5:5078 SIP/2.0
Via: SIP/2.0/UDP 2.0.0.3:5078;branch=z9hG4bK-32584-1-0
From: sipp ;tag=32584SIPpTag091
To: sut
Call-ID: 1-32584@2.0.0.3
CSeq: 1 INVITE
Contact: sip:sipp@2.0.0.3:5078
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: 127
```

```
v=0
o=user1 53655765 2353687637 IN IP4 2.0.0.3
s=-
c=IN IP4 2.0.0.3
t=0 0
m=audio 6000 RTP/AVP 18
a=rtpmap:18 G729/8000
```

Error Message:

```
Unexpected UDP message received:
SIP/2.0 400 Bad Request
Call-ID: 1-32584@2.0.0.3
CSeq: 1 INVITE
From: sipp ;tag=32584SIPpTag091
To: sut ;tag=sip+1+10b0000+3621b373
Via: SIP/2.0/UDP 2.0.0.3:5078;branch=z9hG4bK-32584-1-0
Server: CISCO-SBC/2.x
Content-Length: 0
Contact:
```

Example: Common IP Address Media Bypass

The following example shows how to configure the Common IP Address Media Bypass feature on the access-side-1 adjacency:

```
Router# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip access-side-1
Router(config-sbc-sbe-adj-sip)# media bypass auto-nat-tag-gen
Router(config-sbc-sbe-adj-sip)# end
Router# show sbc mySBC sbe adjacencies access-side-1 detail
SBC Service "mySBC "
Adjacency access-side-1 (SIP)
Status: Detached
```

```
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```

```
Register unencrypted convert: Disabled
Warrant Match-Order: None
Media Bypass Max Out Data Length: 1000
Auto bypass NAT: Enabled
```

Example: Limiting Resource Usage

This section describes examples related to implementing the Limiting Resource Usage feature.

In the following example, the local media gateway is configured to support up to 1000 audio transcoded streams.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# media-policy audio_limit1
Router(config-sbc-sbe-media-pol)# type gateway
Router(config-sbc-sbe-media-pol)# transcode audio maximum 1000
Router(config-sbc-sbe-media-pol)# exit
Router(config-sbc-sbe)# media-gateway policy type local
Router(config-sbc-sbe-mg-pol)# media limits audio_limit1
```

In the following example, the remote media gateway at 192.0.2.26 is configured to support up to 1500 audio transcoded streams.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# media-policy audio_limit2
Router(config-sbc-sbe-media-pol)# type gateway
Router(config-sbc-sbe-media-pol)# transcode audio maximum 1500
Router(config-sbc-sbe-media-pol)# exit
Router(config-sbc-sbe)# media-gateway policy type remote ipv4 192.0.2.26 port 2000
Router(config-sbc-sbe-mg-pol)# media limits audio_limit2
```

In the following example, a default media gateway policy is configured to enable media gateways to support up to 2000 audio transcoded streams. This default media gateway policy is applied on a media gateway (local or remote) when no other media policy is applied on the media gateway.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# media-policy audio_limit3
Router(config-sbc-sbe-media-pol)# type gateway
Router(config-sbc-sbe-media-pol)# transcode audio maximum 2000
Router(config-sbc-sbe-media-pol)# exit
Router(config-sbc-sbe)# media-gateway policy type default
Router(config-sbc-sbe-mg-pol)# media limits audio_limit3
```

In the following example, a CAC policy is configured to restrict all destination numbers other than 911 to at most 5 media streams on which audio transcoding or audio transrating can be performed. Note that the CAC table commands to apply this restriction to all numbers other than 911 have not been included in this sequence of commands.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# media-policy media_streams_limit1
Router(config-sbc-sbe-media-pol)# type cac-policy
```

```
Router(config-sbc-sbe-media-pol)# transcode audio maximum 5
Router(config-sbc-sbe-media-pol)# transrate audio maximum 5
Router(config-sbc-sbe-media-pol)# exit
Router(config-sbc-sbe)# cac-policy-set 22
Router(config-sbc-sbe-cacpolicy)# cac-table table1
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
.
.
.
Router(config-sbc-sbe-cacpolicy-cactable-entry)# media limits media_streams_limit1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy)# complete
```

Example: Configuration the CAC Threshold

The following example shows how to configure a charge of 10 per session and a call admission limit of 50, which allows 5 calls per second (50/10) through the system:

```
Router(config)# call admission new-model
Router(config)# call admission limit 50
Router(config)# call admission pppoe 10 1
```

Configuration Examples for Implementing Call Routing

This section provides the following configuration examples:

- Example: Routing with No Load Balancing, page 7-168
- Example: Least Cost Routing, page 7-169
- Example: Weighted Routing, page 7-170
- Example: Time-Based Routing, page 7-170
- Example: Regular Expression Based Routing, page 7-174
- Example: Trunk-Group ID Routing, page 7-174

Example: Routing with No Load Balancing

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# first-call-routing-table start_routing
Router(config-sbc-sbe-rtgpolicy)# rtg-dst-address-table start_routing
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address XXX
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# next-table internal_routing
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address XXXX
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # next-table external_routing
Router(config-sbc-sbe-rtgpolicy)# rtg-src-adjacency-table internal_routing
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address sip_to_foo
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency sip_to_foo
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
```

```
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address sip_to_bar
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip_to_bar
Router (config-sbc-sbe-rtgpolicy) # rtg-dst-address-table external_routing
Router (config-sbc-sbe-rtgpolicy-rtgtable) # entry 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address 208111
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip_to_foo
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit
Router (config-sbc-sbe-rtgpolicy-rtgtable) # entry 2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address 208222
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip_to_bar
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip_to_bar
Router (config-sbc-sbe-rtgpolicy-rtgtable) = entry 3
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address X
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address X
```

Example: Least Cost Routing

The following example configures a routing table that matches on category and then for each entry routes the call to a different least-cost table to choose the adjacency.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-category-table 1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-category internal
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action next-table least_int_cost
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-category external
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action next-table least_ext_cost
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy)# rtg-least-cost-table least_int_cost
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 50
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy) # rtg-least-cost-table least_ext_cost
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 50
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency SipAdj3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 100
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

Example: Weighted Routing

In the above example, no two entries in one table have the same cost, so the weight parameter is left at the default of 1. If two or more entries with equal cost exist, and are selected for routing, then calls are distributed based on the weight configured (weight being the relative weight of an entry with respect to the lowest weight in the table). For example, if there are three entries of equal cost and weights of entry1, entry2, and entry3 are 1, 2, and 4 respectively, entry2 will route twice the number of calls as entry1, and entry3 will route four times the number of calls as entry1.

In the following example, all calls are routed to entry 1 because it has the lowest cost. However if routing fails, the remaining three entries all have the same cost, so the weight parameters determine which entry is picked. 80% of calls will be routed to SipAdj2 by entry 2, and the remaining 20% will be evenly divided between SipAdj3 and SipAdj4 (weights of entry 3 and entry 4 are left at a default of 1).

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtqpolicy) # rtg-least-cost-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 50
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # weight 8
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 50
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# cost 50
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

Example: Time-Based Routing

The following example shows two entries, one that routes traffic to Adj1 at all times and a second with a higher precedence that routes traffic to Adj2 if the time is between 9 AM and 6 PM on a weekday. When the two time periods overlap, the one with the higher precedence is chosen.

The two times ranges in entry 1 and entry 2 overlap. In this case, a call made between 9 AM to 6 PM on weekdays matches on both the entries but entry 2 is preferred due to its higher precedence.

If multiple ranges are specified as in entry 2, the Cisco Unified Border Element (SP Edition) will match the entry only during the intersection of the ranges. For example, entry 2 matches calls made Monday through Friday between 9 AM to 6 PM. The range is not Monday 9 AM to Friday 6 PM.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
```

Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time date yr 2006 2020 mon 1 12 day 1 31 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 5 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 1 5 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 9 17 min 0 59 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2 Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete

The following example configures a rule that routes traffic through adjacency SipAdj1 at all times, and through SipAdj2 between Monday 9 AM and Friday 6 PM.

```
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable) # entry 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # precedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 1 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 9 23 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 2 4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable) # entry 4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 5 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 0 17 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

In the configuration above, entry 2, entry 3, and entry 4 together specify the range Monday 9:00 AM through Friday 6:00 PM. This could also be accomplished by having one route for the entire time Monday through Friday with separate ranges to divert traffic during nights as follows:

```
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# grecedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 1 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
```

```
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router (config-sbc-sbe-rtgpolicy-rtgtable)# entry 3
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 1 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 0 8 min 0 59
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 20
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router (config-sbc-sbe-rtgpolicy-rtgtable)# entry 4
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 5 5
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 18 23 min 0 59
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 20
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 5 5
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 18 23 min 0 59
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 20
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
```

The following example shows how to configure a rule that would route traffic through adjacencies SipAdj1 and SipAdj2 on Monday and Wednesday, respectively, between 9 AM and 6 PM, and through SipAdj3 at all other times.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 1 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time tod hr 9 17 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 3 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 9 17 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

The following example shows how to configure a rule that would route traffic through adjacency SipAdj1 on Saturdays and Sundays between 01 Mar 2008 through 30 Mar 2009, and through SipAdj2 all other times.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time date yr 2008 2009 mon 3 3 day
1 30
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 6 7
```
```
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

The following example shows how to configure a rule that would route traffic through adjacency SipAdj1 between 10:00 PM and 6:00 AM from Friday to Monday, and through SipAdj2 otherwise.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 5 1
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time tod hr 22 6 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

```
Note
```

Time and day of the week are wrapping ranges, so the minimum can be larger than the maximum. For example, a single routing entry with the ranges Friday through Monday and 22:00 through 06:00 will match before 6 AM and after 10 PM on Friday, Saturday, Sunday and Monday.

In the following example, a user has all his routers running GMT no matter where they were so that they can be synchronized. But one router in New York has a time-based routing table that routes traffic to SipAdj1 at all times apart from Monday through Friday from 9 AM to 6 PM when it routes traffic to SipAdj2. The user wants these match times to refer to local time so it is necessary enter a **time-offset** command (New York is five hours behind GMT) as shown in the example below.

```
Router# configure terminal
Router(config) # sbc mySbc
Router(config-sbc) # sbe
Router(config-sbc-sbe) # time-offset hour 5 min 0 negative
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # rtg-time-table table1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# use-time-offset
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-time date yr 2006 2020 mon 1 12 day
1 31
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 5
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# use-time-offset
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time tod hr 9 17 min 0 59
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-time dow 6 7
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# precedence 10
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SipAdj2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
```

Example: Regular Expression Based Routing

The following example shows how to configure the regular expression based routing to match the user name or domain part of a source or destination SIP URI.

```
Router# configure terminal
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-dst-address-table MyRtgTable
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address user regex
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy)# rtg-src-domain-table MyRtgTable
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-domain cisco.com regex
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable)# exit
```

Example: Trunk-Group ID Routing

The following example shows how to configure the TGID routing to match the TGID parameters of a source or destination SIP URI.

```
Router# configure terminal
Router(config) # sbc mysbc
Router(config-sbc) # sbe
Router(config-sbc-sbe) # adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# tgid-routing
Router(config-sbc-sbe-adj-sip)# exit
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# rtg-src-trunk-group-id-table MyRtgTable
Router(config-sbc-sbe-rtgpolicy-rtgtable) # entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SIP-AS540-PSTN-GW2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-type tgid
Router(config-sbc-sbe-rtqpolicy-rtqtable-entry)# tgid-context example-domain tgid
trunkgroup1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# exit
Router(config-sbc-sbe-rtgpolicy-rtgtable) #
```



Call Duration Monitoring

Cisco Unified Border Element (SP Edition) supports the Call Duration Monitoring feature that is used to gracefully terminate calls whose duration has exceeded a configured maximum amount of time. You can configure the maximum call duration to be applied to a call.

Using this feature, the SBC can terminate SIP, H.323, and SIP to H.323 interworked calls, regardless of the signaling and media activity within those calls.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



This feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Feature History for Call Duration Monitoring Feature

Release	Modification
Cisco IOS XE Release 2.5	The Call Duration Monitoring feature was introduced on the
	Cisco IOS XR.

Contents

This chapter contains the following sections:

- Prerequisites, page 8-2
- Information About Call Duration Monitoring, page 8-2
- Configuration Example—Call Duration Monitoring, page 8-7

Prerequisites

The following prerequisite is required to implement this feature:

Before implementing these features, Cisco Unified Border Element (SP Edition) must already be configured.

Information About Call Duration Monitoring

Cisco Unified Border Element (SP Edition) supports the Call Duration Monitoring feature that is used to gracefully terminate calls whose duration has exceeded a maximum amount of time configured with this feature.

If the duration of a call exceeds the configured maximum set by the **max-call-duration** command, Cisco Unified Border Element (SP Edition) gracefully tears down the call.

- For SIP call branches, the SBC sends a SIP BYE to the endpoints.
- For H.323 call branches, the SBC sends a RELEASE COMPLETE message to the endpoints.

If there is a renegotiation in progress when the maximum duration is reached, the SBC attempts to terminate the call as gracefully as possible.

The SBC can terminate SIP, H.323, and SIP to H.323 interworked calls, regardless of the signaling and media activity within those calls.

The SBC will terminate calls under the following conditions:

• Duration has exceeded a maximum amount of time configured by the **max-call-duration** command for this feature.



Note If the **max-call-duration** command is set to the default of zero (0), this results in disabling the Call Duration Monitoring function; and call duration can be determined by other factors, such as no media flow or calls not answered within a specified period of time.

- If no media has flowed on that call for a specified period of time, as configured by the **media-timeout** command.
- Calls that are not answered within a specified amount of time.

Calls are terminated according to whichever timer expires first.

After this feature is configured, the SBC starts a timer to individually monitor each call passing through it. The timer is started:

- For SIP calls, when the call is connected (not on receipt of the message establishing a new call).
- For H.323 calls, when the first SETUP message establishing a new call is received.

Once the timer has been started it cannot be reset.

However, if SBC re-routs a call during call setup, the maximum call duration timer is restarted because the configured maximum duration may have changed (based on the new routing information).

Information About Call Duration Monitoring

Configuring Call Duration Monitoring

This task configures Call Duration Monitoring.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type {policy-set | limit {list of limit tables}}
- 7. entry entry-id
- 8. cac-scope {list of scope options}
- **9.** max-call-duration {*num*}
- **10. action** [**next-table** *goto-table-name* | **cac-complete**]
- 11. exit
- 12. exit
- 13. complete

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	

	Command or Action	Purpose
Step 5	<pre>cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# cac-table</pre>	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set.
	StandardListByAccount	

	Command or Action	Purpose
Step 6	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
		• <i>account</i> —Compare the name of the account.
		• <i>adj-group</i> —Compare the name of the adjacency group.
		• <i>adjacency</i> —Compare the name of the adjacency.
		• <i>all</i> —No comparison type. All events match this type.
		• <i>call-priority</i> —Compare with call priority.
		• <i>category</i> —Compare the number analysis assigned category.
		• <i>dst-account</i> —Compare the name of the destination account.
		• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
		• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
		• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
		• <i>event-type</i> —Compare with CAC policy event types.
		• <i>src-account</i> —Compare the name of the source account.
		• <i>src-adj-group</i> —Compare the name of the source adjacency group.
		• <i>src-adjacency</i> —Compare the name of the source adjacency.
		• <i>src-prefix</i> —Compare the beginning of the calling number string.
		Note For Limit tables, the event or message or call matches only a single entry.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
		Note For Policy Set tables, the event or call or message is applied to all entries in this table.

	Command or Action	Purpose
Step 7	entry entry-id	Enters the CAC table entry configuration mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 8	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each of the entries at which limits are applied in a policy set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope call	- account —Events that are from the same account.
		 adjacency—Events that are from the same adjacency.
		 adj-group—Events that are from members of the same adjacency group.
		- call —Scope limits are per single call.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		- dst-number —Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server)

	Command or Action	Purpose
Step 9	<pre>max-call-duration {num}</pre>	Configures the maximum duration (in seconds) for which a call may exist.
	Example:	num range is from 0 to 2147483 seconds
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-call-duration 6000	By default, the max-call-duration is 0, which results in disabling the Call Duration Monitoring feature.
Step 10	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to perform after this entry in an admission control table. Possible actions are:
	Example:	• Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name</i> argument.
	# action cac-complete	• Stop processing for this scope using the cac-complete keyword.
Step 11	exit	Exits from entry to cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 12	exit	Exits from cactable to cacpolicy mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit</pre>	
Step 13	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # complete</pre>	

Configuration Example—Call Duration Monitoring

The following is a configuration example for the Call Duration Monitoring feature:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table 1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# max-call-duration 6000
Router(config-sbc-sbe-cacpolicy-cactable-entry)#
```



IP Realm Support

The IP Realm feature is supported on the Cisco Unified Border Element (SP Edition) unified model. This feature allows the grouping of addresses known to a data border element (DBE) into realms and supports a method for the signaling border element (SBE) to specify which realm it requires an address from. IP Realm support enables an IP realm to be configured under an adjacency and to be associated with a media address pool.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Support for IP Realm

Release	Modification
Cisco IOS XE Release 2.5	This feature was introduced on the unified model on the Cisco IOS XR.

Contents

This module contains the following sections:

- Prerequisites, page 9-1
- Information About IP Realm Support on the Unified Model, page 9-2
- Configuring IP Realm Under an Adjacency—Unified Model, page 9-3

Prerequisites

The following prerequisites are required to implement Support for IP Realm:

Before implementing Support for IP Realm, Cisco Unified Border Element (SP Edition) must already be configured.

Information About IP Realm Support on the Unified Model

The IP Realm feature is supported in the unified model by means of the configuration of the IP realm under an adjacency and the IP realm association with a media address or a pool of media addresses. In effect, the adjacency is configured with the realm it belongs to, and the media address or a pool of media addresses is configured to belong to a realm. A call coming in on an adjacency is matched up with a specific media address or media address pool based on the configured realm.

The IP Realm feature adds support to both the SIP and H.323 adjacency configuration to require calls on specific adjacencies to request addresses from a specific realm. For example, when media addresses are to be allocated, the media pools with realm configuration matching the realm configuration on the adjacency are used. If there is no media pool with a matching realm configuration, then pools without any realm tags are used. If there is one or more pools that can be used, the selection criteria is not deterministic.

Each DBE address range may only belong to a single realm. This realm may be changed while the SBC is activated. However the realm change only affects calls set up after the realm change is made and does not affect calls already in existence.

Each adjacency may only select addresses from a single realm. The realm for an adjacency may be changed at any time, but the changed realm only affects new calls.

If there is no address pool with a matching realm the call setup is rejected, resulting in a SIP request failing with error code 503 "Service Unavailable" and an H.323 release complete with a release completion reason of "gatewayResources."

Media Address Assignment

The user is able to assign a media address or a media address range to a particular realm. If a realm parameter is specified on an incoming adjacency, the SBC selects a media address or an address from a pool that has a matching realm. This allows users to customize their realm matching to implement features, such as wildcarding of realms.



All of the other address range selection criteria must also match, that is, VPN ID, class of service (for port ranges).

If the IP realm configuration is absent under the adjacency, then an address is selected from a pool with any or no realm.

If an IP realm is specified under the adjacency, but there is no address pool with a matching realm, the call setup is rejected, resulting in an error code from the DBE of 510 "Insufficient resources."

IP Realm Identifier

The IP Realm Identifier is used to indicate to which packet network the media addresses belong. The IP Realm identifier is a string, which may be in a domain name format, for example, "mynet.net" or any other string format. The format of the realm string is up to the user with certain restrictions.

The IP Realm Identifier should be provisioned between the SBE and the DBE. Each of the different IP realms possibly interconnecting with a DBE should have a different identifier.

Allowed Characters	ASCII	Allowed Characters	ASCII	Allowed Characters	ASCII
A - Z	0x41 - 0x5A	&	0x26	?	0x3F
a - z	0x61 - 0x7A	!	0x21	@	0x40
0 - 9	0x30 - 0x39	_	0x5F	^	0x5E
+	0x2B	/	0x2F	`	0x60
-	0x2D	•	0x27	~	0x7E
*	0x2A	\$	0x24	١	0x5C
(0x29)	0x29	%	0x25
	0x7C		0x2E		

Realms strings are case-insensitive and are made up of the characters in Table 9-1.

Table 9-1 IP Realm Identifier String - Allowed Character Set

Configuring IP Realm Under an Adjacency—Unified Model

To configure an IP Realm under an adjacency in the unified model, you need to perform both of the following tasks:

- Tag the adjacency with the realm it belongs to using the realm command.
- Configure the media address or media addresses in a pool to belong to a realm using the **media-address ipv4** or **media-address pool ipv4** command.

Tagging an Adjacency with a Realm

In the SBC unified model, the adjacencies need to be tagged with the realm that they belong to. This will enable subsequent calls to use media addresses from that realm.

The following example shows how to tag the SIP adjacency Cisco-gw with the realm cisco.com:

```
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip Cisco-gw
Router(config-sbc-sbe-adj-sip)# realm cisco.com
```

The following example shows the running configuration after the SIP adjacency Cisco-gw is tagged with the realm cisco.com:

```
Router# show run
adjacency sip Cisco-gw
signaling-address ipv4 200.100.50.8
realm cisco.com
```

Configuring a Media Address or a Pool of Media Addresses to Belong to a Realm

In the SBC unified model, you must configure either a media address or the pool of media addresses to be associated with a realm. If the port range is not configured, the SBC selects the default port range.

The following example configures the media address 40.0.0.1 to belong to the cisco.com realm:

Router(config-sbc)# media-address ipv4 40.0.0.1 realm cisco.com Router(config-sbc-media-address)# port-range 10000 20000 any

The following example configures a pool of media addresses from which the SBC can select. The SBC can select any address from 40.0.0.2 to 40.0.0.31 as the media address to be associated with the cisco.com realm:

Router(config-sbc)# media-address pool ipv4 40.0.0.2 40.0.0.31 realm cisco.com Router(config-sbc-media-address)# port-range 10000 20000 any

The following example shows the running configuration after configuring media address 40.0.0.1 to the cisco.com realm:

```
Router# show run
media-address ipv4 40.0.0.1 realm cisco.com
port-range 10000 20000 any
```

Show Commands—Unified Model

The following are show commands that can be used to display IP realm information in the unified model.

The **show sbc dbe addresses** command lists the H.248 control addresses, media addresses, and IP realm information configured on a DBE:

```
Router# show sbc global dbe addresses
```

```
SBC Service "global"
No controllers configured.
Media-Address: 40.0.0.1
VRF: Global
Port-Range (Service-Class): 10000-20000 (any)
Realm: cisco.com
```

The **show sbc sbe adjacencies** command lists the adjacencies information, including the IP realm information, configured on an SBE:

Router# show sbc global sbe adjacencies Cisco-gw detail

```
SBC Service "global"
Adjacency Cisco-gw (SIP)
Status: Detached
Signaling address: 111.45.103.119:default
Signaling-peer: :5060 (Default)
Force next hop: No
Account:
Group: None
In header profile: Default
Out header profile: Default
In method profile: Default
Out method profile: Default
In body profile: None
Out body profile: None
In UA option prof: Default
Out UA option prof: Default
In proxy opt prof: Default
Out proxy opt prof: Default
Priority set name: None
Local-id: None
Rewrite REGISTER: Off
Target address: None
```

NAT Status: Auto Detect Reg-min-expiry: 3000 seconds Fast-register: Enabled Fast-register-int: 30 seconds Register aggregate: Disabled Registration Required: Disabled Register Out Interval: 0 seconds Parse username params: Disabled Supported timer insert:Disabled Suppress Expires: Disabled p-asserted-id header-value: not defined p-assert-id assert: Disabled Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default Force next hop: No Home network Id: None UnEncrypt key data: None SIPI passthrough: No Passthrough headers: Media passthrough: No Client authentication: No Incoming 100rel strip: No Incoming 100rel supp: No Out 100rel supp add: No Out 100rel reg add: No Parse TGID parms: No IP-FQDN inbound: IP-FQDN outbound: FODN-IP inbound: FQDN-IP outbound: Outbound Flood Rate: None Hunting Triggers: Global Triggers Add transport=tls param: Disabled Redirect mode: Pass-through Security: Untrusted-Unencrypted Ping: Disabled Ping Interval: 32 seconds Ping Life Time: 32 seconds Ping Peer Fail Count: 3 Ping Trap sending: Enabled Ping Peer Status: Not Tested Rewrite Request-uri: Disabled Registration Monitor: Disabled DTMF SIP NOTIFY Relay: Enabled DTMF SIP NOTIFY Interval: 2000 DTMF SIP default duration: 200 DTMF Preferred Method: SIP NOTIFY Realm : cisco.com Statistics setting: Disabled

Configuring IP Realm Under an Adjacency—Unified Model



Managing Emergency Calls

With the excessive use of VoIP, it is important to understand the various situations that might need prioritized attention. One of the most important situations that must be handled immediately is emergency service numbers, for example, using VoIP to dial a 911 number, which requires immediate attention. It is imperative to provide dedicated and immediate high-priority access for emergency numbers and calls with the specified SIP resource priority header. To analyze and examine the number of emergency calls, the Emergency Call feature has been implemented on the Cisco ASR 1000 Series Router.

Feature History for Managing Emergency Calls

Release	Modification
Cisco IOS XE Release 2.5	H.323-to-SIP support for emergency calls was introduced on the Cisco ASR1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.2S	The Emergency Call Statistics feature was introduced on the Cisco ASR 1000 Series Routers.

Contents

This chapter contains the following sections:

- H.323 to SIP Support for Emergency Calls, page 10-1
- Overview: Emergency Call Statistics, page 10-2
- Restrictions for Emergency Calls, page 10-2
- Performance Impact, page 10-2

H.323 to SIP Support for Emergency Calls

Cisco Unified Border Element (SP Edition) supports H.323 to SIP call routing for emergency calls. Cisco Unified Border Element (SP Edition) routes voice and video calls according to the configured session routing policy. A call is categorized as "emergency" based on the dialed number or on the Resource-Priority header if it is originated on the SIP side. Based on the emergency categorization, special routing and Call Admission Control (CAC) logic is applied.

Overview: Emergency Call Statistics

Certain emergency numbers, such as 911, 999, and so on, need more priority than normal calls. Using the Session Border Controller (SBC), you can configure and define a category and assign a priority for emergency numbers. It is important to know how many such emergency calls are currently in progress, and under which category the calls have been classified.

The Emergency Call feature has been introduced to analyze the number of emergency calls that are assigned to a particular category or a specific priority. The scope of displaying the emergency call statistics varies. The emergency call statistics can either be displayed globally for the SBC system or for a specific adjacency. To display category-wise and priority-wise emergency calls globally for the SBC system, use the **show sbc sbcname sbe call-stats global emergence** command. To display category and priority-wise calls for a specific adjacency, use the **show sbc sbc sbc call-stats adjacency** *word* **emergence** command. The per-adjacency statistic calls displays both received and sent calls separately on that adjacency.

The emergency call identification can be performed with the help of different mechanisms:

- Dedicated call setup priority information, such as the SIP resource priority header.
- Analysis of dialled number for known emergency numbers, such as 911, 999, and so on.
- Configuration based on the CAC policy.



The Billing Manager must be enabled and in active state for displaying the emergency call statistics.

Restrictions for Emergency Calls

The following restriction is applicable to the Emergency feature:

• The display of emergency call statistics is always disabled if Billing Manager is not active.

Performance Impact

The Emergency Call feature requires the Billing Manager to be configured and in active state. The performance cost of running the emergency calls feature is 2%. When emergency call statistics are not requested, there is no performance cost, and only the Billing Manager needs to be configured and active. The base occupancy of this feature is less than 0.1%, and the per-call occupancy cost of this feature is less than 0.25%. When emergency call statistics are requested (global or per adjacency), the requested statistics are displayed within half a second.



Unexpected Source Address Alerting

You can configure Cisco Unified Border Element (SP Edition) to provide alerts for any unexpected source addresses that are received. After an unexpected source address is received, a log is created and a Simple Network Management Protocol (SNMP) trap is generated.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

To locate documentation for other commands that appear in this chapter, use the command reference master index, or search online.

Note

For Cisco IOS XE Release 2.4, this feature is supported in both the unified model and the distributed model.

Feature History for Unexpected Source Address Alerting

Release	Modification
Cisco IOS XE Release	This feature was introduced for the unified model on the Cisco ASR 1000
2.4	Series Aggregation Services Routers.

Contents

This module contains the following sections:

- Prerequisites—Implementing Unexpected Source Address Alerting, page 11-2
- Restrictions for Unexpected Source Address Alerting, page 11-2
- Unexpected Source Address Alerting, page 11-2
- Configuring Unexpected Source Address Alerting, page 11-3
- Examples of Configuring Unexpected Source Address Alerting, page 11-4

Prerequisites—Implementing Unexpected Source Address Alerting

The following prerequisite is required to implement the unexpected source address alerting feature:

Before implementing unexpected source address alerting, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions for Unexpected Source Address Alerting

Review the following restrictions for unexpected source address alerting:

- This configuration option should only be enabled on trusted networks where any single such instance might indicate a threat to network security.
- Alerts on the same flow are rate-limited as are the total number of alerts reported at any one time to ensure management systems are not flooded with reports. There is not a 1-to-1 correspondence between alerts and incorrect packets.
- Diagnosing and resolving the issue of rogue packets is beyond the scope of the Cisco Unified Border Element (SP Edition) function.
- Any and all packets from unexpected sources are dropped.

Unexpected Source Address Alerting

If a packet with unexpected source address/port is received by the data border element (DBE) on a media address, port, or (if applicable) Virtual Routing Forwarding (VRF) used by a current call, then the DBE creates a log and generates an SNMP trap on the appropriate media-flow-stats MIB.

The log (level 63) is output to the console automatically (by default). The log is a member of the MEDIA debug log group. The log includes the local address, port, and VRF where the packets were received and also the source address and port of the received packet.

An alert is generated the first time an unexpected packet is received on a port after the port is opened for a call. If additional unexpected packets are received on the same media port, additional alerts are generated. Any additional alerts are rate-limited. After the call is completed, the media port is assigned to a new call, and the state is reset. A new alert is then generated if any additional unexpected packets are subsequently received.

The SNMP trap that is generated will contain the following fields:

- The address and port where the unexpected packet was received.
- The address and port where the unexpected packet originated.

Configuring Unexpected Source Address Alerting

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. unexpected-source-alerting
- 5. end
- 6. show sbc sbc-name dbe media-flow-stats vrf vrf-name [ipv4 A.B.C.D [port] port number]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	unexpected-source-alerting	Sets alerting for unexpected source addresses.
	<pre>Example: Router(config-sbc-sbe)# unexpected-source-alerting</pre>	The no form of this command removes alerting for any unexpected source addresses that are received.
Step 5	exit	Exits SBE configuration mode and enters SBC configuration mode.
	Example: Router(config-sbc-sbe)# exit	

	Command or Action	Purpose
Step 6	end	Exits the SBC configuration mode and returns to Privileged EXEC mode.
	Example:	
	Router(config-sbc)# end	
Step 7	<pre>show sbc service-name dbe media-flow-stats vrf vrf-name [ipv4 A.B.C.D [port port-number]]</pre>	Displays detailed information about the media flow statistics configured on the DBE.
	Example: Router# show sbc mysbc dbe media-flow-stats vrf vpn3 ipv4 10.1.1.1 port 24000	

Examples of Configuring Unexpected Source Address Alerting

This section provides a sample configuration for configuring unexpected source address alerting including an example of the information added to the media flow statistics.

To configure unexpected source address alerting, use the following commands:

```
configure terminal
sbc mysbc
sbe
unexpected-source-alerting
end
```



DoS Prevention and Dynamic Blacklisting

Denial of Service (DoS) prevention and dynamic blacklisting is used by Cisco Unified Border Element (SP Edition) to block malicious endpoints from attacking the network.

Cisco Unified Border Element (SP Edition) monitors signaling traffic and dynamically detects potential attacks without disrupting the rest of the services that it provides. The attacks can then be blocked internally or externally.

DoS attacks are generally performed on Internet services to deny these services to others. They are usually aimed at the provider of the service, and are either purely malicious vandalism or part of an attempt at extortion.

Blacklisting is the process of matching inbound packets based on parameters, such as source IP addresses, and preventing the packets that match those parameters from being processed.

Dynamic blacklists put in place automatically (subject to a set of configurable constraints) by Cisco Unified Border Element (SP Edition) when it detects an attempt to disrupt traffic flowing through it. Dynamic blacklisting does not require management interference. It can occur within milliseconds of the start of an attack and can change and adapt as the attack changes providing immediate network protection.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XR Software Release, this feature is supported in the unified model only.

Feature History for DoS Prevention and Dynamic Blacklisting

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced in Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 3. 2S	Minor, major, and critical alert traps introduced. The policy-rejection event was renamed as cac-policy-rejection, and the routing-failure event was renamed as rtg-policy-rejection. The na-policy-rejection event was introduced.

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Contents

This chapter contains the following sections:

- Prerequisites for DoS Prevention and Dynamic Blacklisting, page 12-2
- Restrictions for DoS Prevention and Dynamic Blacklisting, page 12-2
- Information About DoS Prevention and Dynamic Blacklisting, page 12-2
- Overriding Dynamic Blacklisting Default Thresholds, page 12-4
- Dynamic Blacklisting Behavior, page 12-5
- How to Configure Dynamic Blacklisting, page 12-6
- Examples of Configuring, Removing, and Displaying Dynamic Blacklisting, page 12-10

Prerequisites for DoS Prevention and Dynamic Blacklisting

Following are the prerequisites are required for dynamic blacklisting:

- You must already have Cisco Unified Border Element (SP Edition) configured.
- You need to configure blacklisting to override default blacklisting thresholds when the SBE is configured and before you start using Cisco Unified Border Element (SP Edition).

Restrictions for DoS Prevention and Dynamic Blacklisting

The following are restrictions for DoS prevention and dynamic blacklisting:

- Only Session Initiation Protocol (SIP) traffic is analyzed. Attacks over H.323 are not protected. However, an attack over SIP may also result in H.323 traffic being blocked.
- Port specific blacklist configuration is not possible.

Information About DoS Prevention and Dynamic Blacklisting

Cisco Unified Border Element (SP Edition) monitors the following events as "reasons" for initiating DoS detection policies:

- **authentication-failure**—If Cisco Unified Border Element (SP Edition) is locally authenticating the UAs or peers, then any authentication failure will count as one event.
- **bad-address**—This event is generated when an unexpected source sends a packet that reaches Cisco Unified Border Element (SP Edition); the packet will be dropped.
- rtg-policy-rejection—This event is generated when traffic fails to find a match in the routing policy. In Cisco IOS XE Release 3.2S, the routing-failure event is renamed as rtg-policy-rejection.
- endpoint-registration— This event is generated when an endpoint is registering through Cisco Unified Border Element (SP Edition) and the registration is rejected.
- **corrupt-message**—This event is generated when a signalling message cannot be decoded by the application or contains a protocol exception/violation.

- cac-policy-rejection—This is a complex category because it monitors CAC policy failures, that
 is, a negative result from the CAC policy. This category includes rate, count, and bandwidth
 limits, and makes no distinction between them. In Cisco IOS XE Release 3.2S, the
 policy-rejection event is renamed as cac-policy-rejection.
- spam—Endpoints may send unwanted or spam calls (sometimes called Spam over Internet Telephony (SPIT)). Spam results from too many unexpected signaling messages. Examples of spam include receipt of a SIP response that does not match an earlier sent request, and receipt of excessive retransmissions of a SIP message.
- **na-policy-rejection**—This event is generated when there are repeated call rejections due to an invalid source number or destination number. This event is considered as a DoS attack.

There are two types of events that would cause blacklisting: low-level and high-level attacks.

Low-level attacks

An overwhelming volume of traffic sent at line rate to devices that perform a significant amount of processing per packet.

• High-level attacks

Attacks on any bottlenecks within the signaling plane or application layers.

Blacklist enablement is defined as 'When an 'E'vent (for example, authentication-failure) that is being monitored, occurs exceeding the 'N'umber of times configured (trigger-size <>) within the 'W'indow (trigger-period <>), then activate the dynamic access control list for a 'T'ime period (timeout <>).

Any given endpoint can have up to three blacklisted events being monitored at a given time on a per-port, per-address, and per-VPN basis. Within the address source type, there is the following order of precedence:

- Limits configured per specific IPv4 address
- Default limits of the parent VRF address space
- Default limits of the global address space (if different from the parent VRF)
- The hard-coded address limits.

The SBC packet filter (SPF) is a new component designed to defend against low-level attacks. The SPF resides with the Media Packet Forwarder (MPF) component on the network processing unit (NPU) and provides low-level DoS prevention for standalone data border element (DBE) and unified SBC deployment scenarios.

A new component is added to the signaling border element (SBE) to detect high-level attacks and create dynamic blacklists based on these attacks. The dynamic blacklist is configured using the command line interface (CLI). It receives events from other SBE components and generates alerts to start or stop the blacklisting of certain messages. Events that might form part of a high-level attack are detected by other SBE components and sent to the SBE Dynamic Blacklisting Component to collects statistics on their rate of occurrence.

Blacklist Alert Traps

From Cisco IOS Release XE 3.2S, the blacklist settings are configured to implement alert traps. Minor, major, and critical traps are set to be triggered at much lower thresholds values. Blacklist alert traps do not cause any loss of service and not only generate a log message when the threshold is exceeded, but also an SNMP trap, if configured. To enable SNMP SBC blacklist traps, use the **snmp-server enable traps sbc blacklist** command.

These traps can be monitored and modified to detect a DoS attack.

Overriding Dynamic Blacklisting Default Thresholds

Dynamic blacklisting is on by default. Default thresholds are set for Trigger Size, Trigger Period, and Blacklisting Period for each reason. A reason may be an Authentication Failure, Bad Address, Routing Failure, Endpoint Registration, Corrupt Message, Spam, Routing Policy Rejection, or Number Analysis Policy Rejection.

We highly recommend you configure blacklisting to override default thresholds for call setup and registration messages at the time the SBE is configured and before you start using Cisco Unified Border Element (SP Edition). Doing this will ensure that your planned call setup rate or registration message rate does not trigger spam blacklist that will impede traffic flow. It is important to configure the call setup or registration messages thresholds to be above the messages or registration messages per second rate for each SIP-based call in order for traffic to flow through properly. The default values for Trigger Size, Trigger Period, and Blacklisting Period are 40 events per second, or 4 events per 100 milliseconds. This means that traffic over 40 packets per second would trigger blacklisting.

For the following SIP-based call flow, this example describes how to calculate a suitable trigger size threshold for call setup messages per second:

SIP-based call (caller) has: Send INVITE Receive 100 Trving Receive 180 Ringing Receive 200 OK to confirm Session Establishment Send ACK to complete Session Establishment Send BYE Receive 200 OK SIP-based call (callee) has: Send INVITE Send 100 Trying Send 180 Ringing Send 200 OK to confirm Session Establishment Receive ACK to complete Session Establishment Receive BYE Send 200 OK

There are 14 messages or packets for each SIP-based call. If you have a call setup rate of up to 20 calls per second (CPS), then 14 messages x 20 CPS = 280 messages per second. Therefore for a call setup rate of up to 20 CPS, you would configure a trigger size threshold of at least 280 messages per second.

In the following configuration example, you have raised the trigger size to 280 messages or packets per second:

```
blacklist global
reason spam
trigger-size 280
trigger-period 1 seconds
```

Similar to calculating call set up messages per second, the following example describes how to calculate a suitable trigger size threshold for registration messages:

There is one message per registration per second for each SIP-based call. If you have 20 registrations per second, then 1 messages x 20 registrations = 20 messages per second. Therefore for a registration rate of up to 20 registrations per second, you would configure a trigger size threshold of at least 20 messages per second.

Although Dynamic Blacklisting is on by default, you can turn it off by setting the timeout for every reason to zero. However, note that when timeout is set to zero for any unit value, such as milliseconds or seconds, the unit value returned in a **show run** command displays as "day." You can use the **show sbc sbe blacklist configured-limits** command to display the default trigger-size, trigger-period and timeout and configured limits. See ?\$paranum>Examples of Using the show Commands with Blacklisting? section on page 12-11 for an example of this command.

Dynamic Blacklisting Behavior

The following is a description of dynamic blacklisting behavior:

- A global rate limit is applied to ensure that the overall load across all sources and destinations does not exceed the CPU capacity (the default limiter 8000 pps/1000 Mbps).
- The hard-coded initial settings for each event type on each IP address are configured by default to hold 4 events for 100 milliseconds. If the configured values are exceeded, the IP address is blacklisted for 10 minutes.
- If you have an explicitly configured limit for a single IP address or port, any trigger and blocking time values defined in that configuration will override the default. Table 12-1 displays where the parameters of the event limits at each scope for a given message can be configured. The limits are different if the message source is on a global address space or VPN.
- Media packets must match a valid entry in the flow table or they are dropped.

Scope of Event	Event Limit Parameter Sources (Highest Priority First)						
Limit	Global Address Space	VPN					
Port	1. Explicit limit for this port	1. Explicit limit for this port					
	2. Default for this IP address	2. Default for this IP address					
Address	1. Explicit limit for this address	1. Explicit limit for this address					
	2. Default for global IP addresses	2. Default for addresses on this					
	3. Hard-coded initial settings	VPN					
		3. Default for global IP addresses					
		4. Hard-coded initial settings					
VPN	Explicit limit for the global address space.	1 . Explicit limit for this VPN					
		2. Limit set for the global address space					

Table 12-1 Priority of Event Limit Parameters

- Valid media packets must not exceed bandwidth limits established in call signaling. Non-conferment packets are dropped.
- Signaling packets are rate-limited by the source port in an attempt to halt forceful packet floods early (the default limiter is 1000 pps/100 mpbs).
- Signaling packets that are not destined to a valid local port are dropped.
- Signaling packets are rate-limited by destination port (the default limiter is 4000 pps/500 Mbps).
- Limits can be configured for specific events from the following source(s): a VPN ID, an IP address, or a port at a specific IP address.

• Default limits on event rates may be defined for all source IP addresses on a VPN, and for all ports on a given IP address. The default limits on each IP address are automatically set at the start of the day, but their parameters can be reconfigured. By default, no event limits are configured for ports.

Cisco Unified Border Element (SP Edition) monitors events per IP address by default. You can also configure Cisco Unified Border Element (SP Edition) to monitor an entire VPN or a particular port. If any limit in a VPN is then exceeded, the entire VPN is blacklisted. If a limit for a port is exceeded, the port and its IP address are blacklisted.

- Packets are classified as either signaling or media according to the port from where they are sent:
 - Ports below 10,000 are signaling.
 - Ports above 10,000 are media.
- When only a global address space blacklist is defined (no VRF specific blacklist), this will be used to blacklist addresses in all configured VRFs.
- VRF based blacklist limits will override any per source or address-default limits already set. You cannot use per IP address scope to override behavior in VRF space.
- Cisco Unified Border Element (SP Edition) generates an SNMP trap when a blacklist is activated.

How to Configure Dynamic Blacklisting

You can configure dynamic blacklisting as explained in the following sections:

- Configuring Blacklist Parameters for an IP Address, Port, or VPN, page 12-6
- Configuring an End to Blacklisting, page 12-9

Configuring Blacklist Parameters for an IP Address, Port, or VPN

To configure the event limits for a specific source, use the following commands.



You must configure blacklisting to override the default blacklisting thresholds when the SBE is configured, and before you start using Cisco Unified Border Element (SP Edition).

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. blacklist ipv4 addr
- 5. description text
- 6. reason event
- 7. trigger-size number
- 8. trigger-period time
- 9. critical-alert-size number-of-events
- **10. major-alert-size** *number-of-events*

- 11. minor-alert-size number-of-events
- **12. timeout** *timeframe*
- 13. end
- 14. show sbc service-name sbe blacklist configured-limits
- 15. show sbc service-name sbe blacklist source
- 16. show sbc service-name sbe blacklist current-blacklisting

DETAILED STEPS

	Command or Action	Purpose		
Step 1	configure terminal	Enables global configuration mode.		
	Example: Router# configure			
Step 2	sbc service-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters the SBC configuration mode.		
	Example: Router(config)# sbc mysbc			
Step 3	sbe	Enters the SBE entity mode within an SBC service.		
	Example: Router(config)# sbe			
Step 4	blacklist ipv4 addr	Enters the blacklist submode for configuring the event limits for a given source.		
	Example: Router(config)# blacklist ipv4 25.25.25.5	The no form of this command changes the event limits that have been configured to default values.		
		Note Event limit parameters that are not configured in this submode are configured with the default, as follows:		
		- port—port-default value for its address.		
		- IP address—address-default value for the VPN.		
		- VPN—value for the global address space.		
		- global address space—no limit.		
Step 5	description text	Adds a description for the source and its event limits using a readable text string format.		
	Example:	The no form of this command removes the description.		
	Router(config-sbc-sbe-blacklist)# description NAT of XYZ Corp	This description is displayed when the show command is used for this source.		

	Command or Action	Purpose					
Step 6	reason event	Enters the reason submode for configuring a limit for a specific event type on the source.					
	Example: Router(config-sbc-sbe-blacklist)# reason	The no form of this command returns the event limit to its default values. An event includes:					
	authentication-failure						
		• authentication-failure—Requests that fail authentication.					
		• bad-address—Packets from unexpected addresses.					
		• rtg-policy-rejection—Requests that fail to be routed by SBC.					
		• endpoint-registration—All endpoint registrations.					
		• cac-policy-rejection—Requests that are rejected by the CAC policy.					
		 corrupt-message—Signaling packets that are too corrupt to be parsed by the relevant protocol. 					
		 na-policy-rejection—Requests that are rejected by configured number analysis policy. 					
Step 7	trigger-size number	Defines the number of events from the specified source that are allowed before the blacklisting is triggered and all packets are blocked from the source.					
	Example: Router(config-sbc-sbe-blacklist-reason# trigger-size 5	Range can be 0 to 65535,					
Step 8	trigger-period time	Defines the period of time that events are considered.					
	Example: Router(config-sbc-sbe-blacklist-	<i>time</i> is expressed as <i>number unit</i> where <i>number</i> is an integer and <i>unit</i> is one of: milliseconds, seconds, minutes, hours, or days.					
	reason)# trigger-period 20 milliseconds	Default period of time is between 10 milliseconds and 23 days.					
Step 9	timeout time	Defines the length of time when packets from the source are blocked if the configured limit is exceeded.					
	Example:	time can have the following values:					
	Router(config-sbc-sbe-blacklist- reason)# timeout 180 seconds	• 0 = the source is not blacklisted					
		• never = the blacklisting is permanent					
		• <i>number unit</i> where <i>number</i> is an integer and <i>unit</i> is seconds, minutes, hours, or days					
		Default period of time is less than 23 days.					
Step 10	critical-alert-size number-of-events	Defines the number of specified events that must occur before the critical alert is triggered.					
	Example: Router(config-sbc-sbe-blacklist- reason)# critical-alert-size 655	<i>number-of-events</i> can have any value ranging from 1 to 65535.					

	Command or Action	Purpose
Step 11	major-alert-size number-of-events	Defines the number of specified events that must occur before the major alert is triggered.
	Example: Router(config-sbc-sbe-blacklist- reason)# major-alert-size 300	<i>number-of-events</i> can have any value ranging from 1 to 65535.
Step 12	<pre>minor-alert-size number-of-events</pre>	Defines the number of specified events that must occur before the minor alert is triggered.
	Example: Router(config-sbc-sbe-blacklist- reason)# minor-alert-size 20	<i>number-of-events</i> can have any value ranging from 1 to 65535.
Step 13	end	Exits the reason mode and enters Privileged EXEC mode.
	Example: Router(config-sbc-sbe-blacklist-reason)# end	
Step 14	<pre>show sbc service-name sbe blacklist configured-limits</pre>	Displays detailed information about the explicitly configured limits.
	Example: Router# show sbc mysbc sbe blacklist global configured-limits	Any values not explicitly defined for each source are displayed in brackets.
Step 15	show sbc service-name sbe blacklist source	List the limits that are currently in place for a specific source (in this example, VPN). This includes any defaults or explicitly configured limits.
	Example: Router# show sbc mysbc sbe blacklist vpn3 ipv4	It also includes any defaults of a smaller scope that are configured at this address.
	172.19.12.12	Any values that are not explicitly configured are bracketed (these are the values that are inherited from other defaults).
Step 16	<pre>show sbc service-name sbe blacklist current-blacklisting</pre>	Lists the limits that are causing the source(s) to be blacklisted.
	Example: Router# show sbc mysbc sbe blacklist current-blacklisting	

Configuring an End to Blacklisting

Use the following command to remove the source from the blacklist:

• clear sbc service-name sbe blacklist source

For the *service-name* parameter, enter the name of the SBC.

For the source parameter enter the name of the blacklist.

Examples of Configuring, Removing, and Displaying Dynamic Blacklisting

This section provides a sample configuration and output for dynamic blacklisting, removing a source from being blacklisted, and also displaying configured limits.

Example of Configuring Dynamic Blacklisting

This blacklist is configured for global address space with one authentication failure from all possible address sources to be captured within a 100 milliseconds window. The ACL created (blacklist) should never timeout.

```
Router(config-sbc-sbe)# blacklist global
Router(config-sbc-sbe-blacklist)# address-default
Router(config-sbc-sbe-blacklist-addr-default)# reason authentication-failure
Router(config-sbc-sbe-blacklist-addr-default)# timeout never
Router(config-sbc-sbe-blacklist-addr-default)# trigger-size 1
Router(config-sbc-sbe-blacklist-addr-default)# trigger-period 100 milliseconds
```

This blacklist is configured for global address space, five packets from unexpected source within a one minute window. The ACL is to time out in 24 hours.

```
Router(config-sbc-sbe)# blacklist global
Router(config-sbc-sbe-blacklist)# ipv4 10.5.1.21
Router(config-sbc-sbe-blacklist-ipv4)# reason bad-address
Router(config-sbc-sbe-blacklist-ipv4)# timeout 1 days
Router(config-sbc-sbe-blacklist-ipv4-reason)# trigger-size 5
Router(config-sbc-sbe-blacklist-ipv4-reason)# trigger-period 1 minutes
```

Example of Removing a Source from the Blacklist

The following example shows the syntax for removing blacklist from Cisco Unified Border Element (SP Edition):

```
Router# clear sbc mysbc sbe blacklist blacklist
Router#
```

Example of Displaying All the Configured Limits

The following example shows the configured limits for various types of blacklisting:

```
Router# show sbc uut105-1 sbe blacklist configured-limits
```

SBC Service "uut105-1"

Blacklist Defaults

================							
Reason	Trigger	Trigger	Blacklisting	Minor	Major	Critical	
	Size	Period	Period	Alert	Alert	Alert	
Auth-failure	(4)	(100 ms)	(10 mins)	not set	not set	not set	
Bad-address	(4)	(100 ms)	(10 mins)	not set	not set	not set	
RTG-policy-rejec	ction (4)	(100 ms)	(10 mins)	not set	not set	not set	
Endpoint-registr	ation (4)	(100 ms)	(10 mins)	not set	not set	not set	
CAC-policy-reject	ction (4)	(100 ms)	(10 mins)	not set	not set	not set	
Corrupt-message	(4)	(100 ms)	(10 mins)	not set	not set	not set	

(30) (100 ms) (10 mins) not set not set not set Spam NA-policy-rejection (4) (100 ms) (10 mins) not set not set not set VRF: 172.18.53.56 _____ Trigger Trigger Blacklisting Minor Major Critical Size Period Period Alert Alert Alert Reason not set NA-policy-rejection (4) (100 ms) (10 mins) 2 not set _____ Router#

Examples of Using the show Commands with Blacklisting

The following example shows the command required to list the limits that are currently in place for a specific source (in this example, VPN). This includes any defaults or explicitly configured limits. It also includes any defaults of a smaller scope that are configured at this address. Any values that are not explicitly configured are bracketed (these are the values that are inherited from other defaults).

Router# show sbc mysbc sbe blacklist vpn3 ipv4 172.19.12.12

SBC Service "mySbc" SBE dynamic blacklist vpn3 172.19.12.12

	vpn3	1	7	2		19		12		12
--	------	---	---	---	--	----	--	----	--	----

=======================================	==			
Reason	Trigger Size	Trigge Perio	er od	Blacklisting Period
Authentication	(20)	10 r	ms	(1 hour)
Bad address	(20)	10 r	ms	(1 hour)
Routing	(20)	10 r	ms	(1 hour)
Registration	(5)	100 r	ms	(10 hours)
Policy	(20)	10 r	ms	(1 day)
Corrupt	40	10 r	ns	(1 hour)

Default for ports of vpn3 172.19.12.12

		===========	
Reason	Trigger Size	Trigger Period	Blacklisting Period
Authentication	20	1 sec	1 hour
Bad address	20	1 sec	1 hour
Routing	20	1 sec	1 hour
Registration	5	30 sec	10 hours
Policy	20	1 sec	1 day
Corrupt	20	100 ms	1 hour

The following example shows the command required to list the limits that are causing the source(s) to be blacklisted:

Router# show sbc mysbc sbe blacklist current-blacklisting

SBC Service "mySbc" SBE dynamic blacklist current members

	-			
Address	5	Port	Reason	Remaining
Source		Source	Blacklist	Time
======				
Global	addresses			

125.125.111.123 125.125.111.253 144.12.12.4	All UDP 85 TCP 80	Authentication Registration Corruption	15 mins 10 secs Never ends
VRF: vpn3			
========			
Source	Source	Blacklist	Time
Address	Port	Reason	Remaining
132.15.1.2	TCP 285	Registration	112 secs
172.23.22.2	A11	Policy	10 hours

The following example shows the configured limits:

```
Router# show sbc MySBC sbe blacklist configured-limits
```

```
SBC Service "MySBC"
```

Blacklist Defaults _____ Trigger Trigger Blacklisting Minor Major Critical Reason Size Alert Period Period Alert Alert Auth-failure(4)(100 ms)(10 mins)not setnot setBad-address(4)(100 ms)(10 mins)not setnot setRTG-policy-rejection(4)(100 ms)(10 mins)not setnot setEndpoint-registration(4)(100 ms)(10 mins)not setnot set not set not set not set not set CAC-policy-rejection (4) (100 ms) (10 mins) not set not set not set Corrupt-message (4) (100 ms) (10 mins) not set not set not set Spam (30) (100 ms) (10 mins) not set not set not set NA-policy-rejection (4) (100 ms) (10 mins) not set not set not set -----VRF: 172.18.53.56 _____ Trigger Blacklisting Minor Major Critical Trigger Reason Period Period Alert Alert Alert Size NA-policy-rejection (4) (100 ms) (10 mins) 2 not set not set _____



Watch out for the default configurations already in effect. Only the applied configurations are modified.

This example shows current blacklisting:

Router# show sbc MySBC sbe blacklist current-blacklisting

SBC Service "MySBC" SBE dynamic blacklist current members

Global addresses			
=======================================			
Source	Source	Blacklist	Time
Address	Port	Reason	Remaining
10.5.1.31All	Authent	ication Forever	



Implementing Interworking DTMF

Cisco Unified Border Element (SP Edition) enables interworking between in-channel real-time transport protocol (RTP) signaling using the audio/telephone-event MIME type (RFC 2833) to and from out-of-band signaling using the SIP INFO or SIP NOTIFY method.

The Dual Tone Multifrequency (DTMF) Method Interworking and ACCEPT Header Handling feature introduces an adjacency setting that modifies the only auto detection behavior for INFO method.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller, and may be commonly referred to as the session border controller (SBC) in this document.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Implementing Interworking DTMF on Cisco Unified Border Element (SP Edition)

Release	Modification
Cisco IOS XE Release 2.1	Interworking DTMF was introduced on the Cisco IOS XR.
Cisco IOS XE Release 2.4	Introduced support for DTMF Relay Using SIP NOTIFY Messages on the unified model.
Cisco IOS XE Release 3.1S	The DTMF Method Interworking and ACCEPT Header Handling feature support was added to Cisco ASR 1000 Series Router.

Contents

This chapter contains the following sections:

- Restrictions, page 13-2
- Prerequisites—Implementing Interworking DTMF, page 13-2
- Information About Interworking DTMF, page 13-2
- Implementing Interworking DTMF, page 13-4
- DTMF Relay Using SIP NOTIFY Messages, page 13-5
- DTMF Method Interworking and ACCEPT Header Handling, page 13-9

Restrictions

The following are restrictions of the Implementing Interworking DTMF feature:

- When the SBC inspects the accept header in the endpoint's messages, the absence of the accept header means "application/sdp" is supported.
- When audio transcoding is in operation, the SBC does not support sending and receiving RFC 2833 in-band packets to and from the SBC and interworking RFC 2833 packets with out-of-band SIP INFO or SIP NOTIFY Relay messages on the other call leg.
- The SBC does not support the scenario where a caller only supports sending RFC 2833 in-band packets to a callee that supports both RFC 2833 and out-of-band SIP INFO and SIP NOTIFY Relay. In this case, the DTMF digits received out-of-band on the callee side is not able to be translated into RFC 2833 packets on the caller side.
- The SBC does not support configurable outbound RFC 2833 payload type for SIP to SIP calls when the inbound call side does not support RFC 2833.

Prerequisites—Implementing Interworking DTMF

The following prerequisites are required to implement interworking DTMF:

Before implementing interworking DTMF, Cisco Unified Border Element (SP Edition) must already be configured.

Information About Interworking DTMF

Cisco Unified Border Element (SP Edition) automatically selects the best DTMF Interworking technique based on the combined capabilities of the endpoints in a call. See Figure 13-1 for a sample call flow.

The SBC supports the signaling of DTMF using the following modes:

- Media-stream signalling using RTP payload (RFC2833)
- INFO-based DTMF relay (RFC 2976)
- NOTIFY-based DTMF relay

The SBC can interwork between any of these modes using the most performance efficient methods.

DTMF interworking for RFC 2833 in-band packets when transcoding is not supported.

Inspection of the arriving INVITE helps determine the caller's support for DTMF interworking.

To determine whether the caller supports the INFO method, the SBC inspects the Allow header for the INFO method if the Allow header is present. However, the INVITE must also contain an Accept header that contains application/dtmf-relay for the SBC to detect support for DTMF in the INFO method.

Support for the unsolicited NOTIFY method can be determined by the presence of a Call-Info header indicating the NOTIFY method.

An INFO or NOTIFY message is expected to carry a single DTMF tone with an optional duration. If no duration is specified, the default is 250 milliseconds (ms) for an INFO message and 200 ms for a NOTIFY message.
If the SBC determines that either the INFO method or the NOTIFY method for DTMF is supported by the originator, support for both INFO and NOTIFY methods are advertised by the presence of Call-Info and Accept headers on the outbound call. Interworking between these methods is efficient and improves the probability of finding a suitable method for DTMF interworking.

In the case of interworking of DTMF relay using Network Terminating Equipment (NTE) (RFC 2833) and out-of-band DTMF using SIP INFO or SIP NOTIFY, Cisco Unified Border Element (SP Edition) intercepts the NTE packets with DTMF digits and converts them into the appropriate signaling methods through the Route Processor (RP). In the reverse direction, the RP instructs the Cisco Unified Border Element (SP Edition) to inject NTE DTMF packets into an RTP stream.

DTMF Packet Generation

When the NTE packets are to be inserted in the middle of a stream that is already sending RTP voice packets, then the NTE packets will replace the RTP voice packets in a one-to-one manner so that subsequent voice packets will not need to update their RTP sequence numbers.

DTMF Packet Detection

To detect DTMF NTE packets, Cisco Unified Border Element (SP Edition) looks at the payload type of every RTP packet and compares it with that of NTE. In case of a match, Cisco Unified Border Element (SP Edition) looks at the event number to determine that it is a DTMF digit. Cisco Unified Border Element (SP Edition) then copies these packets to the RP. Figure 13-1 illustrates this process.

Γ



Figure 13-1 Sample Call Flow with INFO (RFC-2833) DTMF Interworking

Implementing Interworking DTMF

The following section describes how to configure the default duration of a DTMF event.

Note that Cisco Unified Border Element (SP Edition) may require you to configure header Allow, header Accept, and method INFO as shown below:

```
sbc test
sbe
sip header-profile default
header Allow
header Accept
sip method-profile default
method INFO
```

Configuring Default Duration of a DTMF Event

This task configures the default duration of a DTMF event.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. dtmf-duration duration

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC called "mysbc" and enters into SBC configuration mode
	Example: Router(config)# sbc mysbc dbe	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	dtmf-duration duration	Configures the default duration of a DTMF event in milliseconds.
	Example: Router(config-sbc-vdbe)# dtmf-duration 300	

DTMF Relay Using SIP NOTIFY Messages

In Cisco IOS XE Release 2.4, Cisco Unified Border Element (SP Edition) adds support for DTMF Relay Using SIP NOTIFY Messages. This is an out-of-band procedure for DTMF relay and is sometimes referred to as NOTIFY-based DTMF Relay.

DTMF tones are the tones that are generated when a telephone key is pressed on a touchtone phone. Sometimes the called endpoint needs to hear those tones, such as when you enter digits during the call in response to a menu. However, low-bandwidth codecs can distort the sound. DTMF relay allows that tone information to be reliably passed from one endpoint to the other. By default, SIP uses in-band signaling, sending the DTMF information in the voice stream. If no DTMF relay method is configured, the tones are sent in-band. However, you can configure DTMF relay to use SIP NOTIFY messages for transmitting DTMF tone information. Cisco Unified Border Element (SP Edition) supports two out-of-band procedures for DTMF relay. One uses SIP INFO methods, and the other uses SIP NOTIFY methods. The SIP INFO method sends DTMF digits in INFO messages. It is always enabled. When a gateway receives an INFO message containing DTMF relay information, it sends the corresponding tone.

SIP NOTIFY DTMF relay is negotiated by including a Call-Info field in the SIP INVITE and response messages, on a per-adjacency basis. This field indicates an ability to use NOTIFY for DTMF tones and the duration of each tone in milliseconds. When a DTMF tone is generated, the caller sends a NOTIFY message to the callee. When the callee receives the NOTIFY, it responds with SIP 200 OK and plays the DTMF tone.



For Cisco IOS XE Release 2.4 and later, this feature is supported in the unified model only.

You can configure a preferred SIP signaling DTMF transport method for endpoints on an adjacency. If Cisco Unified Border Element (SP Edition) has received DTMF information on a call and is sending it to an endpoint on the adjacency, Cisco Unified Border Element (SP Edition) uses a preferred DTMF method to send the information, provided the endpoint supports this method. You can set one of the following DTMF relay methods as the preferred method:

- SIP NOTIFY DTMF Relay (default value)
- SIP INFO DTMF Relay

Use the **dtmf prefer sip** [info | notify] command to configure the preferred relay method.

The default on the Cisco Unified Border Element (SP Edition) is the SIP-NOTIFY relay method. However, Cisco Unified Border Element (SP Edition) uses the RTP-NTE in-band DTMF relay method if the other side does not support SIP-NOTIFY. If no DTMF relay method is configured, the tones are sent in-band.

When SIP NOTIFY relay is enabled on an adjacency, then:

- The SBC accepts in-call, out-of-subscription NOTIFY messages with a DTMF Payload. These messages are not required to contain a Subscription-State header.
- The SBC accepts a Call-Info header in an INVITE message specifying a telephone-event that indicates support for SIP NOTIFY DTMF Relay.
- Configure the NOTIFY interval. You need to configure the maximum interval in milliseconds that the SBC waits between NOTIFY messages for a single DTMF event.

In this case, the SBC has not received an inbound Call-Info header specifying the negotiated duration, so this value is used instead.

Use the **dtmf sip notify interval** command.

• You can also configure a default duration. This specifies the duration in milliseconds that the SBC advertises on the outbound DTMF transport method if the inbound side of the call does not supply a duration.

Use the **dtmf sip default duration** command.

Configuring Default Duration of a SIP NOTIFY DTMF Relay Event

This task configures parameters for a SIP NOTIFY DTMF Relay:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. dtmf prefer sip {*info* | *notify*}
- 6. dtmf sip notify interval int_ms
- 7. dtmf sip default duration *dur_ms*
- 8. end
- 9. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures an adjacency on the SBC and enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip SoftSwitch	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	<pre>dtmf prefer sip {info notify}</pre>	(Optional) Configure SIP NOTIFY DTMF relay as the preferred DTMF transport method for endpoints on this
	Example: Router(config-sbc-sbe-adj)# dtmf prefer sip notify	adjacency. This is optional because SIP NOTIFY is the default on the SBC.

	Command or Action	Purpose
Step 6	<pre>dtmf sip notify interval int_ms</pre> Fxample:	(Optional) Configures the maximum interval in milliseconds that the SBC waits between NOTIFY messages for a single DTMF event.
	Router(config-sbc-sbe-adj)# dtmf sip notify 1000	<i>int_ms</i> is the duration in milliseconds (ms.) The range is 1 to 65535 ms. The default is 2000 ms.
Step 7	dtmf sip default duration dur_ms	(Optional) Specifies the duration in milliseconds that the SBC advertises on the outbound DTMF transport method if the inbound side of the call does not supply a duration.
	Router(config-sbc-sbe-adj)# dtmf sip default duration 300	dur_ms is the duration in milliseconds (ms). The range is 1 to 65535 ms. The default is 200 ms.
Step 8	end	Exits sip adjacency configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj)# end	
Step 9	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Display all the fields in the specified SIP adjacency, including that SIP NOTIFY relay is enabled, the interval and default duration in milliseconds.
	Example: Router# show sbc mySBC sbe adjacencies SoftSwitch detail	

SIP NOTIFY Examples

The following example disables SIP NOTIFY relay for adjacency ADJ2 and configures SIP INFO as the preferred DTMF Relay method:

```
configure terminal
sbc mySbc
sbe
adj sip ADJ2
dtmf disable sip notify
dtmf prefer sip info
dtmf sip default duration 330
```

The following example displays all the fields in the SoftSwitch SIP adjacency, showing that the SIP NOTIFY relay method is enabled, and the interval and default duration in milliseconds:

```
router# show sbc mySBC sbe adjacencies SoftSwitch detail
SBC Service "mySBC"
  Adjacency SoftSwitch (SIP)
    Status:
Signaling address:
    Status:
                                 Attached
                               100.100.100.100:5060, VRF Admin
                               10.10.51.10:5060
    Signaling-peer:
    Force next hop:
                                No
    Account:
    Group:
                                None
    In header profile: Default
Out header profile: Default
    In method profile:
Out method profile:
                                Default
                                Default
    In UA option prof:
                                Default
    Out UA option prof:
                               Default
    In proxy opt prof:
                                 Default
```

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Out proxy opt prof:	Default
Priority set name:	None
Local-id:	None
Rewrite REGISTER:	Off
Target address:	None
Register Aggregate:	Disabled
NAT Status:	Auto Detect
Reg-min-expiry:	30 seconds
Fast-register:	Enabled
Fast-register-int:	30 seconds
Authenticated mode:	None
Authenticated realm:	None
Auth. nonce life time:	300 seconds
IMS visited NetID:	None
Inherit profile:	Default
Force next hop:	No
Home network Id:	None
UnEncrypt key data:	None
SIPI passthrough:	No
Rewrite from domain:	Yes
Rewrite to header:	Yes
Media passthrough:	No
Preferred transport:	UDP
Hunting Triggers:	Global Triggers
Redirect mode:	Pass-through
Security:	Untrusted
Outbound-flood-rate:	None
Ping-enabled:	No
Signaling Peer Status:	Not Tested
DTMF SIP NOTIFY Relay:	Enabled
DTMF SIP NOTIFY Interval:	1000 ms
DTMF SIP default duration:	300 ms
DTMF Preferred Method:	SIP NOTIFY

s

DTMF Method Interworking and ACCEPT Header Handling

The SBC can be configured to perform the following functions to support the INFO method in any circumstance:

- Automatically detect support for DTMF in INFO (default behavior).
- Does not send DTMF in INFO, and rejects DTMF in INFO, if received. ٠
- Accepts that DTMF in INFO is supported, regardless of the indication in the Accept header.

Auto detection does not detect support for DTMF-based relay in the following events:

- The Allow header contains the INFO method, but does not have the Accept header.
- The Accept header is present, but does not contain the application/dtmf-relay information. ٠ Therefore, the SBC can be configured to assume support of DTMF in INFO.

Configuring DTMF Relay in INFO Message

By default, auto detection of support for DTMF-based relay in INFO message occurs, and therefore, no configuration is required. However, to override auto detection so that support for this method is always assumed, not considering the arriving INVITE message.

This section contains information about the following configurations:

- Configuring SBC to Assume Support for INFO-Based DTMF Relay, page 13-10
- Configuring SBC to Disable INFO-Based DTMF Relay, page 13-11

Configuring SBC to Assume Support for INFO-Based DTMF Relay

This task configures parameters to always assume support for INFO-based DTMF relay.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. dtmf sip info always-supported
- 6. end
- 7. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures an adjacency on the SBC and enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip SoftSwitch	Use the <i>adjacency-name</i> argument to define the name of the service.

	Command or Action	Purpose
Step 5	dtmf sip info always-supported	(Optional) Assumes the INFO method as the preferred DTMF transport method for the endpoints on the adjacency.
	Example: Router(config-sbc-sbe-adj)# dtmf sip info always-supported	Note Use the no dtmf sip info command to turn on auto detection of DTMF support.
Step 6	end	Exits the SIP adjacency configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the fields in the specified SIP adjacency.
	Example: Router# show sbc mySBC sbe adjacencies SoftSwitch detail	

Configuring SBC to Disable INFO-Based DTMF Relay

This task configures parameters to permanently disable support for DTMF-based relay in INFO.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. dtmf disable sip info
- 6. end
- 7. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySBC	

	Command or Action	Purpose
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures an adjacency on the SBC and enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip SoftSwitch	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	dtmf disable sip info Fxamnle:	(Optional) Turns off the automatic detection of DTMF relay using the INFO method as the preferred DTMF transport method for the endpoints on the adjacency.
	Router(config-sbc-sbe-adj)# dtmf disable sip info	Note Use the no form of this command to turn on auto detection of DTMF relay.
Step 6	end	Exits the SIP adjacency configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the fields in the specified SIP adjacency.
	Example: Router# show sbc mySBC sbe adjacencies SoftSwitch detail	

DTMF Relay Using SIP INFO Message Examples

The following example shows how to configure the SBC to always assume support for INFO-based DTMF relay:

```
configure terminal
sbc mySbc
sbe
adjacency sip adj1
dtmf sip info always-supported
```

The following example shows how to configure SBC to disable support for DTMF-based relay in INFO permanently:

```
configure terminal
sbc mySbc
sbe
adjacency sip adj1
dtmf disable sip info
```

The following example shows the output of the **show sbc sbe adjacencies detail** command. It also shows that the SBC is configured to always assume support for INFO-based DTMF relay:

```
Router# show sbc asr1k-sbc sbe adjacencies sipp-1 detail
SBC Service "asr1k-sbc"
Adjacency sipp-1 (SIP)
Status: Attached
```

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Signaling address: 10.10.100.120:5080 IPsec server port: 0 10.10.100.10:10000 Signaling-peer: Signaling-peer status: Not Tested Signaling-peer priority: 2147483647 Signaling-peer switch: always Peer status: Not Tested Current peer index: 0 Force next hop: No Force next hop select: Out-of-dialog Admin Domain: Account: Group: None In header profile: Default Out header profile: Default In method profile: Default Out method profile: Default Out error profile: Default In body profile: None Out body profile: None In UA option prof: Default Out UA option prof: Default In proxy opt prof: Default Out proxy opt prof: Default Priority set name: None Local-id: None Rewrite REGISTER: Off Register contact username: Rewrite Target address: None NAT Status: Auto Detect 3000 seconds Reg-min-expiry: Fast-register: Enabled Fast-register-int: 30 seconds SoftSwitch-shield: Disabled Expires-header: add-not-present Register aggregate: Disabled Registration Required: Disabled Register Out Interval: 0 seconds Parse username params: Disabled Supported timer insert:Disabled Suppress Expires: Disabled p-asserted-id header-value: not defined p-assert-id assert: Disabled Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default Force next hop: No Home network Id: None UnEncrypt key data: None SIPI passthrough: No Passthrough headers: Media passthrough: Yes Incoming 100rel strip: No Incoming 100rel supp: No Out 100rel supp add: No Out 100rel req add: No Parse TGID parms: No IP-FODN inbound: IP-FQDN outbound: FQDN-IP inbound: FQDN-IP outbound: Outbound Flood Rate: None

Hunting Triggers: Global Triggers Add transport=tls param: Disabled Redirect mode: Pass-through Security: Untrusted-Unencrypted Privacy: Inherit-profile (default) TLS mutual authentication: No Ping: Disabled Ping Interval:32 secondsPing Life Time:32 secondsPing Peer Fail Count:3Ping Trap sending:Enabled Ping Suppression: Ping Bad Response Code: 300-399 Ping Peer Status: Not Tested Rewrite Request-uri: Disabled Registration Monitor: Disabled DTMF SIP INFO Relay: Always supported DTMF SIP NOTIFY Relay: Enabled DTMF SIP NOTIFY Interval: 2000 DTMF SIP default duration: 200 DTMF Preferred Method: SIP NOTIFY None Realm: Statistics setting: Summary IMS RX: Disabled Enabled IMS Rf: IMS Nass: Disabled IMS realm name: PANI: Warrant Match-Order: None Media Bypass Max Out Data Length: 1000 Media Bypass Tag List: None



Cisco Unified Border Element (SP Edition) Redundancy—High Availability Support

This chapter describes high availability support for Cisco Unified Border Element (SP Edition) on the Cisco ASR 1000 Series Aggregation Services Routers.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

N, Note

For Cisco IOS XE Release 2.4 and later, this feature is supported in both the unified model and the distributed model.

Feature History for Cisco Unified Border Element (SP Edition) Redundancy—High Availability Support

Release	Modification
Cisco IOS XE Release 2.4	Added support for Cisco Unified Border Element (SP Edition) unified model.
Cisco IOS XE Release 2.1	This feature was introduced on the Cisco IOS XR on the data border element (DBE) for the distributed model.

Contents

This chapter contains the following sections:

- Integrated Session Border Controller High Availability, page 14-2
- Hardware Redundancy, page 14-2
- Software Redundancy, page 14-2
- Route Processor Redundancy (RPR), page 14-3
- SSO Support, page 14-3
- ISSU Support, page 14-4

Integrated Session Border Controller High Availability

The Cisco ASR 1000 Series Routers include the Cisco ASR 1002, Cisco ASR 1004, and Cisco ASR 1006 Routers. The different models support different types of redundancy. Integrated Session Border Controller supports the redundancy available on each model.

On the Cisco ASR 1002 and Cisco ASR 1004 Routers, only software redundancy is available. These routers have dual Cisco IOS software modules running on the same Route Processor, with one active and the other in standby mode. However, these routers can have hardware redundancy by using interchassis hardware redundancy.

The Cisco ASR 1006 Routers offer dual hardware redundancy and software redundancy.

Cisco Unified Border Element (SP Edition) high availability is provided in the standard image for the Cisco ASR 1000 Series Routers. There is no special configuration required.

For additional information, see the "High Availability Overview" section in the *Cisco ASR 1000 Series Aggregation Services Routers Software Configuration Guide*. Also see the *Cisco IOS High Availability Configuration Guide* for information on high availability features that are on other Cisco platforms and that work identically on the Cisco ASR 1000 Series Aggregation Services Routers.

Hardware Redundancy

Integrated Session Border Controller supports use of a redundant or standby Route Processor (RP) and redundant Embedded Services Processor (ESP) on the Cisco ASR 1006 Router. The Cisco ASR 1006 Router has an ESP as well as an RP for dual hardware redundancy. If the active RP or active ESP hardware fails, the system performs a switchover to the standby RP or standby ESP. RP and ESP hardware redundancy support is independent. An RP failure does not require a switchover of the ESP hardware and an ESP failure does not require an RP switchover.

Hardware redundancy is available only on the Cisco ASR 1006 Router.

Software Redundancy

On the Cisco ASR 1000 Series Routers, Cisco IOS runs as one of many processes within the Cisco IOS XE operating system. This architecture is different than on traditional Cisco IOS, where all processes are run within Cisco IOS. The Cisco ASR 1000 Series Router architecture allows for software redundancy opportunities not available on other Cisco IOS platforms.

Integrated Session Border Controller supports software redundancy by running a Standby peer SBC module within the Standby IOS process. If the Active SBC module fails, then the Active IOS process switches over to the Standby IOS process and the old Standby Integrated SBC module resumes processing as the Active. The Standby IOS process may reside on the same Route Processor as the active IOS process (Cisco ASR 1002 and Cisco ASR 1004 Routers) or it may be on a redundant, standby RP (Cisco ASR 1006 Router).

On the Cisco ASR 1002 and Cisco ASR 1004 Routers, a standby Cisco IOS process is running on the same Route Processor as the active Cisco IOS process. In the event of a Cisco IOS failure, the Router switches to the standby Cisco IOS process. No redundant Route Processor or redundant ESP is available on the Cisco ASR 1002 Series and Cisco ASR 1004 Series Routers.

On the Cisco ASR 1006 Routers, both unified and distributed configurations can operate with a redundant Route Processor and a redundant ESP. In the event of failure of the active Cisco IOS process, the router switches to the standby Cisco IOS process, running on a separate standby Route Processor.

Cisco Unified Border Element (SP Edition) redundancy at the ESP level is provided only if a standby, redundant ESP is used. SBC components running on the active ESP have identical peer components running on the standby ESP. In this case, if the SBC components running on the active ESP fail, then a switchover to the backup ESP occurs.

The following types of software redundancy are supported on Cisco Unified Border Element (SP Edition):

- Route Processor Redundancy (RPR)
- Stateful Switchover (SSO)
- In-Service Software Upgrade (ISSU)

Route Processor Redundancy (RPR)

RPR allows you to run with a standby RP or standby Cisco IOS process without state synchronization. In the event of a fatal error on the active RP (or active Cisco IOS process), the system switches to the standby RP (or standby Cisco IOS process), which then completes its initialization. Because all the state information held by the former "Active" is lost, the new "Active" has to configure itself and relearn all the state information.

Upon an RPR-based RP switchover event, all SBC calls already established (in a steady state) at the time of the switchover are lost. SBC calls in the process of being established at the time of the switchover are dropped as gracefully as possible. No new calls can be established briefly after the initial switchover event.

RPR redundancy can allow for Cisco IOS fast software upgrades when ISSU is unavailable. In RPR mode, no Cisco IOS SBC state information is synchronized to the "Standby." Therefore, all calls are dropped upon an RPR-based switchover.



RPR is supported on the Cisco ASR 1000 Series Routers while RPR+ is not. You can use Stateful Switchover (SSO) instead of RPR+.

SSO Support

Integrated Session Border Controller support for Stateful Switchover (SSO) allows for stateful Cisco IOS process switchovers where critical state information is synchronized between one Route Processor used as the active processor and the other RP used as the standby processor, or between active and standby Cisco IOS processes on the same RP. When Cisco IOS is configured for SSO, the SBC module running on the active IOS process constantly "replicates" its internal state to its standby peer SBC module on the standby IOS process. In this way, the standby SBC module is kept in sync with the active IOS process and has all the state information necessary to retain active calls and resume call processing in the event the active IOS process fails and an SSO occurs.

For information on SSO, see the *Cisco IOS High Availability Configuration Guide* at the following URL: http://www.cisco.com/en/US/docs/ios-xml/ios/ha/configuration/12-2sr/ha-12-2sr-book.html

ISSU Support

Integrated Session Border Controller supports In-Service Software Upgrade (ISSU) with a redundant RP or redundant IOS process. The ISSU process allows software to be updated or otherwise modified on a standby RP or standby IOS process while packet forwarding on the active RP or active IOS process continues. For the Cisco ASR 1000 Series Routers, ISSU compatibility depends on the software package being upgraded and the hardware configuration.

Although ISSU between a distributed-only version and a unified version of Cisco IOS XE software may be supported, the unified features introduced in Cisco IOS XE Release 2.4 are not available to the distributed-only version if you should do a software downgrade. In such cases, we advise you to unconfigure unified Cisco Unified Border Element (SP Edition) before performing a downgrade to a Cisco IOS XE software version that does not support unified Cisco Unified Border Element (SP Edition). The same restriction does not apply to a distributed-only Cisco Unified Border Element (SP Edition) configuration.

See the "High Availability Overview" section in the *Cisco ASR 1000 Aggregation Services Router Software Configuration Guide* for more, updated information on ISSU compatibility.

For information on the ISSU process, see the *Cisco IOS In Service Software Upgrade and Enhanced Fast* Software Upgrade Process document at:

http://www.cisco.com/en/US/docs/ios/12_2sb/feature/guide/sb_issu.html.



Interchassis High Availability

The Interchassis High Availability feature provides geographically dispersed multibox redundancy. The unified session border controller (SBC) and the distributed SBC support the box-to-box high availability.

Interchassis High Availability feature is supported by the Cisco ASR 1001 Series Routers, Cisco ASR 1002 Series Routers, Cisco ASR 1004 Series Routers, Cisco ASR 1006 Series Routers, and Cisco ASR 1013 Series Routers.

Cisco Unified Border Element (SP Edition) was earlier known as Integrated Session Border Controller. It is referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all the Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 3.2S	This feature was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.3S	Added support information pertaining to the Cisco ASR 1006 Series Router, Cisco ASR 1013 Series Router, and interchassis-intrachassis conversion.
Cisco IOS XE Release 3.7S	Added information about upgrading interchassis redundancy.

Feature History for Interchassis High Availability

Contents

This module contains the following sections:

- Prerequisites for Interchassis High Availability, page 15-2
- Restrictions for Interchassis High Availability, page 15-2
- Information About Interchassis High Availability, page 15-3
- Assigning a Redundancy Group to the SBC, page 15-12
- Managing and Monitoring Interchassis High Availability, page 15-14

- Upgrading Interchassis Redundancy, page 15-16
- Configuration Examples for Interchassis High Availability, page 15-17

Prerequisites for Interchassis High Availability

Following are the prerequisites pertaining to the Interchassis High Availability feature:

- The interfaces shared by the SBC must have the same redundant interface identifier (RII).
- The active device and the standby device must have the same peripheral configuration as the SBC features such as the SBC interfaces, virtual routing and forwarding (VRF), routes, sbc redundancy groups, and so on. The SBC-specific configuration will be replicated to the standby. Therefore, only the active Cisco ASR 1000 Series Router requires the full SBC-specific configuration.
- The active device and the standby device must run on the identical version of the Cisco IOS XE software.
- The active device and the standby device must be connected through an L2 connection for the control path.
- The Embedded Service Processor must be the same on both the active and standby devices. RP's must also match and have similar physical port adapter configuration.
- Network Time Protocol (NTP) must be configured or the clock must be set identical on both Cisco ASR 1000 Series Routers to allow timestamps and call timers to match.
- The latency times must be minimal on all control and data links to prevent timeouts.
- Physically redundant links, such as Gigabit Ether Channel must be used for control and data path.

Restrictions for Interchassis High Availability

Following are the restrictions pertaining to the Interchassis High Availability feature:

- Clustering of more than two SBCs for redundancy is not supported.
- The failover time for a box-to-box application is higher for a non-box-to-box application.
- LAN and MESH scenarios are not supported.
- If a dual IOS daemon is configured, the device does not support the interchassis high availability configuration.
- Only the SBC Active-Standby mode is supported.
- The SBC interfaces must be used for signaling and media addresses. Physical interface IP addresses must not be used.
- VRF's must be defined in the same order on both active and standby routers for an accurate synchronization of the SBC data.
- When the configuration is replicated to the standby router, it is not committed to the startup configuration, it is in the running configuration. The user must execute the **write memory** command to commit changes on the standby router that have been synchronized from the active router.
- Coexistence of interchassis high availability and intrachassis high availability is not supported.
- In Cisco ASR 1001 Series Routers, Cisco ASR 1002 Series Routers, and Cisco ASR 1004 Series Routers, the interchassis redundancy is not supported with software redundancy.

- In Cisco ASR 1006 Series Routers and Cisco ASR 1013 Series Routers, interchassis redundancy is not supported with intrachassis redundancy. It is supported with a single RP and ESP in the chassis.
- When CUBE-SP is in inter-chassis redundancy mode, customer need to use **sync** command in the active box to sync the configuration file from active box to standby box so that the latest configuration of CUBE-SP will be synchronized in the running configuration file in the standby box.

Information About Interchassis High Availability

The Interchassis High Availability feature enables the configuration of pairs of routers to act as backup for each other. This feature can be configured to determine the active router based on a number of failover conditions. When a failover occurs, the standby router seamlessly takes over and starts processing call signaling and performing media forwarding.

Groups of redundant interfaces are known as redundancy groups. Figure 15-1 depicts the active-standby device scenario. It shows how the redundancy group is configured for a pair of routers that have a single outgoing interface.

The routers are joined by a configurable control link and data synchronization link. The control link is used to communicate the status of the routers. The data synchronization link is used to transfer stateful information from the SBC, and to synchronize the stateful database for the calls and media flows. Each pair of redundant interfaces are configured with the same unique ID number, also known as the RII.



Figure 15-1 Redundancy Group Configuration

The status of the redundancy group members is determined through the use of Hello messages sent over the control link. If either of the routers do not respond to a Hello message within the configured amount of time, it is considered that a failure has occurred, and a switchover is initiated. To detect a failure in milliseconds, the control links run the failover protocol integrated with the Bidirectional Forwarding Detection (BFD) protocol. You can configure the following parameters for the Hello messages:

- Active timer
- Standby timer
- Hellotime-The interval at which Hello messages are sent.
- Holdtime–The amount of time before the active or the standby router is declared to be down.

The hellotime defaults to 3 seconds to align with the Hot Standby Router Protocol (HSRP), and the holdtime defaults to 10 seconds. You can also configure these timers in milliseconds by using the **timers hellotime msec** command.



B2B HA redundancy holdtime should be at least 3 seconds, and is recommended holdtime is 5 seconds.



If you allocate a large amount of memory, for example, 1 GB, to the log buffer, the CPU utilization and memory utilization of the router increases. This issue is compounded if you set small intervals for the hellotime and the holdtime. If you want to allocate a large amount of memory to the log buffer, we recommend that you accept the default values for the hellotime and holdtime. For the same reason, we also recommend that you do not use the **preempt** command.

To determine which pairs of interfaces are affected by the switchover, you must configure the RII for each pair of redundant interfaces.

Priority can be configured in the startup or running configuration, whereas the run-time priority is the priority of the router at any given time. The run-time priority can be similar to the configured priority if no decrements have been made, or it may be lowered based on the interface faults and decrements. The following priority factors can cause a switchover:

• The router with the highest priority value is the active router. If a fault occurs on either the active router or the standby router, the priority of the router is decremented by a configurable amount known as the decrement value. If the priority of the active router falls below the priority of the standby router, a switchover occurs, and the standby router becomes the active router. This default behavior can be overridden by disabling the preemption attribute for the redundancy group. You can also configure each interface to decrease the priority when the L1 state of the interface goes down. This amount overrides the default amount configured for the redundancy group.



Note

By default, preemption is not enabled. It can be enabled using the **preempt** command. When preemption is configured, the standby router initiates the failover. However, if you configure SBC on the router, we recommend that you do not use the **preempt** command. If the **preempt** command has been configured and if a failover occurs, the B2B state changes might not progress in a manner that permits a guaranteed amount of time for SBC synchronization.

Each failure event that causes a modification of a redundancy group's priority generates a syslog entry that contains a time stamp, information about the redundancy group that was affected, the previous priority, the new priority, and a description of the failure event cause.

• When the priority of a router or interface falls below a configurable threshold level, the active router initiates the failover.

A switchover to the standby router can also occur under the following circumstances:

- Power loss or reload occurs on the active router, including crashes.
- The redundancy group on the active router is reloaded manually using the **redundancy application reload group rg-number self** command.

Two consecutive Hello messages that are missed on any monitored interface forces the interface into testing mode. When this occurs, both the units first verify the link status on the interface, and then execute the following tests:

- Network activity test
- ARP test
- Broadcast ping test

Exclusive Virtual IP and Exclusive Virtual MAC

Virtual IP (VIP) and Virtual MAC are used by the SBC application to control the interfaces that receive traffic. An interface on one device is paired with another interface on another device, and both the interfaces are associated with the same redundancy group. The interface that is associated with an active redundancy group exclusively *owns* the VIP Address and the Virtual MAC. The Address Resolution Protocol (ARP) process on that device sends ARP replies for ARP requests, if any, pertaining to the VIP, and the Ethernet controller for the interface is programmed to receive the packets destined for the Virtual MAC. When a redundancy group failover occurs, the *ownership* of the VIP and Virtual MAC changes. The interface associated with the newly active redundancy group sends a gratuitous ARP, and programs the interface's Ethernet controller to accept the packets destined for the Virtual MAC.

LAN-LAN Topology

The Interchassis High Availability feature supports the LAN-LAN topology. Figure 15-2 shows a LAN-LAN topology. Traffic is often directed to the SBC by configuring static routing in the upstream or downstream routers to an appropriate SBC interface IP address. In addition, the Cisco ASR 1000 Series Routers can participate in dynamic routing with either upstream or downstream routers. The dynamic routing configuration supported on the LAN-facing interfaces can introduce a dependency on routing protocol convergence, thus increasing the failover time.



WAN-LAN Topology

The Interchassis High Availability feature supports the WAN-LAN topology.



However, asymmetric routing is not supported in the Interchassis High Availability feature.

Figure 15-3 shows a WAN-LAN topology in which the LAN is similar to that present in the LAN-LAN topology. For the WAN, VIP is not required. The SBC interface network can be distributed on both the Cisco ASR 1000 Series Routers through dynamic routing. Routing protocols, such as OSPF, ISIS, and BGP, can run over the WAN links.

For a traffic failover caused by a WAN-facing router failure, the immediate WAN link or other WAN connectivity is dependent on the routing protocol convergence. Although subsecond failover cannot be achieved in these failure scenarios, fault detection can be minimized by tuning the routing protocol keep-alive timers and using the BFD feature, if available. Using the IOS Track feature and decreasing the redundancy group's priority values on the Cisco ASR 1000 Series Router to trigger failovers when the WAN links have failed, helps minimize the SBC downtime by failing over to a standby router with full connectivity.



Transport by Redundancy Group and SBC

Redundancy group requires each client to establish a connection between the standby and active devices. The Cisco ASR 1000 Series Router platform implementation for box-to-box uses a connection between the standby and active routers using Stream Control Transmission Protocol (SCTP). This connection is used by the Redundancy Facility client to exchange the events and status used to keep the two boxes in synchronization. The platform also has an MCP client that uses a reliable User Datagram Protocol (UDP) connection for exchanging the platform-specific status and events.

The SBC has its own client, and uses a TCP connection for exchanging status, events, and replication data. These connections can be viewed using the **show redundancy application transport clients** command, and the details of the connections, ports, and IP addresses, can be viewed using the **show redundancy application group** command.

Interchassis-Intrachassis Conversion

From Cisco IOS XE Release 3.3S, Interchassis High Availability feature is also supported on Cisco ASR 1006 Series Routers and Cisco ASR 1013 Series Routers.

Intrachassis high availability occurs when the Cisco ASR 1000 Series Router has two routing processors (RP), with one RP in active mode and the other RP in standby mode. Interchassis high availability occurs when there are two Cisco ASR 1000 Series Router, with one router in active mode and the other in standby mode, and each router has one RP.

The following sections list the steps involved in high availability interchassis-intrachassis conversion:

- Intrachassis to Interchassis Conversion, page 15-8
- Interchassis to Intrachassis Conversion, page 15-10

Intrachassis to Interchassis Conversion

The following steps describe the procedure involved in dual RPs to single RP box-to-box conversion:

- **Step 1** Configure the Cisco ASR 1006 Series Router and Cisco ASR 1013 Series Router with dual RPs and dual forwarding processors (FPs) in the Stateful Switchover (SSO) mode.
- **Step 2** Configure the SBC functionality and generate test calls to ensure proper operation.
- **Step 3** Remove one RP and one FP from a box, using either OIR or CLI shutdown methods.
- **Step 4** Configure application redundancy:

```
Router(config)# interface GigabitEthernet0/1/1
Router(config-if) # redundancy rii 600
Router(config-if)# redundancy group 1 ip 10.2.3.4 exclusive decrement 200
Router(config-if) # exit
Router(config) # redundancy
Router(config-red) # application redundancy
Router(config-red-app)# group 1
Router(config-red-app-grp)# name SBC
Router(config-red-app-grp)# data GigabitEthernet 0/0/1
Router(config-red-app-grp)# control GigabitEthernet 0/0/2 protocol 1
Router(config-red-app-grp)# timers delay 100 reload 400
Router(config-red-app-grp)# track 1 decrement 1
Router(config-red-app-grp)# track 2 decrement 1
Router(config-red-app-grp)# exit
Router(config-red-app)# protocol 1
Router(config-red-app-prtcl)# name BFD
Router(config-red-app-prtcl)# timers hellotime 4 holdtime 6
Router(config-red-app-prtcl)# authentication md5 key-string 0 n1 100
```

Step 5 Add the SBC application redundancy configuration after the RG is shutdown:

```
Router(config)# redundancy
Router(config-red)# application redundancy
Router(config-red-app)# group 1
Router(config-red-app-grp)# name SBC
Router(config-red-app-grp)# shutdown
Router(config-red-app-grp)# exit
Router(config-red-app)# exit
Router(config-red-app)# exit
Router(config-red)# exit
Router(config-red)# exit
Router(config)# sbc redundancy-group 1 tcp
Router(config)# redundancy
Router(config-red)# application redundancy
Router(config-red-app)# group 1
```

Router(config-red-app-grp)# no shutdown

- Step 6 Save the SBC configuration and use the no sbc command to remove the SBC configuration: Router(config) # no sbc ASR1
- Step 7 Check whether the Cisco ASR 1000 Series Router is in the ACTIVE mode or UNKNOWN mode because another Cisco ASR 1000 Series Router is not yet configured:

Router# show redundancy application transport group

- Step 8 Configure the SBC again using the saved configuration. Router(config)# sbc ASR1
- **Step 9** Place a test call to ensure that the SBC is functioning well.
- **Step 10** Bring the second Cisco ASR 1000 Series Router online with a single RP and single FP.
- **Step 11** Configure application redundancy on the second Cisco ASR 1000 Series Router:

```
Router(config)# interface GigabitEthernet0/1/1
Router (config-if) # redundancy group 1 ip 10.1.1.1 exclusive decrement 50
Router(config-if)# redundancy rii 10
Router(config-if) # exit
Router(config) # redundancy
Router(config-red) # application redundancy
Router(config-red-app)# group 1
Router(config-red-app-grp)# name SBC
Router(config-red-app-grp)# data GigabitEthernet 1/0/1
Router(config-red-app-grp)# control GigabitEthernet 0/0/1 protocol 1
Router(config-red-app-grp)# timers delay 100 reload 400
Router(config-red-app-grp)# track 1 decrement 1
Router(config-red-app-grp)# track 2 decrement 1
Router(config-red-app-grp)# exit
Router(config-red-app) # protocol 1
Router(config-red-app-prtcl)# name BFD
Router(config-red-app-prtcl)# timers hellotime 4 holdtime 6
Router(config-red-app-prtcl)# authentication md5 key-string 0 n1 100
```

Step 12 Add the SBC application redundancy configuration to the second Cisco ASR 1000 Series Router after the RG is shut down:

```
Router(config)# redundancy
Router(config-red)# application redundancy
Router(config-red-app)# group 1
Router(config-red-app-grp)# name SBC
Router(config-red-app-grp)# shutdown
Router(config-red-app)# exit
Router(config-red)# exit
Router(config)# sbc redundancy-group 1 tcp
Router(config)# redundancy
Router(config-red)# application redundancy
Router(config-red-app)# group 1
Router(config-red-app)# group 1
Router(config-red-app)# group 1
```

Step 13 Configure the second Cisco ASR 1000 Series Router such that it is in the STANDBY HOT mode:

Router# show redundancy application transport group

Step 14 Check whether the first Cisco ASR 1000 Series Router is still in the ACTIVE mode:
 Router# show redundancy application transport group
 Step 15 Check whether the SBC configuration is synchronized to the Cisco ASR 1000 Series Router that is in the STANDBY mode:
 Router# show run
 Step 16 Place a test call to check whether the SBC is still functioning.

Interchassis to Intrachassis Conversion

The following steps describe the procedure involved in single RP box-to-box to dual RPs conversion:

- **Step 1** Configure two Cisco ASR 1000 Series Routers with single RP's and single FP's in the box-to-box mode.
- **Step 2** Generate test calls with multiple failovers to ensure proper box-to-box operation.
- **Step 3** Shut the RG on both the Cisco ASR 1000 Series Routers:

Router(config)# redundancy Router(config-red)# application redundancy Router(config-red-app)# group 1 Router(config-red-app-grp)# shutdown

- **Step 4** Remove the SBC redundancy configuration from both the Cisco ASR 1000 Series Routers: Router (config) # no sbc redundancy-group 1 tcp
- Step 5Remove the RG configuration from both the Cisco ASR 1000 Series Routers:Router(config-red)# no application redundancy
- Step 6 Save the SBC configuration, and use the **no sbc** command to remove the SBC configuration: Router(config)# **no sbc ASR1**
- **Step 7** Add one RP and one FP to the Cisco ASR 1000 Series Router that is in the ACTIVE mode.
- **Step 8** Configure the SBC again using the saved configuration:

Router(config) # **sbc ASR1**

Step 9 Check whether the SBC application of the primary Cisco ASR 1000 Series Router has been activated and is functioning correctly:

Router# show redundancy application transport group

- **Step 10** Generate test calls to verify whether the SBC is functioning, and leave this call active in order to be able to perform the subsequent steps.
- **Step 11** Configure the SSO redundancy:

Router(config)# redundancy
Router(config-red)# mode sso

- **Step 12** Check whether the configuration is synchronized and there are no interruptions in the SBC traffic: Router# **show run**
- **Step 13** Place a test call to ensure that the SBC is still functioning.

Configuring Interchassis High Availability

To configure Interchassis High Availability, see the following sections in the "Configuring Firewall Stateful Inter-Chassis Redundancy" chapter of *Security Configuration Guide: Zone-Based Policy Firewall Cisco IOS XE Release 3S*:

http://www.cisco.com/en/US/partner/docs/ios-xml/ios/sec_data_zbf/configuration/xe-3s/sec-data-zbf-x e-book.html

This chapter provides information about the following topics:

- Configuring the Redundancy Application Group
- Configuring the Redundancy Group Protocol
- Configuring Virtual IP Address and Redundant Interface Identifier
- Configuring Control and Data Interface

Configuring Static Routing with Interchassis High Availability

When a static route is used in an upstream and downstream router or Layer 3 switch, VIP must be configured on the LAN-facing interface on the Cisco ASR 1000 Series Router. The static route that has an SBC interface IP address as the destination IP address sets the VIP address as the next hop address. Although this scenario offers the best convergence time during a failover, it faces an unicast flooding problem in the LAN between the router or the Layer 3 switch and the Cisco ASR 1000 Series Router.

The default ARP table aging time is 4 hours, while the MAC table aging time is only a couple of minutes. A MAC aging timer, which is greater or equal to the ARP timeout, is required to prevent unicast flooding for both upstream LAN and downstream LAN. After the ARP table is timed out, it sends an ARP request towards the VIP. The active Cisco ASR 1000 Series Router replies to the ARP request with a VMAC. The MAC table is refreshed and the unicast flooding problem is resolved.

To increase the MAC aging timer or decrease the ARP aging timer for the VLAN with the unicast flooding problem, use one of the following commands on the router or the Layer 3 switch:

- The arp timeout command on a VLAN interface
- The mac-address-table aging-time vlan command

Configuring Dynamic Routing with Interchassis High Availability

The SBC interface must be included as part of the Open Shortest Path First (OSPF) area so that the SBC is advertised when the box becomes active. The following example shows an OSPF configuration, illustrating the SBC box-to-box application with routing:

```
router ospf 200
router-id 4.4.4.10
priority 11
nsf
network 4.4.0.0 0.0.255.255 area 0
interface SBC1
ip address 10.2.0.1 255.255.255.0 secondary
ip address 10.2.0.10 255.255.255.0 secondary
ip address 10.2.0.100 255.255.255.0
ip ospf 200 area 0
```



To prevent duplicate IP addresses, the SBC interface is held in a down/down state on the standby router.

Assigning a Redundancy Group to the SBC

This task shows how to assign a redundancy group to the SBC:



Configuration on the SBC interface is similar on both active and standby routers. However, redundancy group traffic interfaces have different IP addresses and a shared redundancy IP address. While performing this procedure on the Cisco ASR 1001 Router, Cisco ASR 1002 Router, and Cisco ASR 1004 Router, set the redundancy mode to NONE.

SUMMARY STEPS

- 1. enable
- 2. configure terminal
- 3. redundancy
- 4. application redundancy
- 5. group id
- 6. shutdown
- 7. exit
- 8. exit
- 9. exit
- 10. sbc redundancy-group group-number tcp
- 11. redundancy
- 12. application redundancy
- 13. group id
- 14. no shutdown
- 15. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode. Enter your password if prompted.
	Example: Router> enable	
Step 2	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	

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	Command or Action	Purpose
Step 3	redundancy	Enters the redundancy configuration mode.
	Example:	
	Router(config)# redundancy	
Step 4	application redundancy	Enters the redundancy application configuration mode.
	Example: Router(config-red)# application redundancy	
Step 5	group id	Enters the redundancy application group configuration mode.
	Example: Router(config-red-app)# group 1	• <i>id</i> —Specifies the redundancy group ID that ranges from 1 to 2.
Step 6	shutdown	To assign a redundancy group to the SBC, the redundancy group must be shut down.
	Example: Router(config-red-app-grp)# shutdown	
Step 7	exit	Exits from the redundancy application group configuration mode and enters the redundancy application configuration mode
	Example: Router(config-red-app-grp)# exit	inde.
Step 8	exit	Exits from the redundancy application configuration mode and enters the redundancy configuration mode.
	Example: Router(config-red-app)# exit	
Step 9	exit	Exits from the redundancy configuration mode and enters the global configuration mode.
	Example: Router(config-red)# exit	
Step 10	sbc redundancy-group group-number tcp	Assigns the redundancy group to the SBC in order to track the following:
	Example: Router(config)# sbc redundancy-group 1 tcp	• <i>group-number</i> —Specifies the redundancy group number.
		• tcp —Specifies the Transmission Control Protocol TCP), and the redundancy group protocol.
Step 11	redundancy	Enters the redundancy configuration mode.
	Example: Router(config)# redundancy	
Step 12	application redundancy	Enters the redundancy application configuration mode.
	Example: Router(config-red)# application redundancy	

	Command or Action	Purpose
Step 13	group id	Enters the redundancy application group configuration mode:
	Example: Router(config-red-app)# group 1	• <i>id</i> —Specifies the redundancy group ID that ranges from 1 to 2.
Step 14	no shutdown	The redundancy group gets activated.
	Example: Router(config-red-app-grp)# no shutdown	
Step 15	end	Exits the redundancy application group configuration mode and enters the Privileged EXEC mode.
	Example: Router(config)# end	

Managing and Monitoring Interchassis High Availability

You can manage and monitor the Interchassis High Availability feature as explained in the following sections:

- Managing and Monitoring the Redundancy Group infrastructure, page 15-14
- Managing and Monitoring an SBC Redundancy Group, page 15-15

Managing and Monitoring the Redundancy Group infrastructure

To manage and monitor the redundancy group infrastructure, use the following commands:

- redundancy application reload group *group-number* {peer | self}—Forces an active redundancy group to reload, and a standby redundancy group to become the active redundancy group, without affecting the status of the active redundancy group.
- **show redundancy application** {*group-id* | **all**}—Shows the summary information pertaining to the specified group or all the groups.
- **show redundancy application faults** {*group-id* | **all**}—Shows information about the faults pertaining to the specified group or all the groups.
- **show redundancy application if-mgr** {*group-id* | **all**}—Shows information about the if-mgr pertaining to the specified group or all the groups.
- **show redundancy application interface** *interface*—Shows the interface information associated with the redundancy groups.
- **show redundancy application protocol** *group-id*—Shows the protocol information pertaining to the specified group or all the groups.
- **show redundancy application transport** {*group-id* | **clients**}—Shows transport information pertaining to the specified group or all the groups.

To enable debug logging of the specified type of information associated with redundancy groups, use the following commands:

- debug redundancy application vp {event | error}
- debug redundancy application transport {db | trace | event | error | timer}

- debug redundancy application media {packet | event | error | timer | nbr | all}
- debug redundancy application protocol {event | error | media | peer | detail | all}
- debug redundancy application faults {event | error | fault | func | db | all }
- debug platform software rg {tdl | terse | detail | error}

Managing and Monitoring an SBC Redundancy Group

To manage and monitor an SBC redundancy group, use the following commands:

- **show sbc** *name* **rg transport**—Shows the transport information pertaining to an SBC redundancy group.
- show sbc name rg statistics—Shows the transport statistics pertaining to an SBC redundancy group.
- clear sbc name rg—Clears the SBC redundancy group box-to-box statistics.
- **monitor event-trace sbc ha**—Configures event tracing pertaining to the SBC in order to include significant redundancy group events for generating the history for bootup and transition logs to assist in debugging.

The following example shows a sample output of the show sbc name rg transport command:

```
Router# show sbc MySBC rg transport
SBC HA RG connection parameters for domain 2
```

Application Type	1
Handler	53
My IP address	1.0.0.7
My L4 Port	1060
L3 Protocol	1
L4 Protocol	1
Peer IP address	1.0.0.6
Peer L4 Port	1060
My MTU	1464
My L4 Offset	28

The following example shows a sample output of the **show sbc** name **rg** statistics command:

```
Router# show sbc MySBC rg statistics
SBC HA B2B statistics
_____
Number of messages successfully queued
                                             = 407
Number of messages successfully sent
                                             = 407
Number of IPS messages sent
                                             = 370
Number of messages queue failures
                                             = 0
Number of attempted-send message failures
                                            = 0
Number of message header malloc failures
                                             = 0
Number of no packet available failures
                                             = 0
Number of high watermark of queued messages
                                             = 16
Number of high watermark of recv messages
                                             = 15
                                             = 412
Number of messages received
Number of received IPS messages
                                             = 356
Number of received messages discarded
                                             = 0
Number of received messages dropped(no group) = 0
Number of received large IPS messages
                                             = 37
Number of large message send failures
                                             = 0
Number of large message send total
                                             = 0
Number of large message recv failures
                                             = 0
Number of large message not sent, unsupp by peer = 0
```

The following example shows a sample output of the **show monitor event-trace sbc ha all** command. In this example, all the messages from the SBC high availability events are displayed:

Router# show monitor event-trace sbc ha all

```
*Jan 16 10:21:49.718: RF: Is Active, from boot = 0x1
*Jan 16 10:21:49.720: IPC: Initialised as master
*Jan 16 10:21:49.720: RF: Active reached, from boot = 0x1
*Jan 16 10:21:59.448: ILT: Registered on 48, result = 0x1
*Jan 16 10:21:59.448: RF: Start SM on 48
*Jan 16 10:49:02.523: IPC: Session to peer opened
*Jan 16 10:49:02.605: ISSU: Negotiation starting
*Jan 16 10:49:02.605: RF: Delaying progression at 300
*Jan 16 10:49:02.617: ISSU: Negotiation done
*Jan 16 10:49:02.617: RF: Negotiation result = 0x1
*Jan 16 10:49:02.617: RF: Negotiation result = 0x1
*Jan 16 10:49:02.617: RF: Peer state change, peer state = 0x1
*Jan 16 10:49:02.617: RF: Resuming progression at event 300
*Jan 16 10:50:00.853: ISSU: Transformed transmit message
*Jan 16 10:50:00.853: IPC: Queuing message type SBC_HA_MPF_CAPS_MSG_TYPE
*Jan 16 10:50:00.854: IPC: Queued message type SBC_HA_MPF_CAPS_MSG_TYPE
```

Upgrading Interchassis Redundancy

To upgrade interchassis redundancy, perform the following steps:

Two Cisco ASR1000 Series Aggregation Services Routers are required to perform this procedure. While		
the primary router is the active one, the secondary router is the standby one.		
In the primary router, use the show redundancy application group <i>RG Group ID</i> command to display which router is the active one.		
In the secondary router, use the show redundancy application group <i>RG Group ID</i> command to display which router is the standby one.		
Download the latest version of the Cisco ASR 1000 Series Aggregation Services Routers image to both the primary router and the secondary router.		
On the secondary router, change the boot variable to the new image by using the boot system bootflash <i>new image</i> command.		
On the primary router, synchronize the SBC by using the sbc sbc name command and the sync command		
Wait for five minutes to make sure that the SBC configuration is fully synchronized to the standby router		
On the secondary router, save the running configuration by using the write memory command.		
On the primary router, shut down the redundancy group.		
The secondary router immediately becomes the active one, and all the active calls are preserved. Note that the router is still in service when switching over to the active router.		
On the primary router, change the boot variable to the new software image, and save the running configuration.		
Reload the primary router for upgrading, and wait for this router to come up with the upgraded version It might take around 10 to 12 minutes from the time the router is reloaded.		
On the secondary router, shut down the redundancy and execute the no shutdown command for the redundancy group on the primary router as soon as possible.		

- **Step 11** The router will be down for around 100 seconds, and the primary router becomes active and in service with the upgraded software.
- **Step 12** Save the running configuration in the primary router.
- **Step 13** Reload the secondary router for upgrade. When you are asked whether you want to save the configuration before proceeding with the reload, enter No so that the secondary router will come up in the standby state after the upgrade.

The upgrade is completed.

Configuration Examples for Interchassis High Availability

To view the list of configuration examples pertaining to Interchassis High Availability, see the following sections in the "Configuring Firewall Stateful Inter-Chassis Redundancy" chapter of *Security Configuration Guide: Zone-Based Policy Firewall Cisco IOS XE Release 3S* at:

http://www.cisco.com/en/US/partner/docs/ios-xml/ios/sec_data_zbf/configuration/xe-3s/sec-data-zbf-x e-book.html

- Example: Configuring the Redundancy Application Group
- Example: Configuring the Redundancy Group Protocol
- Example: Configuring Virtual IP Address and Redundant Interface Identifier
- Example: Configuring Control and Data Interface





Fax Support

The Cisco Unified Border Element (SP Edition) media components enable fax over IP calls. Cisco Unified Border Element (SP Edition) supports the following types of fax over IP calls, using Session Initiation Protocol (SIP) or H.323:

- G.711 passthrough
- T.38 fax passthrough over the following protocols:
 - RTP: Real-time Transport Protocol
 - UDP-TL: A lightweight transport protocol for fax media that runs over User Datagram Protocol

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for Fax Support

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR, along with support for the unified model.
Cisco IOS XE Release	The following were added in this release:
2.5	• Support for H.323
	• G.711 passthrough support for SIP and H.323 interworking calls
	• T.38 fax support for H.323 to H.323 and SIP to H.323 interworking calls.
	• Fax Upspeed support.

Contents

This module contains the following sections:

- Restrictions for Fax Support, page 16-2
- Fax Support, page 16-2
- Fax Upspeed Support, page 16-3

Restrictions for Fax Support

The following are restrictions for fax support in Cisco Unified Border Element (SP Edition):

- G.711 and T.38 interworking is not supported.
- T.38 fax passthrough does not support H.323 to SIP calls, or SIP to H.323 to SIP calls.
- Cisco proprietary fax is not supported, although it may work in the passthrough mode, because the Cisco Unified Border Element (SP Edition) does not police the RTP payload types, only the bandwidth. Cisco proprietary fax uses RTP.

Fax Support

Cisco Unified Border Element (SP Edition) supports two types of fax over IP:

- G.711 passthrough
- T.38 passthrough

G.711 Passthrough

G.711 passthrough is an International Telecommunication Union (ITU) standard codec operating at a 64 Kbps rate. It is a simple fax method and supports sending fax in the RTP stream of a typical G.711 call. G.711 is used by most VoIP providers because it provides high voice quality. It produces voice sounds similar to a regular or ISDN phone because G.711 does not use compression and uses the same codec used by the public switched telephone network (PSTN) and Integrated Services Digital Network (ISDN). Without the demand on processing power required by compression, G.711 has the lowest latency or lag.

Passthrough is a method of passing a FAX PCM stream across a VoIP network. It involves selecting a low-bandwidth codec (G.711), disabling silence suppression, and enabling echo cancellation. FAX passthrough is signalled by protocol stacks H.323 and SIP.

G.711 passthrough is supported for all cases of SIP and H.323 interworking calls in Cisco IOS XE Release 2.5 and later.

Note

The Cisco Unified Border Element (SP Edition)'s billing records for the call do not show anything explicitly because of the fax nature of the call. They merely report the standard set of metrics for the call, as they would do for a voice call.

T.38 Passthrough

T.38 Passthrough is an ITU standard for sending FAX across IP networks in a real-time mode. In Cisco Unified Border Element (SP Edition), T.38 fax calls are sent in-band using a fax-specific codec (rather than a general-purpose audio codec). T.38 fax uses a separately negotiated stream, which can either be negotiated at the start of the call (bandwidth will be reserved for it at that point), or renegotiated during the call (which may fail).

Passthrough is a method of passing a FAX PCM stream across a VoIP network. It involves selecting a high-bandwidth codec, disabling silence suppression, and enabling echo cancellation. FAX passthrough is signalled by protocol stacks H.323 and SIP.
T.38 fax passthrough is supported for SIP to SIP calls in Cisco IOS XE Release 2.4 and later. Support for SIP to H.323 and H.323 to H.323 calls is added in Cisco IOS XE Release 2.5 and later. In the case of SIP to H.323 calls, only the SIP side of the call will initiate the T.38 passthrough.

T.38 fax passthrough does not support H.323 to SIP calls or SIP to H.323 to SIP calls.

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The bandwidth reserved for a T.38 call is considered sufficient for carrying a T.38 rate of 14, 400 bits per second and does not reflect the signaled rate in T.38.



If an unnumbered datagram protocol transport layer (UDPTL) error correction is used for T.38, then the bandwidth reservation also includes capacity for up to three redundant parity packets in the T.38 stream.

Fax Upspeed Support

Cisco Unified Border Element (SP Edition) supports fax upspeed. Fax upspeed is the ability of the SBC to change a codec in midcall by re-negotiation. The fax upspeed function is only supported when one of the endpoints engaged in the call initiates it; the SBC does not initiate the upspeed action. The SBC is capable of handling midcall codec re-negotiations to and from either H.323 or SIP interfaces.

When an endpoint has determined that the call is a fax or data call and calculates that the codec negotiated is too highly compressed to reliably pass tones, it may initiate a re-Invite to perform a codec re-negotiation that offers the G.711 codec, a higher bandwidth codec. Thus the process of re-negotiating to a higher bandwidth codec is called "fax upspeed." The G.711 codec is also known as PCMA/PCMU.

When the endpoint determines that the fax or data call has ended, then the endpoint can send another Invite re-negotiation to switch back to a lower bandwidth codec.

The same mechanism applies to H.323 call legs, where there is a terminal capability exchange (TCS) and media channels are closed and new ones reopened.

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Codec Handling

A compressor-decompressor (codec) is a device or program that performs a transformation on a data stream or signal. Cisco Unified Border Element (SP Edition) is hard-coded with a set of recognized codecs (see Table 17-1 to Table 17-5), including all commonly used voice and video codecs. The default behavior is to allow all recognized codecs on all calls. Any other codec present in call signaling is removed by Cisco Unified Border Element (SP Edition).

This enhancement allows you to restrict which codec(s) a particular call can use and to configure a minimum permissible packetization period for each permitted codec.

Cisco Unified Border Element (SP Edition) supports passthrough codecs that are passed through the SBC without transcoding. See Table 17-4 on page 17-7 for a list of passthrough codecs by type.

H.323 only supports PCMA, PCMU, G.722, G.723, G.728, G.729, GSM, telephone-event, H.261, H.263, H.264, and T.38 codecs. Therefore in a SIP to H.323 call, if SIP codecs are sent to H.323 that H.323 cannot support, then these codecs are not passed through and the call may fail accordingly.

Cisco Unified Border Element (SP Edition) supports codec transcoding for SIP to SIP calls using an external DSP resource or transcoding resource. Transcoding is the process of translating a media stream encoded using one codec into a media stream encoded using another codec, for example, translating a media stream encoded as PCMU into one encoded as G.726-32. For more information on transcoding, see the Implementing Transcoding chapter.

Cisco Unified Border Element (SP Edition) enables H.323 TCS Codecs support in Cisco IOS XE Release 2.5.1. This support provides the ability to announce media capabilities on behalf of a SIP endpoint to an H.323 endpoint by adding extra offered codecs in the H.245 Terminal Capability Set (TCS) message. See the ?\$paranum>H.323 TCS Codecs Support? section on page 17-16.



For Cisco IOS XE Release 2.4 and later, this feature is supported in the unified model.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

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Feature History for Codec Handling

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 2.4.2	Support for packetization time, p-time, attribute in the SDP offer or answer configured in a codec list was added.
Cisco IOS XE Release 2.5	Support for H.323 calls, and H.323 video codecs H.261, H.263, and H.264 were added. Support for passthrough codecs without transcoding was added.
Cisco IOS XE Release 2.5.1	H.323 TCS Codecs support was added.
Cisco IOS XE Release 2.6	Support for Dynamic CODEC configuration, multiple audio codec, and multiple video codec were added. H.323 support for clear channel data and modem calls was added.
Cisco IOS XE Release 3.2S	Codec preference and reordering support was added.

Contents

This module contains the following sections:

- Prerequisites for Codec Handling, page 17-3
- Restrictions for Codec Handling, page 17-3
- Restriction for H.323 TCS Codecs Support, page 17-3
- Restrictions for Dynamic Codec Configuration, page 17-4
- Restrictions for Codec Reordering, page 17-4
- Codec Handling and Restriction, page 17-4
- Dynamic Codec Configuration, page 17-8
- Configuring Codec Restriction, page 17-9
- Packetization Time, page 17-15
- H.323 TCS Codecs Support, page 17-16
- Codecs Preference and Reordering Support, page 17-17
- Configuration Examples—Configuring Codec Restriction, page 17-22
- Configuration Examples—H.323 TCS Codecs Support, page 17-24
- Configuration Example—Defining a Codec using Dynamic Codec Configuration, page 17-25

Prerequisites for Codec Handling

The following prerequisite is required before you can restrict codecs:

- Before implementing Codec Handling, Cisco Unified Border Element (SP Edition) must already be configured.
- All signaling border element (SBE) and data border element (DBE) configurations required to make simple calls must already be configured.

Restrictions for Codec Handling

Review the following restrictions for codecs:

- For H.323 calls, SIP to H.323 and H.323 to SIP calls, both the callee and caller must use the same codec because any calls requiring transcoding will fail the call setup.
- The media packet forwarder on the DBE polices the bandwidth consumed by each media stream, but it cannot police the type of codecs or the packetization periods.
- Unrecognized codecs cannot be configured as members of the codec whitelist.
- Active calls are not released if there is a change in the codec whitelist during the call.
- If a codec whitelist is configured, Cisco Unified Border Element (SP Edition) removes any unlisted codecs from the call setup flow and media gate allocation.
- Multiple codec whitelists can be configured on a Call Admission Control (CAC) policy basis. For example, the list of codecs allowed for calls from SipAdj1 can be different than the list of codecs allowed for calls from SipAdj2.
- If a codec whitelist has not been configured, all recognized codecs (see Table 17-1 to Table 17-5) are allowed for all calls.
- You must use the textual value of the codec description that appears on the Session Description Protocol (SDP) to configure the codec whitelist, for example "PCMU" or "telephone-event".
- Disallowing all codecs is not supported. However, you could set a bandwidth limit of 0 to achieve the same result.
- Codec lists are not applied to media-bypass calls (those in which Cisco Unified Border Element (SP Edition) does not reserve media resources).
- The format of the codec name is the same as the string used to represent it in SDP, for example PCMU or VDVI. All recognized codec names are listed in Table 17-1 Table 17-5.
- A single codec can only be added to each list once, with a single packetization period.
- For each codec on a list, CAC restricts the signaled packetization period for any stream using that codec to be greater than or equal to the packetization period configured along with the codec in the list. If a stream uses more than one codec in the list, then the greater of all the packetization periods configured for each codec in the list is applied to the stream.

Restriction for H.323 TCS Codecs Support

H.323 TCS Codecs support only applies to H.323-to-SIP or SIP-to-H.323 interworking calls with the caller or callee using SIP.

Restrictions for Dynamic Codec Configuration

The following restrictions apply for Dynamic codec configuration feature:

- Codecs provided with the SBC (system codec) may be modified but not deleted.
- Codec names are case insensitive.
- A maximum of 100 user defined codecs is supported.
- Custom codec payload ID should always be 96.
- When defining new codec, codec type must be specified before all other fields.
- A custom codec is not defined until a codec type is specified.
- The ID of a system codec should not be changed.
- The ID should be unique for a Codec.
- A custom codec should not be deleted if used in a codec list.
- Sample size is only valid for sampling codecs.
- Bandwidth cannot be specified for redundancy or sampling codecs.
- Channels may only be specified for sampling based codecs.

Restrictions for Codec Reordering

- SBC supports codec reordering only for SIP-SDP codecs. Reordering of codecs in H.323 is not supported.
- If media bypass is configured, codec reordering cannot be applied.

Codec Handling and Restriction



The bandwidths listed in the tables below are the bandwidths without the transport layer overheads. Therefore, the actual bandwidths reserved by Cisco Unified Border Element (SP Edition) are higher than the listed values.

Table 17-1 lists sample-based audio codecs.

Payload Type	Codec Name	Clock Rate (Hz)	Sample Size (bits)	Channels
0	PCMU (also known as G.711)	8000	8	1
5	DVI4	8000	4	1
6	DVI4	16000	4	1
8	РСМА	8000	8	1
10	L16	44100	16	2
11	L16	44100	16	1

Table 17-1 Sample-Based Audio Codecs with Packetization Time 10 ms

Payload Type	Codec Name	Clock Rate (Hz)	Sample Size (bits)	Channels
15	G728	8000	2	1
16	DVI4	11025	4	1
17	DVI4	22050	4	1
	G726-40	8000	5	1
2	G726-32	8000	4	1
_	G726-24	8000	3	1
	G726-16	8000	2	1
	L8	8000	8	1
	DAT12	8000	12	2
	L20	44100	10	2
_	L24	44100	24	2

Table 17-1	Sample-Based	Audio Codecs	with Packetization	Time 10 ms	(continued)
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Table 17-2 lists non-sample-based audio codecs.

Table 17-2	Non-Sample-Based Audio Codecs

Payload Type	Codec Name	Packetization Time (ms)	Allocated Bandwidth (bps)
3	GSM	10	13200
4	G723	30	6400
7	LPC	10	5600
9	G722	10	64000
12	QCELP	—	13300
13	CN	10	400
14	MPA	N/A	131072
18	G729	10	8000
18	G.729A	10	8000
_	G729B	20	8000
	G729AB	10	8000
	G729D	10	6400
	G729E	10	11800
	GSM-EFR	10	12400
	iSAC	30	32000
	VDVI	10	25000
	AMR	10	12500
	AMR-WB	10	24420
_	dsr-es201108	10	4800
_	EVRC	10	8550

Payload Type	Codec Name	Packetization Time (ms)	Allocated Bandwidth (bps)
_	EVRC0	10	8550
_	mpa-robust	10	327680
_	G7221	10	32000
_	MP4A-LATM	10	131072
_	SMV	10	8550
_	SMV0	10	8550

Table 17-2 Non-Sample-Based Audio Codecs (continued)

Table 17-3 lists video codecs.

Table 17-3 Video Codecs

Payload Type	Codec Name	Packetization Time (ms)	Allocated Bandwidth (bps)
25	CelB	—	524228
26	JPEG	—	524228
28	nv	—	524228
31	H261	—	524228
32	MPV	—	524228
33	MP2T	—	524228
34	H263	—	524228
NA	H264		21000000
	BMPEG	—	524228
	BT656	—	17000000
	DV	—	150000000
	H263-1998	—	524228
	H263-2000	—	524228
	MP1S	—	1600000
	MP2P	—	524228
	MP4V-ES	—	524228
	raw	N/A	150000000
_	SMPTE292M	N/A	150000000

Table 17-4 lists the supported passthrough codecs without transcoding.

Codec Name	Codec Type	Packetization Time (ms)	Allocated Bandwidth (bps)
PCMA (also known as G.711)	sample-based audio	10	64000
PCMU (also known as G.711)	sample-based audio	10	64000
G726-16	audio	10	NA
G726-24	audio	10	NA
G726-32	audio	10	NA
G728	audio	10	NA
G729 A	non-sample-based audio	10	8000
G729 B	non-sample-based audio	10	8000
G723-53	non-sample-based audio	30	6400
G723-63	non-sample-based audio	30	6400
GSM/GSM-FR	non-sample-based audio	10	13200
GSM-EFR	non-sample-based audio	20	12400
GSM-HR	non-sample-based audio	20	5600
AMR	non-sample-based audio	10	12500
EVRC	non-sample-based audio	10	8550
G722	non-sample-based audio	10	64000
iLBC	non-sample-based audio	20	15200
H.261	video	NA	524228
H.263	video	NA	524228

Table 17-4	Passthrough Codecs without Transcoding
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Table 17-5 lists other audio codecs.

Table 17-5Other Codecs

Codec Name	Packetization Time (ms)	Allocated Bandwidth (bps)
telephone-event	20	1600
tone	20	1600
RED	20	1
parityfec	20	1
T140	100	80
pointer	20	1600
H224	20	6560
T38	N/A	15500
X-NSE	20	1600

Dynamic Codec Configuration

The Dynamic Codec configuration feature allows you to:

- **Define new codecs**—You can create variants of codecs that are included in the SBC. For example, G.729 A is a codec that is included in SBC. You may define a variant called G.729.1 using Dynamic Codec feature.
- **Modify existing codecs**—You can modify certain attributes of the codecs included in the SBC. For example, you can change the bandwidth characteristics of H.264 video codecs, included in the SBC.
- Display codecs supported on SBC—You can view all codecs supported on an SBC.



Aliasing and Codec conversion are not supported by Dynamic Codec Configuration feature in this release.

Audio and Video codecs that are not included in SBC can be defined using the Dynamic Codec Configuration feature.

Multiple Audio Codec Support

SBC transparently passes the following codecs:

- G.711 with silence suppression and RFC 2833 (and RFC 4733)
- G.723
- G.726
- G.726 with silence suppression (Silence suppression is used to save bandwidth by not sending the data when there is no voice during a call.)
- G.729
- G.729a
- G.729a/b
- AMR
- AAC-LD
- AMR-WB
- AMR-WB+
- G.718

Audio codecs not included in SBC can be defined using the Dynamic Codec Configuration feature.

Multiple Video Codec Support

SBC transparently passes media encoded using:

- H.263
- H.264
- H.264 AVC

- H.264 SVC
- Microsoft RT Video
- Helix Real Video

Video codecs not included in SBC can be defined using the Dynamic Codec Configuration feature.

H.323 Support for Clear Channel Data/Modem Calls

The clearmode codec is supported in existing VoIP equipments, and is used in the H.323, SIP, and MGCP signaling protocols. The Cisco SBC supports the clearmode codec in all places where it supports common audio codecs, such as G.711.

Configuring Codec Restriction

You first configure the codecs and then apply them as explained in the following sections:

- Configuring Codecs, page 17-9
- Configuring a CAC Policy to Use a Codec List, page 17-11

Configuring Codecs

To restrict which codec(s) a particular call can use and to configure a minimum permissible packetization period for each permitted codec, you must configure CAC with a list of codecs, provide a description for the list, and then add any codec(s) to the list.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. codec list name
- 5. description text
- 6. codec codec-name [packetization-period packetization-period]
- 7. end
- 8. show sbc service-name sbe codec-list list-name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Configures the submode of the SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	codec list name	Creates a codec list.
	Example: Router(config-sbc-sbe)# sbc mysbc sbe codec list my_codecs	
Step 5	description text	Adds a description for the specified codec list using a readable text string format.
		The no form of this command removes the description.
	Example: Router(config-sbc-sbe-codec-list)# description Legitimate codes	This description is displayed when the show command is used for this codec list. It is also included for each codec list when a summary of all codec lists is displayed.
Step 6	codec codec-name [packetization-period packetization-period]	Adds a codec to a codec list, and sets a minimum packetization period (optional) for the codec.
		The no form of this command (without the packetization period) removes the named codec from the codec list.
	Example: Router(config-sbc-sbe-codec-list)# codec PCMU packetization-period 20	Note If the no form of this command includes the packetization period, only the packetization period for the codec is removed.
Step 7	end	Exits codec-list mode and enters Privileged EXEC mode.
	Example: Router(config-sbc-sbe-codec-list)# end	
Step 8	<pre>show sbc service-name sbe codec-list list-name</pre>	Displays detailed information about the codec lists configured on the SBE.
	Example: Router# show sbc mysbc sbe codec-list my_codecs	If the list name is omitted, for example, my_codecs, then details are displayed for all codec lists on the SBE.

Configuring a CAC Policy to Use a Codec List

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name
- 6. first-cac-table table-name
- 7. cac-table table-name
- 8. table-type {policy-set | limit {list of limit tables}}
- 9. entry entry-id
- **10.** cac-scope {*list of scope options*}
- **11. action** [next-table *goto-table-name* | cac-complete]
- **12.** codec-restrict-to-list *list-name*
- 13. complete
- 14. end
- 15. show sbc service-name sbe cac-policy-set id table name entry entry

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Configures the submode of the SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the submode of CAC policy.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	

	Command or Action	Purpose
Step 5	first-cac-scope scope-name	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope global	Note The first-cac-scope definition is only relevant if the table type configured by the first-cac-table command is a Limit table. In that case, the scope of the first-cac-table is determined by first-cac-scope. If the first-cac-table is a Policy Set table, the first-cac-scope is ignored and defaults to global.
		The <i>scope-name</i> argument configures the scope at which limits should be initially defined. Possible values are:
		• adj-group
		• call
		• category
		• dst-account
		• dst-adj-group
		• dst-adjacency
		• dst-number
		• global
		• src-account
		• src-adj-group
		• src-adjacency
		• src-number
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies.
Step 6	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table	
Step 7	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table first_policy_table	SBE policy set.

	Command or Action	Purpose
Step 8	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
		• <i>account</i> —Compare the name of the account.
		• <i>adj-group</i> —Compare the name of the adjacency group.
		• <i>adjacency</i> —Compare the name of the adjacency.
		• <i>all</i> —No comparison type. All events match this type.
		• <i>call-priority</i> —Compare with call priority.
		• <i>category</i> —Compare the number analysis assigned category.
		• <i>dst-account</i> —Compare the name of the destination account.
		• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
		• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
		• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
		• <i>event-type</i> —Compare with CAC policy event types.
		• <i>src-account</i> —Compare the name of the source account.
		• <i>src-adj-group</i> —Compare the name of the source adjacency group.
		• <i>src-adjacency</i> —Compare the name of the source adjacency.
		• <i>src-prefix</i> —Compare the beginning of the calling number string.
		Note For Limit tables, the event or message or call matches only a single entry.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
		Note For Policy Set tables, the event or call or message is applied to all entries in this table.

	Command or Action	Purpose
Step 9	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 10	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each of the entries at which limits are applied in a policy set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope call	- account —Events that are from the same account.
		 adjacency—Events that are from the same adjacency.
		 adj-group—Events that are from members of the same adjacency group.
		- call—Scope limits are per single call.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		- dst-number —Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server).

	Command or Action	Purpose
Step 11	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to perform after this entry in an admission control table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	• Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name argument</i> .
		• Stop processing for this scope using the cac-complete keyword.
Step 12	codec-restrict-to-list list-name	Configures CAC to restrict the codecs used in signaling a call to the set of codecs provided in the named list.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	If a codec list is empty, all codecs recognized by the SBE are allowed.
	# codec-restrict-to-list my_codecs	The no form of this command, or not setting this command, allows any recognized codecs to be used without restrictions.
		Note This command replaces any codec list that was set up by an earlier CAC entry. To clear all restrictions from an earlier CAC entry, you must configure a codec-restrict-to-list <i>list-name</i> , where <i>list-name</i> is the name of a list containing no codecs.
Step 13	complete	Completes the cac-policy.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 14	end	Exits the cac-policy-set configuration mode and enters Privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# end</pre>	
Step 15	<pre>show sbc service-name sbe cac-policy-set id table name entry entry</pre>	Displays detailed information for a specific entry in a CAC policy table, including any restricted codecs.
	Example: Router# show sbc mysbc sbe cac-policy-set 1 table standard_policy_list entry 1	

Packetization Time

The packetization time (p-time) is the time period a codec is applied to a media stream to build a single digital packet. When a call travels between two end points, both the codec and the p-time for that codec is negotiated. The SBC acts on the p-time attribute in the SDP based on the configuration in the codec list.

SBC adds the ptime attribute to an SDP offer or answer, when one is configured explicitly in a codec whitelist. The exception is if a minimum packetization time is configured as part of SBC policy. This configuration causes SBC to insert a=ptime lines into forwarded offers or answers.

SBC ensures that media clipping does not occur as a result of overestimating the ptime (and hence underestimating the bandwidth requirement).

If ptime is signaled explicitly on the offer and answer, it uses the lower of the two values to calculate the bandwidth allowance for both directions of media.

H.323 TCS Codecs Support

Cisco Unified Border Element (SP Edition) enables H.323 TCS Codecs support in Cisco IOS XE Release 2.5.1. This support provides the ability to announce media capabilities on behalf of a SIP endpoint to an H.323 endpoint by adding extra offered codecs in the H.245 Terminal Capability Set (TCS) message. You configure the additional codecs in the TCS message with Call Admission Control (CAC) policy commands.

H.323 TCS Codecs support applies to SIP to H.323 or H.323 to SIP interworking calls when the caller or callee uses SIP.

Information About H.323 TCS Codecs Support

H.323 TCS Codecs support adds extra capability in the H.245 Terminal Capability Set (TCS) message by providing the ability to configure offered codecs in the TCS message, on behalf of a SIP endpoint for SIP to H.323 or H.323 to SIP interworking calls when the caller or callee uses SIP.

A TCS message is an H.245 message transmitted by an H.323 endpoint sent by the SBC. The TCS message transmitted by the H.323 endpoint is used to indicate the H.323 endpoint's media capabilities, as well as the version of H.245, to the other party.

However, a SIP endpoint's SDP offer and answer may not announce all its capabilities because the SDP offer may be missing T.38 and other codecs, and the SDP answer may be missing many codecs. For example, a call originated by SIP endpoint A may not indicate the support of T.38, which would cause the H.323 endpoint B to be unaware of the capability. Therefore, endpoint B would not be able to send a RequestMode message to the SBC asking endpoint A to switch to the T38 codec.

The H.323 TCS Codecs support feature overcomes this gap by allowing you to add offered codecs in the TCS message that will announce the SIP caller or callee media capabilities to the H.323 endpoint. You do this with the **caller-media-caps** or **callee-media-caps** command in CAC table entry configuration mode. These two commands configure the SIP caller or callee media capability by assigning a codec list that is used to announce media capabilities on behalf of either a SIP caller or callee in a SIP to H.323 or H.323 to SIP interworking call.



In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** commands. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

You can also use the similar **tcs-extra-codecs** command in a CAC table entry to configure additional codecs in a codec list that sends an extra TCS message to the H.323 side. This command announces extra codecs capability to the H.323 endpoint on behalf of the SIP side, whether it is the SIP caller or SIP callee in a SIP to H.323 or H.323 to SIP interworking call.

You can use any one of the three commands independently.



After a codec list has been assigned, it may not be deleted until it is removed from the CAC entry. A codec list must exist before it can be assigned to an entry in a CAC table.

Codecs Preference and Reordering Support

SBC supports reordering of the codecs in a list, and assigning a priority to each codec. You can also apply a codec preference to a list in the CAC policy entry.

For information about restrictions for the Codecs Preference and reordering feature, see the?\$paranum>Restrictions for Codec Reordering? section on page 17-4.

When a SIP endpoint makes an SDP call, a list of codecs is provided for each media stream in an endpoint. The codecs are listed by payload types in the m= attribute in the order of highest to lowest preference. The SIP endpoint chooses the highest-priority codec that is acceptable to the SIP.

For example, a call uses G.711 and G.723 codec, but uses the G.711 where possible. To set the preference to G.711, the SIP places the PCMU and PCMA before G.723 in its m-line:

m=audio 1234 RTP/AVP 8 0 4.

Configuring Reordering

To reorder the codecs in a list, and to prioritize a codec within the list, configure the list of codecs, provide a description for the list, and then add the priority to the codecs in the list.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. codec list name
- 5. description text
- 6. codec codec-name [packetization-period packet-period [priority priority-value] | priority priority-value [packetization-period packet-period]]
- 7. end
- 8. show sbc service-name sbe codec-list list-name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	

	Command or Action	Purpose
Step 3	sbe	Configures the submode of the SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	codec list name	Creates a codec list.
	Example: Router(config-sbc-sbe)# codec list my_codecs	
Step 5	description text	Adds a description for the specified codec list using a readable text string format.
		The no form of this command removes the description.
	Example: Router(config-sbc-sbe-codec-list)# description Legitimate codes	This description is displayed when the show command is used for this codec list. It is also included for each codec list when a summary of all codec lists is displayed.
Step 6	<pre>codec codec-name [packetization-period packet-period [priority priority-value] priority priority-value [packetization-period packet-period]]</pre>	Adds a codec to a codec list, sets a minimum packetization period (optional) for the codec, and sets a priority for the codec that is used for reordering.
		The default codec preference priority is 100
	Example:	A smaller priority value indicates a higher priority, for example if the codec preference value is 1, which is the highest priority and places the codec at the top of the list.
	Router(config-sbc-sbe-codec-list)#codec G/23 priority 1	Using the no form of this command (without the packetization period) removes the named codec from the codec list.
		Note If the no form of this command includes the packetization period, only the packetization period for the codec is removed.
Step 7	end	Exits the codec-list mode and enters the Privileged EXEC mode.
	Example: Router(config-sbc-sbe-codec-list)# end	
Step 8	<pre>show sbc service-name sbe codec-list list-name</pre>	Displays detailed information about the codec lists configured on the SBE.
	Example: Router# show sbc mysbc sbe codec-list my_codecs	If the list name for example, my_codecs, is omitted, details are displayed for all the codec lists on the SBE.

Configuring Codec Preferences

To add codec preferences within a list, configure the CAC with a list of codecs, provide a description for the list, and then add the preference to the codecs.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name
- 6. cac-table table-name
- 7. table-type {policy-set | limit {list of limit tables}}
- 8. entry entry-id
- 9. codec-preference-list list-name
- 10. end
- 11. show sbc service-name sbe cac-policy-set id table name entry entry

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Configures the submode of the SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the submode of CAC policy.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	first-cac-table table-name	Configures the name of the first policy table to processed when performing the admission control stage of the policy.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table</pre>	

DETAILED STEPS

	Command or Action	Purpose
Step 6	<pre>cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# cac-table first policy table</pre>	Enters the cactable mode for configuring an admission control table (creating one, if necessary) within the context of an SBE policy set.

Command or Action	Purpose
<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
Example: Router(config-sbc-sbe-cacpolicy-cactable)#	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
table-type policy-set	• <i>account</i> —Compare the name of the account.
Command or Action Put Step 7 table-type (policy-set limit (list of limit tables)) Control of table Example: Router(config-sbc-sbe-cacpolicy-cactable)# The max average of table Router(config-sbc-sbe-cacpolicy-cactable)# 	• <i>adj-group</i> —Compare the name of the adjacency group.
	• <i>adjacency</i> —Compare the name of the adjacency.
	• <i>all</i> —No comparison type. All events match this type.
	• <i>call-priority</i> —Compare with call priority.
	• <i>category</i> —Compare the number analysis assigned category.
	• <i>dst-account</i> —Compare the name of the destination account.
	• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
	• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
	• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
	• <i>event-type</i> —Compare with CAC policy event types.
	• <i>src-account</i> —Compare the name of the source account.
	• <i>src-adj-group</i> —Compare the name of the source adjacency group.
	• <i>src-adjacency</i> —Compare the name of the source adjacency.
	• <i>src-prefix</i> —Compare the beginning of the calling number string.
	Note For Limit tables, the event or message or call matches only a single entry.
	Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
	When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
	Note For Policy Set tables, the event or call or message is applied to all entries in this table.

	Command or Action	Purpose
Step 8	entry entry-id	Enters the entry mode to create or modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	
Step 9	codec-preference-list list-name	Configures the CAC to set preferences for the codecs in the codecs list.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # codec-preference-list my_codecs</pre>	Using the no form of this command, or not setting this command, allows the recognized codecs, if any, to be used without any usage preference.
		Note This command replaces the codec-preference-list, if any, set up by an earlier CAC entry. To clear all the the preferences from an earlier CAC entry, configure a codec-preference-list <i>list-name</i> , where <i>list-name</i> is the name of a list containing no codecs.
Step 10	end	Exits the cac-policy-set configuration mode and enters the Privileged EXEC mode.
	Example: Router(config-sbc-sbe-cacpolicy)# end	
Step 11	<pre>show sbc service-name sbe cac-policy-set id table name entry entry</pre>	Displays detailed information pertaining to a specific entry in a CAC policy table, including codecs with preferences, if any.
	Example: Router# show sbc mysbc sbe cac-policy-set 1 table standard_policy_list entry 1	

For more information about the commands, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model.*

Configuration Examples—Configuring Codec Restriction

This section provides a sample configuration and output for configuring restrictions on codecs and configuring a CAC policy to use a codec list.

Example of Configuring Codecs

The following example shows the commands required to configure codec restriction.

Figure 17-1 contains three adjacencies (A, B, and C). Any calls involving A need to be configured to use only the G729 and PCMU (G.711) codecs with a minimal preferred packetization period of 10 milliseconds. However, calls between B and C can use any available codecs.

To create a codec list containing the specified codecs configured with a minimal preferred packetization period, use the following commands:

Router# configure terminal Router(config)# sbc mysbc Router(config-sbc)# sbe

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Router(config-sbc-sbe)# codec list allowable_codecs Router(config-sbc-sbe-codec-list)# description The set of codecs allowed on adjacency AdjA Router(config-sbc-sbe-codec-list)# codec g729 packetization-period 20 Router(config-sbc-sbe-codec-list)# codec pcmu packetization-period 10 Router(config-sbc-sbe-codec-list)# exit

After configuring codec restriction, you must configure a CAC policy to use the codec list. See ?\$paranum>Example of Configuring a CAC Policy to Use a Codec List? section on page 17-23.





Example of Configuring a CAC Policy to Use a Codec List

The following example shows the commands required to configure a CAC policy to use a codec list. To configure a codec list, see ?\$paranum>Example of Configuring Codecs? section on page 17-22.

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-table table1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope call
Router(config-sbc-sbe-cacpolicy)# cac-table table1
Router(config-sbc-sbe-cacpolicy-cactable)# match-type adjacency
Router(config-sbc-sbe-cacpolicy-cactable)# match-type adjacency
Router(config-sbc-sbe-cacpolicy-cactable)# match AdjA
Router(config-sbc-sbe-cacpolicy-cactable)# codec-restrict-to-list allowable_codecs
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy)# complete
```



The codec list command line interface (CLI) commands can only be entered at the per-call level in the CAC policy tables. If you configure a codec list at any other level the CAC policy set will not activate. However, a log is displayed after you complete the configuration and the policy set is marked as "complete".

Configuration Examples—H.323 TCS Codecs Support

The following example configures a codec list called "caller-media-caps-list" and assigns that list to the CAC table "cac-tbl-1" in entry 1 to announce media capabilities on behalf of the SIP caller:

```
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# codec list caller-media-caps-list
Router(config-sbc-sbe-codec-list)# codec t38
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table cac-tbl-1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-media-caps caller-media-caps-list
```

The following example configures a codec list called "callee-media-caps-list" and assigns that list to the CAC table "cac-tbl-1" in entry 1 to announce media capabilities on behalf of the SIP callee.

```
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# codec list callee-media-caps-list
Router(config-sbc-sbe-codec-list)# codec t38
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table cac-tbl-1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-media-caps callee-media-caps-list
```

The following example configures a codec list called "tcs-extra-caps-list" and assigns that list to the CAC table "cac-tbl-1" in entry 1 to announce extra codecs capability on behalf of the SIP side, whether it is the SIP caller or callee:

```
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# codec list tcs-extra-caps-list
Router(config-sbc-sbe-codec-list)# exit
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table cac-tbl-1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable)# tcs-extra-codecs tcs-extra-caps-list
```

The following example show command lists the codec list names for caller and callee media capabilities and extra TCS capabilities for entry 1 in a CAC policy table:

```
Router# show sbc mySBC sbe cac-policy-set 1 table cac-tbl-1 entry 1
SBC Service mySBC
```

```
CAC Policy Set 1
Active policy set: No
Description:
Averaging period: 60 sec
First CAC table:
```

```
First CAC scope: global
Table name: cac-tbl-1
 Description:
 Table type: policy-set
 Total call setup failures (due to non-media limits): 0
 Entry 1
 CAC scope:
 CAC scope prefix length: 0
 Action: Not set
 Number of call setup failures (due to non-media limits): 0
 Max calls per scope: Unlimited Max call rate per scope: Unlimited
 Max in-call rate:
                      Unlimited
                                     Max out-call rate: Unlimited
 Max reg. per scope: Unlimited
                                    Max reg. rate per scope: Unlimited
 Max channels per scope: Unlimited
                                    Max updates per scope: Unlimited
                                     Early media direction: Both
 Early media:
              Allowed
                      None
 Early media timeout:
                                      Transcoder per scope:
                                                             Allowed
 Callee Bandwidth-Field: None
                                      Caller Bandwidth-Field: None
 Media bypass:
                               Allowed
 Renegotiate Strategy:
                               Delta
 Max bandwidth per scope:
                              Unlimited
 Caller media capabilities:
                               caller-media-caps-list
 Callee media capabilities:
                                callee-media-caps-list
 Extra TCS capabilities:
                                tcs-extra-caps-list
```

Configuration Example—Defining a Codec using Dynamic Codec Configuration

The following example shows the commands required to define a custom codec from a codec included in SBC (system codec):

```
Router# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# codec custom G726-40-4 id 4
Router(config-sbc-sbe-codec-def)# rate 64000
Router(config-sbc-sbe-codec-def)# packet time 100
Router(config-sbc-sbe-codec-def)# bandwidth 128000
Router(config-sbc-sbe-codec-def)# sample size 4
Router(config-sbc-sbe-codec-def)# channels 16
Router(config-sbc-sbe-codec-def)# max-frames-per-packet 12
Router(config-sbc-sbe-codec-def)# media video
Router(config-sbc-sbe-codec-def)# options transcode
Router(config-sbc-sbe-codec-def)# type sampling
```

L



SDP Bandwidth Field Features

Cisco Unified Border Element (SP Edition) supports the Bandwidth Field Interworking and the Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation features in the unified model.

In a Session Initiation Protocol (SIP) message exchange, Cisco Unified Border Element (SP Edition) uses the parameters defined in the Session Description Protocol (SDP) bandwidth-fields (b-line) for calculating the media pinhole bandwidth. During SIP message exchange, the SDP may contain both the bandwidth-fields and coder/decoder (CODEC) information. In that case, Cisco Unified Border Element (SP Edition) would use the bandwidth-field value to allocate sufficient bandwidth for the media pinhole.

During deployment, there might be some endpoints for which it would be better to set the media pinhole bandwidth using a CODEC definition in the Session Description Protocol (SDP) messages instead of using b-line.

The Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation feature allows you to set a media bandwidth flag in a Call Admission Control (CAC) policy entry to ignore the b-line and use CODEC for calculating the media pinhole bandwidth.

Cisco Unified Border Element (SP Edition) supports Bandwidth Field Interworking by supporting the ability to determine how bandwidth lines are translated in the outbound Session Description Protocol (SDP) sent to the caller and callee with bandwidth line passthrough using Application Specific Maximum (AS) and Transport Independent Application Specific Maximum (TIAS) conversion.



For Cisco IOS XE Release 2.4 and later, these features are supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SDP Bandwidth Field Features

Release	Modification
Cisco IOS XE Release	Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth
2.4	Allocation feature was introduced on the Cisco IOS XR, along with the unified model

Cisco IOS XE Release 2.5	Bandwidth Field Interworking feature was introduced on the Cisco IOS XR.
Cisco IOS XE Release 3.1S	Media Bandwidth Policy feature was added.
Cisco IOS XE Release 3.2S	Per-adjacency codec string interworking feature was added.

Contents

This chapter contains the following sections:

- Prerequisites for Implementing SDP Bandwidth Field Features, page 18-2
- Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation Feature, page 18-2
- Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation Configuration: Example, page 18-7
- Information About Media Bandwidth Policy, page 18-8
- Configuring Media Bandwidth Policy, page 18-9
- Bandwidth Field Interworking Feature, page 18-17
- Bandwidth Field Interworking Configuration: Examples, page 18-22
- Per-Adjacency Codec String Interworking, page 18-23

Prerequisites for Implementing SDP Bandwidth Field Features

The following prerequisite is required to implement SDP Bandwidth Field features:

Before implementing SDP Bandwidth Field features, Cisco Unified Border Element (SP Edition) must already be configured.

Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation Feature

The following sections are in the "Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation" feature:

- Information About Calculating Bandwidth in SIP Calls, page 18-3
- Configuring Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation, page 18-3
- Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation Configuration: Example, page 18-7

If these lines are not present, the SBC calculates the baseline bandwidth by inspecting each of the possible CODECs in the stream and calculating the baseline bandwidth based on them. The bandwidth allocated per CODEC is configurable.

calculate the baseline bandwidth required for the media stream.

Information About Calculating Bandwidth in SIP Calls

The SBC then adjusts the baseline bandwidth to take into account any necessary packetization and Real Time Control Protocol (RTCP) bandwidth overheads.

The SBC analyzes each media stream in a call and calculates the bandwidth required. For SIP calls containing SDP, the SBC looks for any b=CT, b=AS, or b=TIAS lines. If present, these lines are used to

Certain endpoints do not conform to the bandwidth requirements that the SBC calculates for a media stream, for example:

- Endpoints that start renegotiating the bandwidth for a call can start using additional bandwidth before the renegotiation is complete.
- Endpoints that request an incorrect bandwidth for secure media using the b-line, because they do not take into account the increased payload size required for the encryption.
- Endpoints that transmit data in multiple formats in parallel (such as high and low definition video using different payloads in a single stream) without taking into account all formats when calculating the bandwidth requirements.

To allow interoperation with these endpoints without dropping packets, the SBC allows the per-CODEC bandwidth to be configurable. This allows the SBC administrator to set a suitably large maximum value for CODECs supported by these endpoints. This is sufficient because the endpoints in question use a well known set of CODECs.

However, if the endpoint includes an explicit bandwidth (b-) line, then the SBC uses that to calculate the bandwidth instead of the maximum value. The Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation feature uses the **media bandwidth-fields ignore** command to set a media flag in a Call Admission Control (CAC) policy entry to ignore the b-line and use CODEC to calculate the bandwidth.

Configuring Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation

This task configures the Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table *table-name*
- 7. table-type {policy-set | limit {list of limit tables}}

- 8. entry entry-id
- **9.** cac-scope {*list of scope options*}
- 10. media bandwidth-fields ignore
- **11. action** [next-table *goto-table-name* | cac-complete]
- 12. exit
- 13. exit
- 14. complete

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table StandardListByAccount	
Step 6	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	SBE policy set.

	Command or Action	Purpose
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
		• <i>account</i> —Compare the name of the account.
		• <i>adj-group</i> —Compare the name of the adjacency group.
		• <i>adjacency</i> —Compare the name of the adjacency.
		• <i>all</i> —No comparison type. All events match this type.
		• <i>call-priority</i> —Compare with call priority.
		• <i>category</i> —Compare the number analysis assigned category.
		• <i>dst-account</i> —Compare the name of the destination account.
		• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
		• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
		• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
		• <i>event-type</i> —Compare with CAC policy event types.
		• <i>src-account</i> —Compare the name of the source account.
		• <i>src-adj-group</i> —Compare the name of the source adjacency group.
		• <i>src-adjacency</i> —Compare the name of the source adjacency.
		• <i>src-prefix</i> —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
Step 8	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy- cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Choose a scope at which CAC limits are applied within each of the entries in a policy set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope src-adjacency	- account —Events that are from the same account.
		 adjacency—Events that are from the same adjacency. adj-group—Events that are from members of the same adjacency group.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		- dst-number—Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server).
Step 10	media bandwidth-fields ignore	Sets the media flag to ignore the b-line and use CODEC to calculate the baseline bandwidth required for the media
	Example:	stream.
	<pre>Router(config-sbc-sbe-cacpolicy-cactable-entry) # media bandwidth-fields ignore</pre>	

	Command or Action	Purpose
Step 11	action [next-table goto-table-name cac-complete]	Configures the action to perform after this entry in an admission control table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	 Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name argument</i>. Stop processing for this scope using the cac-complete keyword.
Step 12	exit	Exits from entry to cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 13	exit	Exits from cactable to cacpolicy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 14	complete	Completes the CAC policy set when you have committed the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	

Option to Use CODEC Instead of Bandwidth-Field for Media Bandwidth Allocation Configuration: Example

The following example shows how to set the media flag to ignore the b-line and use CODEC to calculate the baseline bandwidth required for the media stream:

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-table StandardListByAccount
Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable-entry)# media bandwidth-fields ignore
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
```

Information About Media Bandwidth Policy

Previous to this release, SBC disabled or rejected media calls that exceeded the bandwidth allowed by the Call Admission Control (CAC). However, some applications, such as Telepresence, require other options. SBC now provides the ability to diminish the video stream to a lower bandwidth, while allowing the audio stream to remain unchanged.

In this release, bandwidth restrictions are enhanced by allowing the user to configure one of three media bandwidth options, using the **media police** command:

- strip
- reject
- degrade

These options can be configured for all media types or for video only if desired.

strip

If an individual media stream exceeds the bandwidth limit for a call, that media stream is disabled by setting the port to zero (0). If after the above stage has completed, the sum of the bandwidths of all remaining streams exceeds the bandwidth limit for a call, the request is rejected.

When the port is set to zero (0), the call is ended and the following message is displayed on the screen:

incompatible sites

reject

If an individual media stream exceeds the bandwidth limit for a call, the request is rejected. If the sum of the bandwidths of all media streams exceeds the bandwidth limit for a call, the request is rejected.

degrade

If a media stream exceeds the bandwidth limit for a call, the video stream is downgraded to a lower (non-zero) bandwidth that brings the media stream within the bandwidth limit for the call.

Note Only the video stream is downgraded. Audio streams are not downgraded. If the audio stream exceeds the bandwidth for a call, the media stream cannot be downgraded.

Restrictions

The degrade option is not supported on H.323 calls.

Using the degrade option may cause a 2 to 5 percent performance degradation.

Configuration

You configure the Media Bandwidth Policy by configuring the media policy mode, using the **media policy** command, and by configuring the minimum bandwidth for the for the analog-to-digital codec (enCOder/DECoder) hardware, using the **bandwidth** command.



The codec name must be one of the system codecs that SBC can recognize. To see a list of the system codecs, use the **show sbc sbc sbc codecs** command.

The **max-bandwidth-per-scope** command specifies the maximum bandwidth limit for all media streams in all directions, including packet overheads.

The **bandwidth min** command specifies the unidirectional, minimum bandwidth limit bandwidth and does not include packet overhead.
See the ?\$paranum>Configuring Media Bandwidth Policy? section on page 18-9 for the configuration steps and the ?\$paranum>end? section on page 18-30 for configuration examples.

Configuring Media Bandwidth Policy

This section provides the following step procedures:

- Configuring the Media Policy Mode, page 18-9
- Configuring the Codec Minimum Bandwidth, page 18-10
- Configuring the Maximum Bandwidth Per Scope, page 18-12

Configuring the Media Policy Mode

Use the following procedure to configure the media policy mode.

SUMMARY STEPS

- 1. config
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type policy-set
- 7. entry entry-id
- 8. media police strip | reject | degrade
- 9. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	config	Enters global configuration mode.
	Example: Router# config	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example:	
	Router(config) # sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example:	
	Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.
Step 6	table-type policy-set	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	
Step 7	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 8	media police strip reject degrade	Configures how SBC handles media streams that exceed bandwidth limits for media calls.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # media police degrade	
Step 9	end	Exits configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Configuring the Codec Minimum Bandwidth

Use the following procedure to configure the minimum bandwidth for the codec.

SUMMARY STEPS

- 1. config
- 2. sbc sbc-name
- 3. sbe
- 4. codec custom custom-name id
- 5. type variable
- 6. media video
- 7. bandwidth min bandwidth-value
- 8. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# config	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	codec custom custom-name id	Specifies the name of the custom analog-to-digital codec (enCOder/DECoder) and enters codec definition mode.
	Example: Router (config-sbc-sbe)# codec custom h263-c id 96	
Step 5	type variable	Sets the type of the codec to variable.
	Example: Router (config-sbc-sbe-codec-def)# type variable	
Step 6	media video	Sets the media type to video.
	Example: Router (config-sbc-sbe-codec-def)# media video	
Step 7	bandwidth min bandwidth-value	Sets the minimum bandwidth for the codec.
	Example: Router (config-sbc-sbe-codec-def)# bandwidth min 328000	Note The bandwidth min command specifies the unidirectional, minimum bandwidth limit and does not include packet overhead.
Step 8	end	Exits configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Configuring the Maximum Bandwidth Per Scope

Use the following procedure to configure the bandwidth limit for all media streams.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. description description
- 6. first-cac-table table-name
- 7. first-cac-scope scope-name
- 8. cac-table table-name
- 9. table-type policy-set
- 10. entry entry-id
- 11. max-bandwidth-per-scope bandwidth
- **12**. action cac-complete
- 13. media police degrade
- 14. complete
- 15. codec system sys-codec id payload id
- 16. type variable
- 17. bandwidth min bandwidth-value
- 18. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# config t	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example:	
	Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	description description	Configures descriptive text for this policy set.
	Example: Router(config-sbc-sbe-cacpolicy)# description bandwidth degrade	
Step 6	<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table my_table</pre>	Configures the name of the first policy table to process. A CAC policy may have many tables configured. To start the application of the CAC policy, the first table that is used needs to be defined.
		<i>table-name</i> —The admission control table that should be processed first.
Step 7	<pre>first-cac-scope scope-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope call</pre>	Configures scope at which limits should be initially defined when performing the admission control stage of the policy. Each CAC policy has a scope that is applied to it. This CAC policy applies on a per call basis.
		scope-name has one of the following values:
		• adj-group —Limits for events from members of the same adjacency group.
		• call —Limits are per single call.
		• category—Limits per category.
		• dst-account —Limits for events sent to the same account.
		• dst-adj-group —Limits for events sent to the same adjacency group.
		• dst-adjacency —Limits for events sent to the same adjacency.
		• dst-number —Limits for events that have the same adjacency number.
		• global —Limits are global (May not be combined with any other option).
		• src-account —Limits for events from the same account.
		• src-adj-group —Limits for events from the same adjacency group.
		• arc-adjacency —Limits for events from the same adjacency.
		• src-number —Limits for events that have the same source number.

	Command or Action	Purpose
Step 8	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set.
	Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.
Step 9	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set. For Policy Set tables, the event or call or message is applied to all entries in this table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set</pre>	
Step 10	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 11	max-bandwidth-per-scope bandwidth	Configures the maximum limit for the bandwidth in bps, Kbps, Mbps or Gbps for an entry in an admission control table.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-bandwidth-per-scope 6000000 bps	Note The max-bandwidth-per-scope command specifies the maximum bandwidth limit for all media streams in all directions, including packet overheads.
Step 12	<pre>action cac-complete Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	Configures the action to perform after this entry in an admission control table. In this case, stop processing for this scope using the cac-complete keyword.
Step 13	media police strip reject degrade	Configures how SBC handles media streams that exceed bandwidth limits for media calls. In this case, degrade .
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # media police degrade</pre>	
Step 14	complete	Completes the CAC-policy or call-policy set after committing the full set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) complete	
Step 15	codec system sys-codec id payload id	Enters codec definition mode to modify an existing codec.
	Example: Router(config-sbc-sbe-cacpolicy)# codec system H263 id 34	

	Command or Action	Purpose
Step 16	type variable	Sets the type of the codec to variable.
	Example: Router (config-sbc-sbe-codec-def)# type variable	
Step 17	bandwidth min bandwidth-value	Sets the minimum bandwidth for the codec.
	Example: Router (config-sbc-sbe-codec-def)# bandwidth min 370000	Note The bandwidth min command specifies the unidirectional, minimum bandwidth limit bandwidth and does not include packet overhead.
Step 18	end	Exits configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Media Policy Configuration: Examples

This section provides the following configuration examples:

- Media Policy Mode Configuration: Example, page 18-16
- Codec Minimum Bandwidth Configuration: Example, page 18-16
- Maximum Bandwidth Per Scope Configuration: Example, page 18-16

Media Policy Mode Configuration: Example

The following example shows how to configure SBC to degrade media streams to lower bandwidths when requests exceed bandwidth limits.

```
Router# config t
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table cac-tbl-1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# media police degrade
Router(config-sbc-sbe-cacpolicy-cactable-entry)#
```

Codec Minimum Bandwidth Configuration: Example

The following example shows how to configure the maximum bandwidth limit to 400,000 bps for media calls:

```
Router# config t
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# codec system H263 id 34
Router(config-sbc-sbe-codec-def)# bandwidth 400000
```

The following example shows how to configure the minimum bandwidth limit to 328,000 bps, specifically for video type media calls:

```
Router# config t
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router (config-sbc-sbe)# codec custom h263-c id 96
Router (config-sbc-sbe-codec-def)# type variable
Router (config-sbc-sbe-codec-def)# media video
Router (config-sbc-sbe-codec-def)# bandwidth min 328000
```

Maximum Bandwidth Per Scope Configuration: Example

The following example shows how to configure the bandwidth limit for all media streams:

```
Router# config t
Router(config)# sbc SBC1
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# description bandwidth degrade
Router(config-sbc-sbe-cacpolicy)# first-cac-table my_table
Router(config-sbc-sbe-cacpolicy)# first-cac-scope call
```

```
Router(config-sbc-sbe-cacpolicy)# cac-table testSecure
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth-per-scope 6000000 bps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# media police degrade
Router(config-sbc-sbe-cacpolicy-cactable-entry) complete
Router(config-sbc-sbe-cacpolicy)# codec system H263 id 34
Router(config-sbc-sbe-codec-def)# type variable
Router(config-sbc-sbe-codec-def)# bandwidth min 370000
Router(config-sbc-sbe-cacpolicy-cactable-entry)# end
```

Bandwidth Field Interworking Feature

The following sections are in the Bandwidth Field Interworking feature:

- Information About Bandwidth Field Interworking, page 18-17
- Configuring Bandwidth Field Interworking, page 18-18
- Bandwidth Field Interworking Configuration: Examples, page 18-22

Information About Bandwidth Field Interworking

Cisco Unified Border Element (SP Edition) supports Bandwidth Field Interworking by supporting the ability to determine how bandwidth lines are configured in the outbound Session Description Protocol (SDP). Cisco Unified Border Element (SP Edition) supports the Application Specific Maximum (AS) and Transport Independent Application Specific Maximum (TIAS) bandwidth modifiers in the SDP.

The SDP includes an optional bandwidth attribute with the following syntax, according to RFC 3556, Session Description Protocol (SDP) Bandwidth Modifiers:

b=<modifier>:<bandwidth-value>

The <modifier> is an alphanumeric word that indicates the bandwidth to be used by the media or session. The <bandwidth- value> default is kilobits per second.

The AS bandwidth modifier is used to specify the total bandwidth for a single media stream from one source.

The TIAS bandwidth value is the maximum bandwidth required by the SDP session level or media stream without counting IP or other transport layers like TCP or UDP (RFC 3890).

Cisco Unified Border Element (SP Edition) supports translation between the AS and TIAS bandwidth formats which are configured for each adjacency by means of the following commands:

- caller-bandwidth-field [as-to-tias | tias-to-as]
- callee-bandwidth-field [as-to-tias | tias-to-as]

When you configure the bandwidth line to the as-to-tias setting, this causes the SBC, in an outbound SDP offer, to convert a b=AS line into a b=TIAS line. If there are multiple b=AS lines, only the first line is converted into a b=TIAS line and the rest are ignored.

Translating from a bandwidth modifier of AS into TIAS can be useful in the following situations:

• If operating with upstream devices that only support the AS bandwidth modifier, use of the TIAS bandwidth modifier downstream may improve the accuracy of bandwidth calculations in the network. In some network scenarios, use of the AS bandwidth modifier may lead to incorrect bandwidth calculations, for example, if routing between an IPv4 and IPv6 network (see RFC3890).

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• For interoperability purposes—if there are downstream devices that do not understand the AS bandwidth modifier.

When you configure the bandwidth line to the tias-to-as setting, this causes the SBC, in an outbound SDP offer, to convert a b=TIAS line into a b=AS line if there is not already a b=AS line associated with that SDP media descriptor. If there are multiple b=TIAS lines, only the first is converted into a b=AS line and the rest are ignored.

Translating from a bandwidth modifier of TIAS into AS can be useful in the following situation:

• For interoperability purposes—if there are downstream nodes that do not understand the TIAS bandwidth modifier.

The SBC supports translation between these two formats. If bandwidth line conversion is enabled for the offerer-side adjacency, then an answer has its bandwidth lines converted to the specified format before being sent back to the offerer. Similarly, If bandwidth line conversion is enabled for the answer-side adjacency, then an offer has its bandwidth lines converted to the specified format before being sent to the answer.

The same rules are applied to translation of bandwidth lines in the answer and translation of bandwidth lines in the offer. The rules are as follows:

- The SBC conforms to whichever outgoing bandwidth line format is configured. If the outgoing adjacency is configured to prefer a specific style of bandwidth line format, then that format is used. Thus any AS or TIAS bandwidth lines are translated to that format.
- If the offerer's adjacency has no configured bandwidth line format preference, but a translation in bandwidth line format was made on the offer to accommodate the answer-side adjacency's preference, then the reverse translation is done on the answer.

For example, the answer adjacency is configured to translate to TIAS bandwidth lines. The offerer's adjacency has no preference. The offerer makes an SDP offer containing b=AS lines, which are then converted by the SBC to the b=TIAS style for the outgoing offer.

The answerer responds with a b=TIAS line which represents an increased bandwidth requirement. This increased bandwidth line is translated back to b=AS before being sent to the offerer because that is the style the offerer last offered with.

Configuring Bandwidth Field Interworking

This task configures the Bandwidth Field Interworking feature.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure
- **2. sbc** *service-name*
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name

- 6. cac-table table-name
- 7. table-type {policy-set | limit {list of limit tables}}
- 8. entry entry-id
- **9.** cac-scope {*list of scope options*}
- 10. caller-bandwidth-field [as-to-tias] [tias-to-as]
- 11. callee-bandwidth-field [as-to-tias] [tias-to-as]
- **12. action** [next-table *goto-table-name* | cac-complete]
- 13. exit
- 14. exit
- 15. complete

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table StandardListByAccount	
Step 6	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	SBE policy set.

Command or Action	Purpose
<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set</pre>	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible available Limit tables are:
	• <i>account</i> —Compare the name of the account.
	• <i>adj-group</i> —Compare the name of the adjacency group.
	• <i>adjacency</i> —Compare the name of the adjacency.
	• <i>all</i> —No comparison type. All events match this type.
	• <i>call-priority</i> —Compare with call priority.
	• <i>category</i> —Compare the number analysis assigned category.
	• <i>dst-account</i> —Compare the name of the destination account.
	• <i>dst-adj-group</i> —Compare the name of the destination adjacency group.
	• <i>dst-adjacency</i> —Compare the name of the destination adjacency.
	• <i>dst-prefix</i> —Compare the beginning of the dialed digit string.
	• <i>event-type</i> —Compare with CAC policy event types.
	• <i>src-account</i> —Compare the name of the source account.
	• <i>src-adj-group</i> —Compare the name of the source adjacency group.
	• <i>src-adjacency</i> —Compare the name of the source adjacency.
	• <i>src-prefix</i> —Compare the beginning of the calling number string.
	Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
	When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
entry entry-id	Enters the mode to create or modify an entry in an admission control table.
Example:	
Router(config-sbc-sbe-cacpolicy- cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Choose a scope at which CAC limits are applied within each entry in a Policy Set table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # cac-scope src-adjacency</pre>	<i>list of scope options</i> —Specifies one of the following strings used to match events:
		• <i>account</i> —Events that are from the same account.
		• <i>adjacency</i> —Events that are from the same adjacency.
		• <i>adj-group</i> —Events that are from members of the same adjacency group.
		• <i>call</i> —Scope limits are per single call.
		• <i>category</i> —Events that have same category.
		• <i>dst-account</i> —Events that are sent to the same account.
		• <i>dst-adj-group</i> —Events that are sent to the same adjacency group.
		• <i>dst-adjacency</i> —Events that are sent to the same adjacency.
		• <i>dst-number</i> —Events that have the same destination.
		• <i>global</i> —Scope limits are global
		• <i>src-account</i> —Events that are from the same account.
		• <i>src-adj-group</i> —Events that are from the same adjacency group.
		• <i>src-adjacency</i> —Events that are from the same adjacency.
		• <i>src-number</i> —Events that have the same source number.
Step 10	caller-bandwidth-field [as-to-tias] [tias-to-as] Example:	Configures the SBC to convert a b=AS line format into a b=TIAS line format or a b=TIAS line format into a b=AS line format in an outbound Session Description Protocol (SDP) sent to the caller.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-bandwidth-field as-to-tias	AS = Application Specific Maximum
		TIAS = Transport Independent Application Specific Maximum
Step 11	callee-bandwidth-field [as-to-tias] [tias-to-as] Example:	Configures the SBC to convert a b=AS line format into a b=TIAS line format or a b=TIAS line format into a b=AS line format in an outbound Session Description Protocol (SDP) sent to the callee.
	Router (config-sbc-sbe-cacpolicy-cactable-entry)	AS = Application Specific Maximum
	# callee-bandwidth-field tias-to-as	TIAS = Transport Independent Application Specific Maximum

	Command or Action	Purpose
Step 12	<pre>action [next-table goto-table-name cac-complete]</pre>	Configures the action to perform after this entry in an admission control table. Possible actions are:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	 Identify the next CAC table to process using the next-table keyword and the <i>goto-table-name</i> argument. Stop processing for this scope using the cac-complete keyword.
Step 13	exit	Exits from entry to cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 14	exit	Exits from cactable to cacpolicy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 15	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # complete</pre>	

Bandwidth Field Interworking Configuration: Examples



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

The following example shows that the SBC is configured to convert an AS bandwidth line format into a TIAS bandwidth line format on the offerer-side adjacency (caller side), and to convert a TIAS bandwidth line format into an AS bandwidth line format on the answerer-side adjacency (callee side):

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table 1
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-bandwidth-field as-to-tias
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-bandwidth-field tias-to-as
```

The following example lists detailed information, including caller and callee bandwidth field information, for entry 1 of CAC table 1:

Router# show sbc mysbc sbe cac-policy-set 1 table 1 entry 1

```
SBC Service "mysbc"
```

CAC Policy Set 1 Active policy set: Yes Description: Averaging period: 60 sec First CAC table: 1 First CAC scope: global Table name: cacTable Description: Table type: policy-set Total call failures: 0 Entry 1 CAC scope: Action: CAC complete Number of calls rejected: 0 Max calls per scope: Unlimited Max call rate per scope: Unlimited Max in-call rate: Unlimited Max out-call rate: Unlimited Max reg. rate per scope: Unlimited Unlimited Max reg. per scope: Max channels per scope: Unlimited Max updates per scope: Unlimited Early media: Allowed Early media timeout: None Early media direction: Both Transcoder per scope: Allowed Caller Bandwidth-Field: AS-to-TIAS Callee Bandwidth-Field: TIAS-to-AS Media bypass: Allowed Media flag: Ignore bandwidth-fields (b=) Renegotiate Strategy: Max bandwidth per scope: Delta Unlimited SRTP Transport: Trusted-Only (by default) Standard Caller hold setting: Callee hold setting: Standard Callee privacy setting:StandardCallee privacy setting:Never hideCaller voice QoS profile:Default Caller video OoS profile: Default Caller sig QoS profile: Default Callee voice QoS profile: Default Callee video QoS profile: Default Callee sig QoS profile: Default Restrict codecs to list: Default Restrict caller codecs to list: Default Restrict callee codecs to list: Default Caller inbound SDP policy: None Caller outbound SDP policy: None None Callee inbound SDP policy: Callee outbound SDP policy: None

Per-Adjacency Codec String Interworking

The following sections are in the Per-Adjacency Codec String Interworking feature:

- Information about Per-Adjacency Codec String Interworking, page 18-24
- Restrictions for Per-Adjacency Codec String Interworking, page 18-24
- Configuring Per-Adjacency Codec String Interworking, page 18-24
- Configuration Example for Per-Adjacency Codec String Interworking, page 18-30

Information about Per-Adjacency Codec String Interworking

From Cisco IOS Release 3.2S, the SBC can interpret non-standard SDP, and convert codecs between different non-standard forms of SDP or convert non-standard SDP to standard SDP, so that different non-standard devices can interwork through the SBC.

The SBC works through each codec in the message and determines whether the codec name received on the corresponding inbound SDP is a standard name or a variant, and then converts the codec:

- If the codec is standard, SBC searches through the variant list looking for a matching variant of that standard codec. The matching variant, if found, is converted or passed through unchanged.
- If the codec is a variant, SBC searches through the profile to see if that variant is listed.
 - Listed variant passes through unchanged
 - Unlisted variant can get converted to a matching variant of the same standard codec.
 - Unlisted variant that does not have a matching variant of the same standard codec gets converted to the standard representation

In the following scenarios, the codec convert should be avoided and SBC should use the standard SDP form:

- To ensure consistency of the H.248 interface, SDP fragments sent by SBC-SIG in H.248 commands to the media gateways (MGs) must use standard representations. The H.248 interface must be specified to facilitate interoperability with MGs and make it easy for the third party MGs to implement the SBC H.248 profile.
- To ensure consistency of the billing interface, the SDP fragments stored by SBC-SIG in the XML billing records must use standard representations. It is easy for the third party billing servers to parse XML billing records.
- Codec strings do not appear in H.245 signaling messages, an enumerated type is used to represent the codec. Therefore, the codec convert is only applicable for SIP outbound adjacency.

Restrictions for Per-Adjacency Codec String Interworking

The Per-Adjacency Codec String Interworking feature has the following restrictions:

- When a particular variant is used for a given codec on passing through an Offer, the same variant may not be used when passing through the Answer.
- For a given side of the call, SBC cannot be configured to interpret the incoming SDP using one variant but convert outgoing SDP based on another.
- You cannot define a variant that uses a standard IANA codec string because the SBC supports only those variants that use non-standard strings.
- If two variants map to the same standard codec, the transcoder does not convert between them. For example, the SBC cannot transcode between G7231H and G7231L, even though the endpoints can perceive those to be different codecs.

Configuring Per-Adjacency Codec String Interworking

This section explains the following configurations for Per-Adjacency Codec String Interworking feature:

• Configuring Codec Variant Conversion, page 18-25

• Configuring Codec on CAC Policy Set, page 18-27

Configuring Codec Variant Conversion

This task explains how to configure codec variant conversion on the SBC:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. codec system sys-codec id payload-id
- 5. **fmtp** *fmtp*-*string*
- 6. exit
- 7. codec variant codec variant-name
- 8. variant variant-codec-encoded-name
- 9. standard standard-codec-name
- **10. fmtp** *fmtp*-*string*
- 11. exit
- 12. codec variant profile profile-name
- **13**. **variant** *variant-name*
- 14. end
- **15.** show sbc *service-name* sbe codecs variant [profile]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	codec system sys-codec id payload-id	Enters the codec definition mode to modify an existing codec.
	Example:	• <i>sys-codec</i> —The codec included in the SBC.
	Router(config-sbc-sbe)# codec system G723 id 4	• id <i>payload-id</i> —Static payload id. Value can be from 0 to 96.
Step 5	fmtp fmtp-string	Configures the default value of Format-Specific Parameters (FMTP) for SDP.
	Example: Router(config-sbc-sbe-codec-def)# fmtp	• <i>fmtp-string</i> —The FMTP string for SDP, in the name=value format.
	annexa=yes	Note To view the default FMTP values associated with variants, use the show sbc sbe codecs variant command.
Step 6	exit	Exits from the codec definition mode and enters into the SBE entity mode.
	Example:	
C4 7	Router(config-sbc-sbe-codec-def)# exit	
Step 7	codec variant codec variant-name	delete a codec variant.
	Example:	• <i>variant-name</i> —The codec variant name.
	Router(config-sbc-sbe)# codec variant codec G723-H-1	
Step 8	variant variant-codec-encoded-name	Defines the encoded codec variant name.
	Example: Router(config-sbc-sbe-codec-var-codec)# variant G723-H-1	• <i>variant-codec-encoded-name</i> —The variant nonstandard codec string.
		Note '#' is reserved for base variants. Therefore, the variant name cannot start with '#'
Step 9	standard standard-codec-name	Defines the standard codec variant name.
	Example: Router(config-sbc-sbe-codec-var-codec)# standard G723	• <i>standard-codec-name</i> —The standard system codec name.
Step 10	fmtp fmtp-string	Define the FMTP parameters for the codec variant.
	Example:	• <i>fmtp-string</i> —The FMTP string in the name=value format.
	Router(config-sbc-sbe-codec-var-codec)# fmtp bitrate=6.3	Note To view the default FMTP values associated with variants, use the show sbc sbe codecs variant command.
Step 11	exit	Exits the codec variant mode and enters into the SBE entity mode.
	Example: Router(config-sbc-sbe-codec-var-codec)# exit	

	Command or Action	Purpose
Step 12	codec variant profile profile-name	Enters the codec variant profile mode to configure a codec variant profile.
	Example: Router(config-sbc-sbe)# codec variant profile Profile-1	• <i>profile-name</i> —The codec profile name.
Step 13	variant variant-name	Defines the codec variant name.
	Example: Router(config-sbc-sbe-codec-var-prof)# variant G723-H-1	 <i>variant-name</i>—The codec variant name. Note '#' is reserved for base variants. Therefore, the variant name cannot start with '#'
Step 14	end	Exits the codec variant profile mode and enters in the privileged EXEC mode.
	Example: Router(config-sbc-sbe-codec-var-prof)# end	
Step 15	<pre>show sbc service-name sbe codecs variant [profile]</pre>	 Displays information about codec variants on the SBC. profiles—Displays information about codec variant profiles.
	Example: Router# show sbc mySBC sbe codecs variant	Ĩ

Configuring Codec on CAC Policy Set

This task shows how to enable codec convert and configure codec variant profile on a CAC policy set. When the codec variant conversion is enabled or disabled, the following events occur:

- If the codec variant conversion is disabled, the SBC does not take into account the specified variant profile. All the codecs that have been passed are left in their original representation, and any new codecs added by the SBC are added with their standard representation.
- If the codec variant conversion is enabled but the variant profile is not configured, all the codecs are converted to its standard representation.
- If the codec variant is enabled and the variant profile is configured, any codecs matched by the profile are converted to the appropriate variant representation, and any codecs that is not matched by the variant profile is converted to its standard representation.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name

- 6. table-type {policy-set | limit {list of limit tables}}
- 7. entry entry-id
- 8. caller codec convert
- 9. callee codec convert
- **10.** caller codec profile profile-name
- **11.** callee codec profile profile-name
- 12. exit
- 13. exit
- 14. complete

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the SBC service mode.
		Use the service-name argument to define the name of the
	Example: Router(config)# sbc mysbc	service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table StandardListByAccount	the context of an SBE policy set.
Step 6	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	

	Command or Action	Purpose
Step 7	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 8	caller codec convert	To enable or disable codec variant conversion at the caller side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller codec convert</pre>	
Step 9	callee codec convert	To enable or disable codec variant conversion at the callee side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee codec convert</pre>	
Step 10	caller codec profile profile-name	To specify a codec variant profile at the caller side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller codec profile</pre>	• <i>profile-name</i> —The codec variant profile name.
Step 11	callee codec profile profile-name	To specify a codec variant profile at the callee side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee codec profile</pre>	• <i>profile-name</i> —The codec variant profile name.
Step 12	exit	Exits from the entry mode and enters into the cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 13	exit	Exits from the cactable mode and enters into the cacpolicy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 14	complete	Completes the CAC policy set when you have committed the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	

Configuration Example for Per-Adjacency Codec String Interworking

The following example shows how to configure the Per-Adjacency Codec String Interworking feature when caller supports G723 bitrate 6.3 annexa=no codec and callee supports G.723.1 codec variant:

```
configure terminal
sbc MySBC
sbe
codec variant codec PCMU.1
 standard PCMU
 variant PCMU.1
 exit
codec variant profile pcmu-var
  variant PCMU.1
  variant #G.723.1/H
  exit
  cac-policy-set 2
   first-cac-table codec-convert
   first-cac-scope src-adjacency
   cac-table codec-convert
    table-type limit src-adjacency
     entry 1
     match-value CallMgrA
      callee codec profile pcmu-var
      callee codec convert
     media police strip
      action cac-complete
   complete
    end
```



SDP Handling

Cisco Unified Border Element (SP Edition) by default passes through all a= lines in SIP messages containing SDP offers and answers that it forwards. You can also configure Cisco Unified Border Element (SP Edition) to block certain a= lines, either by specifying a whitelist (a finite set of a=lines that are passed through, with all others blocked), or alternatively a blacklist (a finite set of a=lines that are blocked, with all others passed through). Additionally, user exits in the Cisco Unified Border Element (SP Edition) code base allow customers to write their own code to insert and/or strip one or more media-level a= lines when processing an offer on an answer.

The SIP-I Support feature enables Cisco Unified Border Element (SP Edition) to pass through the ISDN User Part (ISUP) parameters in Session Initiation Protocol (SIP) messages that are added by a SIP or Public Switched Telephone Network (PSTN) interworking gateway.

The SIP Non-SDP Body Filtering feature adds support for Cisco Unified Border Element (SP Edition) to process non-SDP bodies, and in particular the ISUP body using SIP-I.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 2.4	The SIP SDP Attribute Passthrough feature was introduced on the Cisco IOS XR.
Cisco IOS XE Release 2.6	The SIP-I Support and SIP Non-SDP Body Filtering features were introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.1S	Repeat SDP on 200 Invite Response feature was added.

Feature History for SDP Handling

Contents

This chapter contains the following sections:

- Configuring SIP SDP Attribute Passthrough, page 19-2
- SIP-I Support and SIP Non-SDP Body Filtering, page 19-17

Configuring SIP SDP Attribute Passthrough

This section contains the following subsections:

- Restrictions for Configuring SIP SDP Attribute Passthrough, page 19-2
- Information about SIP SDP Attribute Passthrough, page 19-3
- Information About Repeat SDP on 200 Invite Response, page 19-3
- Configuring SIP SDP Attribute Passthrough, page 19-6
- Configuring Repeat SDP on 200 INVITE Response, page 19-14
- Example of SIP SDP Attribute Passthrough, page 19-15
- Example of Repeat SDP on 200 INVITE Response Configuration, page 19-16

Restrictions for Configuring SIP SDP Attribute Passthrough

Review the following restrictions for SIP SDP Attribute Passthrough:

- The existing reflect behavior is not supported.
- Wildcard or prefix matching of attribute lines is not supported.
- Distinguishing media-level from session-level a-lines for the purposes of matching is not supported.
- Sophisticated matching conditions (for example, apply only to video streams or apply only to offers) are not supported.
- Attribute blocking in media bypass calls is not supported.
- Blocking function is restricted to unknown attributes.
- The following attributes are ignored by unknown attribute policy because this may interfere with the correct operation of the SBC.
 - a=rtpmap
 - a=fmtp
 - a=sendonly
 - a=recvonly
 - a=inactive
 - a=sendrecv
 - a=ptime
 - a=mid
 - a=group
 - a=curr

- a=des
- a=conf
- a=crypto.

At the point where the policy is applied, a (rate-limited) warning log is issued if the policy attempts to delete one of these lines.

Information about SIP SDP Attribute Passthrough

Additional per-call storage is needed to store the SDP policy that is being applied. This is expected to be ~160 bytes per call.

Information About Repeat SDP on 200 Invite Response

To support interoperation with endpoints that may require an agreed Session Description Protocol (SDP) to be resent for 200 INVITE responses, the user can configure SBC to repeat an agreed SDP, in a 200 INVITE response, when needed, after the successful provisioning of an offer-answer exchange.

This option is configured in the CAC policy for SIP calls. The default is off.

The agreed SDP answer is the SDP answer from the latest completed SDP offer/answer exchange procedure.

When Repeat SDP on 200 Invite Response is configured, the call flow is as shown in the following three figures.

Figure 19-1 shows the call flow for an SDP on the second reliable response.



Figure 19-1 Call Flow for SDP on Second Reliable Response

Figure 19-2 shows the call flow for an SDP on the final response.



Figure 19-2Call Flow for SDP on Final Response

Figure 19-3 shows the call flow for no SDP on the final response.

Figure 19-3 Call Flow for No SDP on Final Response



See the ?\$paranum>Configuring Repeat SDP on 200 INVITE Response? section on page 19-14 for the configuration procedure.

See the ?\$paranum>Example of Repeat SDP on 200 INVITE Response Configuration? section on page 19-16 for and example configuration.

Configuring SIP SDP Attribute Passthrough

This section contains the steps for implementing SIP SDP Attribute Passthrough.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. sip sdp-match-table table-name1
- 5. action whitelist | blacklist
- 6. match-string attribute-name1
- 7. match-string attribute-name2
- 8. exit
- 9. sip sdp-match-table table-name2
- 10. action whitelist | blacklist
- **11.** match-string *attribute-name1*
- 12. match-string attribute-name3
- 13. exit
- 14. sip sdp-policy-table table-name1
- **15.** match-table table-name 1
- 16. exit
- 17. sip sdp-policy-table table-name2
- **18.** match-table table-name2
- 19. exit
- **20**. **cac-policy-set** *number*
- 21. first-cac-table table-name
- **22**. **first-cac-scope** *scope*
- 23. cac-table table-name
- 24. table-type {policy-set | limit {list of limit tables}}
- 25. entry number
- 26. match-value value
- **27.** action action-name
- **28.** caller-inbound-policy policytab-name
- **29.** caller-outbound-policy policytab-name
- **30.** callee-inbound-policy policytab-name
- 31. callee-outbound-policy policytab-name
- 32. exit
- 33. exit
- 34. complete
- 35. exit
- 36. active-cac-policy-set number

- 37. end
- **38.** show sbc service-name sbe cac-policy-set number table number entry number

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	sip sdp-match-table <i>table-name</i>	Adds an existing sdp-match-table into policy.
	Example: Router(config-sbc-sbe)# sip sdp-match-table 1	
Step 5	action whitelist/blacklist	Specifies an SDP policy table action.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# action blacklist	
Step 6	match-string attribute-name1	Configures an SDP attribute matching string.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn1	
Step 7	match-string attribute-name1	Configures an SDP attribute matching string.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn2	
Step 8	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# exit	
Step 9	sip sdp-match-table <i>table-name</i>	Adds an existing sdp-match-table into policy.
	Example: Router(config-sbc-sbe)# sip sdp-match-table 2	

	Command or Action	Purpose
Step 10	action whitelist/blacklist	Adds an action allowing a defined set of attributes and blocking the remaining attributes.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# action blacklist	
Step 11	match-string attribute-name1	Configures an SDP attribute matching string.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn1	
Step 12	match-string attribute-name1	Configures an SDP attribute matching string.
Step 13	Example: Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn2 exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-sdp-match-tbl)# exit	
Step 14	sip sdp-policy-table table-name	Configures an SDP policy table.
	Example: Router(config-sbc-sbe-sip)# sip sdp-policy-table foo	
Step 15	match-table table-name	Configure an SDP match table used in a policy.
	Example: Router(config-sbc-sbe-sdp-policy-tbl)# match-table matchtab2	
Step 16	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-sip-adj)# exit	
Step 17	sip sdp-policy-table table-name	Configures an SDP policy table.
	Example: Router(config-sbc-sbe)# sip sdp-policy-table foo2	
Step 18	<pre>match-table table-name</pre>	Configure an SDP match table used in a policy.
	Example: Router(config-sbc-sbe-sdp-policy-tbl)# match-table matchtab3	
Step 19	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-sdp-policy-tbl)# exit	

	Command or Action	Purpose
Step 20	cac-policy-set number	Enters the submode of CAC policy set configuration.
	Example:	
	Router(config-sbc-sbe)# cac-policy-set 1	
Step 21	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# first-cac-table RootCacTable	
Step 22	first-cac-scope scope	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# first-cac-scope src-adjacency	
Step 23	cac-table table-name	Creates or configures an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# cac-table RootCacTable	

	Command or Action	Purpose
Step 24	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
		list of limit tables can be one of the following values:
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit call-priority	• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
Step 25	entry number	Creates or modifies an entry in a table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 26	match-value key	Configures the match-value of an entry in a CAC Limit table. It is only relevant for Limit table types.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value immediate	The <i>key</i> argument is a string or a keyword based on the table type. The format of the key is determined by the Limit table type (for example, Limit event-type tables or Limit call-priority tables).
		For Limit event-type tables (table-type limit <i>event-type</i>), the match value string options are the following:
		• <i>call-update</i> —Compare the beginning of the calling number string.
		• <i>endpoint-reg</i> —Compare the name of the destination adjacency.
		• <i>new-call</i> —Compare the beginning of the dialed digit string.
		For Limit call-priority tables (table-type limit <i>call-priority</i>), the match value string options are the following:
		• <i>critical</i> —Match calls with resource priority 'critical'.
		• <i>flash</i> —Match calls with resource priority 'flash'.
		• <i>flash-override</i> —Match calls with resource priority 'flash-override'.
		• <i>immediate</i> —Match calls with resource priority 'immediate'.
		• <i>priority</i> —Match calls with resource priority 'priority'.
		• <i>routine</i> —Match calls with resource priority 'routine'.
		For all other Limit tables, enter a name or digit string
		WORD—Name or digit string to match. (Max Size 255).
Step 27	action action-name	Specifies the action to take if this entry is chosen.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete	
Step 28	caller-inbound-policy policytab-name	Configures a caller inbound SDP policy table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-inbound-policy policytab1	

	Command or Action	Purpose
Step 29	caller-outbound-policy policytab-name	Configures a caller outbound SDP policy table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-outbound-policy policytab1	
Step 30	callee-inbound-policy policytab-name	Configures a callee inbound SDP policy table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-inbound-policy policytab2	
Step 31	callee-outbound-policy policytab-name	Configures a callee outbound SDP policy table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-outbound-policy policytab2	
Step 32	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit	
Step 33	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 34	complete	Performs a consistency check on the CAC policy set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# complete</pre>	
Step 35	exit	Returns to the previous submode.
	Example: Router(config-sbc-sbe-cacpolicy)# exit	
Step 36	active-cac-policy-set number	Enters the active CAC policy set.
	Example: Router(config-sbc-sbe)# active-cac-policy-set 1	
Step 37	end	Exits SBE mode and enters Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 38	<pre>show sbc service-name sbe cac-policy-set number table number entry number</pre>	Displays detailed information for a given entry in a CAC policy table.
	Example: Router# do show sbc interwork sbe cac-policy-set 1 table 1 entry 1	

Configuring Repeat SDP on 200 INVITE Response

Use the following procedure to configure SBC to send a repeat SDP on 200 INVITE responses.

SUMMARY STEPS

- 1. config
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type policy-set
- 7. entry entry-id
- 8. sdp repeat answer
- 9. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	config	Enters global configuration mode.
	Example: Router# config	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.
	Command or Action	Purpose
--------	--	---
Step 6	table-type policy-set	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	
Step 7	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 8	sdp repeat answer Example:	Configures SBC to repeat an agreed Session Description Protocol (SDP), in a 200 INVITE response, after the successful provisioning of an offer-answer exchange when
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # sdp repeat answer	needed.
Step 9	end	Exits configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Example of SIP SDP Attribute Passthrough

This section provides a sample configuration and output for SIP SDP Attribute Passthrough.

```
Note
```

The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

```
Router# config t
Enter configuration commands, one per line. End with CNTL/Z.
Router(config) # sbc interwork
Router(config-sbc)# sbe
Router(config-sbc-sbe) # sip sdp-match-table matchtab1
Router(config-sbc-sbe-sdp-match-tbl)# action blacklist
Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn
Router(config-sbc-sbe-sdp-match-tbl)# match-string X-cap
Router(config-sbc-sbe-sdp-match-tbl)# exit
Router(config-sbc-sbe) # sip sdp-match-table matchtab2
Router(config-sbc-sbe-sdp-match-tbl)# action blacklist
Router(config-sbc-sbe-sdp-match-tbl)# match-string X-sqn
Router(config-sbc-sbe-sdp-match-tbl)# match-string X-pc-csuites-rtp
Router(config-sbc-sbe-sdp-match-tbl)# exit
Router(config-sbc-sbe)# sdp-policy-table policytab1
Router(config-sbc-sbe-sdp-policy-tbl) # match-table matchtab1
Router(config-sbc-sbe-sdp-policy-tbl)# exit
Router(config-sbc-sbe) # sip sdp-policy-table policytab2
Router(config-sbc-sbe-sdp-policy-tbl)# match-table matchtab2
Router(config-sbc-sbe-sdp-policy-tbl)# exit
Router(config-sbc-sbe)# cac-policy-set 1
```

```
Router(config-sbc-sbe-cacpolicy)# first-cac-table 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router(config-sbc-sbe-cacpolicy)# cac-table 1
Router (config-sbc-sbe-cacpolicy-cactable) # table-type limit src-adjacency
Router(config-sbc-sbe-cacpolicy-cactable) # entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value sipp1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router (config-sbc-sbe-cacpolicy-cactable-entry) # caller-inbound-policy policytab1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-outbound-policy policytab1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-inbound-policy policytab2
Router (config-sbc-sbe-cacpolicy-cactable-entry) # callee-outbound-policy policytab2
Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit
Router(config-sbc-sbe-cacpolicy-cactable) # exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy) # exit
Router(config-sbc-sbe)# active-cac-policy-set 1
```

This section provides a sample configuration and output for SIP SDP Attribute Passthrough.

Router(config-sbc-sbe)# do show sbc interwork sbe cac-policy-set 1 table 1 entry 1 SBC Service "interwork"

```
Policy set 1 table 1 entry 1
 Match value
                         sipp1
 Action
                         CAC policy complete
 Max calls
                         Unlimited
 Max call rate
                        Unlimited
 Max in-call rate
                        Unlimited
 Max out-call rate
                        Unlimited
 Max registrations
                        Unlimited
 Max reg. rate
                         Unlimited
 Max bandwidth
                         Unlimited
 Max channels
                         Unlimited
                         Allowed
 Transcoder
 Caller privacy setting Never hide
 Callee privacy setting Never hide
 Early media
                         Allowed
 Early media direction Both
 Early media timeout
                        0
 Restrict codecs to list default
 Restrict caller codecs to list default
                                default
 Restrict callee codecs to list
 Media bypass
                         Allowed
 SRTP Transport
                         Not Set
 Callee hold setting
                         Standard
 Caller hold setting
                        Standard
 Number of calls rejected by this entry
 Caller inbound SDP policy
                                        policytab1
 Caller outbound SDP policy
                                       policytab1
 Callee inbound SDP policy
                                       policytab2
 Callee outbound SDP policy
                                        policytab2
```

Example of Repeat SDP on 200 INVITE Response Configuration

The following example shows how to configure SBC to send a repeat SDP on 200 INVITE responses.

```
Router# config t
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table cac-tbl-1
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
```

Router(config-sbc-sbe-cacpolicy-cactable-entry)# sdp repeat answer

SIP-I Support and SIP Non-SDP Body Filtering

This section contains the following subsections:

- Prerequisites, page 19-17
- Restrictions for SIP Non-SDP Body Filtering and SIP-I Support, page 19-17
- Information about SIP Non-SDP Body Filtering and SIP-I Support, page 19-18
- Configuring SIP SDP Attribute Passthrough, page 19-6
- Examples—SIP Non-SDP Body Filtering and SIP-I Support, page 19-22

Prerequisites

The following prerequisite is required to implement SIP Non-SDP Body Filtering and SIP-I Support:

Before implementing SIP Non-SDP Body Filtering and SIP-I Support, Cisco Unified Border Element (SP Edition) must already be configured. See the procedures described in Chapter 3, ?\$paratext>.?

Restrictions for SIP Non-SDP Body Filtering and SIP-I Support

The following restrictions and limitations apply to SIP Non-SDP Body Filtering and SIP-I Support:

- If dual tone multifrequency (DTMF) interworking is enabled for a call, the INFO messages containing a DTMF digit may not pass through.
- The SBC does not support Secure Multipurpose Internet Mail Extensions (S/MIME) encryption or decryption. While the SBC may allow encrypted bodies to pass through, it does not modify them.
- In compliance with Section 8.2.1.1 of RFC 3398, the SBC does not support a From header without a username.
- The total size of the MIME bodies and associated header allowed to pass through is limited to approximately 1000 bytes. The final size allowed depends on the structure of the headers and MIME bodies and should not exceed 2000 bytes.
- The SBC may not preserve the original order of MIME bodies and may insert the SDP as the first body part.
- This feature does not work in conjunction with H.323.
- Since the SBC considers BYE requests on a hop-by-hop basis, it does not pass any information using a BYE response it received.
- The SBC allows the user=phone URI parameter on the Request-URI to pass through.
- The SBC may alter the MIME boundary of a message.

Information about SIP Non-SDP Body Filtering and SIP-I Support

The following sections provide information about the SIP Non-SDP Body Filtering feature and the SIP-I Support feature.

SIP Non-SDP Body Filtering

The SIP Non-SDP Body Filtering feature adds support for Cisco Unified Border Element (SP Edition) to process non-SDP bodies, and in particular the ISUP body using SIP-I. The SBC can pass through, strip out, or reject non-SDP bodies. The message body of a SIP message is described using header fields such as Content-Disposition, Content-Encoding, and Content-Type, which provide information about the body. SBC uses a body profile that you create and associate to filter non-SDP bodies from incoming and outgoing SIP messages, based on the Content-Type header field. A body profile allows a message containing a specific non-SDP body to take one of the following actions:

- To be passed (without altering the message)
- To be stripped of the body (and pass the rest of the message)
- To be rejected
- SBC uses the 'handling' parameter in the message to decide whether to strip the body or reject the message.

Like any other SIP profile, such as a method profile, you need to first create a body profile. Then you can associate the body profile to cause the body profile to take action on incoming and outgoing SIP messages that fall under the SBE mode, or adjacency mode, or method profile mode.

You can create a body profile:

• Using the **sip body-profile** {*profile_name*} command under the SBE mode.

The **body** command and **action** (**body**) command are used in conjunction with the **sip body-profile** command. The **body** command names a body type or content header type for a non-SDP message body. The **action** (**body**) command sets the action to take on a body type in a SIP body profile.

After creating a body profile, you can associate the body profile at the following levels and configuration modes:

- At the SIP signaling entity level (ingress or egress), under SBE mode, using the **sip default body-profile** [[**inbound** | **outbound**] {*profile_name* }] command. The body profile is associated for the entire signaling instance (that is all messages, either ingress or egress, passing through the SBC.)
- At the SIP adjacency level, under SIP adjacency mode, using the **body-profile** [[**inbound** | **outbound**] {*profile_name*}] command. The body profile is associated to an adjacency.
- At the SIP method profile level, under method profile mode, using the **body-profile** {*profile_name*} command. The body profile is associated to a method profile.

SIP-I Support

The SIP-I Support feature enables Cisco Unified Border Element (SP Edition) to pass through the ISDN User Part (ISUP) parameters in Session Initiation Protocol (SIP) messages that are added by a SIP or Public Switched Telephone Network (PSTN) interworking gateway. ISUP is a call control protocol used in SS7 networks primarily for setting up and tearing down telephone calls and for maintenance of the network.

SIP-I is an approach defined by ITU-T Q.1912.5. SIP-I provides an approach for interworking SIP networks and the traditional, circuit-based ISDN User Part (ISUP) networks. SIP-I provides a method for passing through ISUP-specific header parameters through a SIP network so that calls that originate and terminate on the circuit-based ISUP network can cross a SIP network with no loss of information.

SIP-I allows transparent passthrough of ISUP parameters through a SIP network by attaching a copy of the ISUP message to the SIP message at the incoming PSTN gateway. The ISUP message appears as a non-SDP message body on the SIP message. SIP-I has a mechanism to indicate the presence of ISUP (based on Content-Type header) and if the ISUP is mandatory or can be passed through, depending on the Content-Disposition header. The SBC passes through the ISUP message body without coding or decoding the ISUP message.

The mapping between SIP and ISUP protocols is carried out by the Media Gateway Controller (MGC). In the SBC, the ISUP parameters can be carried in the SIP Request-Uniform Resource Identifier (URI) or the SIP message body.

Cisco Unified Border Element (SP Edition) supports the following SIP-I and profile functions:

- Application or SDP is processed on INVITE, UPDATE, and PRACK requests and their responses.
- Application or DTMF-info is processed on INFO to allow DTMF tones to pass through.
- The NOTIFY messages on message or SIP-frag is analyzed to find out whether it indicates that a subscription or refer dialog is to be terminated.

Non-SDP Message Body Example

The following is an example of a non-SDP message body; the SIP message body is not shown in detail for brevity's sake. The non-SDP body present in the example is of type "application/resource-lists+xml":

```
INVITE sip:conf-fact@example.com SIP/2.0
      Content-Type: multipart/mixed; boundary="boundary1"
     Content-Length: 617
      --boundarv1
     Content-Type: application/sdp
     v = 0
     o=alice 2890844526 2890842807 IN IP4 atlanta.example.com
     s=-
     c=IN IP4 192.0.2.1
     t=0 0
     m=audio 20000 RTP/AVP 0
     a=rtpmap:0 PCMU/8000
     m=video 20002 RTP/AVP 31
     a=rtpmap:31 H261/90000
      --boundarv1
     Content-Type: application/resource-lists+xml
     Content-Disposition: recipient-list
      <?xml version="1.0" encoding="UTF-8"?>
      <resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"
        <list>
          <entry uri="sip:bill@example.com"/>
          <entry uri="sip:randy@example.net"/>
          <entry uri="sip:joe@example.org"/>
        </list>
      </resource-lists>
      --boundary1--
```

Implementing SIP Non-SDP Body Filtering

The follow steps describe a sample configuration where a body profile is created with a particular body type and action to take on that body type and then the body profile is associated at the SIP signaling level.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. **sip body-profile** {*profile_name*}
- **5. body** {*WORD*}
- 6. action [pass | nopass | strip | reject]
- 7. exit
- 8. exit
- 9. sip default body-profile [[inbound | outbound] {profile_name}

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip body-profile {profile_name}</pre>	Creates a body profile to filter non-SDP bodies from incoming and outgoing SIP messages. Enters SBE SIP
	Example: Router(config-sbc-sbe)# sip body-profile bodyprofile1	Body configuration mode.

	Command or Action	Purpose
Step 5	<pre>body {WORD} Example: Router(config-sbc-sbe-sip-body)# body</pre>	This command describes the body type or content header type for SBC to act on messages with the specified body type or content header type. Enters SBE SIP Body Element configuration mode.
	application/ISUP	The body name must be in the form of <media-type>/<media-sub-type>, for example, application/ISUP. The body name field is case-insensitive.</media-sub-type></media-type>
		Asterisk (*) is used to match <i>all</i> non-SDP body types. Note that * is also interpreted as a string by the CLI, and is just a token used to indicate wild-card match.
		The following Content-Type descriptions are not allowed: application/sdp and multipart/mixed
		In the example, the command describes the body type that is to act on messages with Content-Type header "application/ISUP."
Step 6	action [pass nopass strip reject]	Sets the action to take on a body type in a SIP body profile for a non-SDP message body.
	<pre>Example: Router(config-sbc-sbe-sip-body-ele)# action strip</pre>	• pass —Instructs the SBC to pass through the body type of the non-SDP message body.
		• nopass —Uses the handling parameter in the message to determine whether to strip the body or reject the entire message with error code 415 (Unsupported media type).
		• strip —Strips the body and passes the rest of the message.
		• reject —Rejects the entire message with an error code.
Step 7	exit	Exits SBE SIP Body Element configuration mode.
	Example: Router(config-sbc-sbe-sip-body-ele)# exit	
Step 8	exit	Exits SBE SIP Body configuration and enters SBE configuration mode.
	Example: Router(config-sbc-sbe-sip-body)# exit	
Step 9	<pre>sip default body-profile [[inbound outbound] {profile_name}</pre>	Associates the body profile at the SIP signaling level, for the entire signaling instance (that is all messages, either ingress or egress, passing through the SBC).
	Example: Router(config-sbc-sbe)# sip default body-profile inbound bodyprofile1	

Examples—SIP Non-SDP Body Filtering and SIP-I Support

The following is a configuration example of SIP Non-SDP Body Filtering:

```
sbc foo
    sbe
        sip body-profile profile1
            body application/ISUP
                    action strip
            body application/QSIG
                    action reject
                    hunt-on-reject
            body *
                    action reject
        sip body-profile profile2
            description test-profile
            body application/ISUP
                 action nopass
            body application/QSIG
                 action pass
      sip body-profile profile3
            body application/ISUP
                 action nopass
            body application/QSIG
                 action pass
        sip default body-profile inbound profile1
        sip default body-profile outbound profile2
        sip method-profile default
                                      !- pre-provisioned
                                       ! "default" method profile
                                       !- used at sbe level
            method INVITE
              body-profile profile1
       sip method-profile mp1
                                       !-create a new method
                                       ! profile used in adj
            method INVITE
              body-profile profile2
       sip method-profile mp2
                                       !-create a new method
                                       ! profile used in adj
            method REGISTER
              body-profile profile1
        adjacency sip adj-1
            body-profile inbound profile2
            body-profile outbound profile1
```

The following example displays all the non-SDP message body profiles in use:

Router# show sbc mySBC sbe sip body-profile

Name	In Use
profile1	Yes
profile2	Yes
profile3	No

The following example displays the details of the specified non-SDP message body profile named "profile2":

Router# show sbc mySBC sbe sip body-profile profile2

Name Description	:	profile2 test-profile
Element Action	:	application/ISUP nopass
Hunt-on-reject:		false
Element Action Hunt-on-reject	: :	application/QSIG pass false



Flexible Media Routing

The Flexible Media Routing feature supports the call legs in which media and signaling are sent over different virtual or physical networks. The signaling network is configured using the **vrf** command in the adjacency submode. All the call legs to and from an adjacency use the same VPN ID for signaling. The media network is configured using the *vrf* parameter in the **media-address** command.

When the Flexible Media Routing feature is enabled, the media address selection overrides the VPN ID-based selection. Therefore, the media VPN ID is no longer compared with the signaling VPN ID. Instead, the SBC selects the media address whose realm matches the adjacency realm. The IPv6 and H.323 protocols support the Flexible Media Routing feature.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to as the session border controller (SBC) in this document.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Flexible Media Routing

Release	Modification
Cisco IOS XE Release 3.5S	The Flexible Media Routing feature was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.

Contents

This chapter contains the following sections:

- Prerequisites for Configuring the Flexible Media Routing Feature, page 20-2
- Configuring the Flexible Media Routing Feature, page 20-2
- Configuration Examples for the Flexible Media Routing Feature, page 20-3
- Changes in XML Billing Records, page 20-4

Prerequisites for Configuring the Flexible Media Routing Feature

The following prerequisite is required to configure the Flexible Media Routing feature: Ensure that the SBC is deactivated before configuring the Flexible Media Routing feature.

Configuring the Flexible Media Routing Feature

This task shows how to configure the Flexible Media Routing feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. no activate
- 4. allow diff-med-sig-vpn
- 5. activate
- 6. exit

DETAILED STEPS

	Command or Action	Purpose	
Step 1	configure terminal	Enables the global configuration mode.	
	Example: Router# configure terminal		
Step 2	sbc sbc-name	Enters the SBC configuration mode.	
	Example: Router(config)# sbc test		
Step 3	no activate	Deactivates the SBC.	
	Example: Router(config-sbc)# no activate	Note If the SBC is active and you run the allow diff-med-sig-vpn command, the system issues a warning message, asking you to first deactivate the SBC. You can reactivate the SBC using the activate command.	
Step 4	allow diff-med-sig-vpn Example:	Allows media and signaling to use different VPN IDs in a call leg. The no version of this command allows media and signaling to use the same VPN ID in a call leg.	
	Router(config-sbc)# allow diff-med-sig-vpn		

	Command or Action	Purpose
Step 5	activate	Reactivates the SBC.
	Example: Router(config-sbc)# activate	
Step 6	exit	Exits the SBC configuration mode.
	Example: Router(config-sbc)# exit	

Configuration Examples for the Flexible Media Routing Feature

The following example shows the SBC behavior when the Flexible Media Routing feature is configured:

```
sbc test
allow diff-med-sig-vpn
```

The following example shows how to configure different VRFs under the signaling and media networks.

```
sbe
   adjacency sip sipp1 ========= incoming
       force-signaling-peer all
       vrf vrf_sipp1
       nat force-on
       inherit profile preset-access
       signaling-address ipv4 192.0.2.1
       statistics method summary
       signaling-port 5060
       remote-address ipv4 192.0.2.3 255.255.255.0
       signaling-peer 192.0.2.3
       realm FMR
       attach
   adjacency sip sipp2 ============== outgoing
       force-signaling-peer all
       vrf vrf_sipp2
       nat force-off
       inherit profile preset-access
       signaling-address ipv4 192.0.2.2
       statistics method summary
       signaling-port 5060
       remote-address ipv4 192.0.2.4 255.255.255.0
       signaling-peer 192.0.2.4
       fast-register disable
       realm FMR
   call-policy-set 1
       first-call-routing-table start-table1
       first-reg-routing-table start-table1
       rtg-src-adjacency-table start-table1
       entry 1
       match-adjacency sipp1
       dst-adjacency sipp2
       action complete
       entry 2
       match-adjacency sipp2
       dst-adjacency sipp1
       action complete
   complete
   call-policy-set default 1
```

Γ

```
network-id 9737
!
media-address ipv4 192.0.2.5 vrf vrf_media realm FMR
port-range 16384 32767 any
activate
!
end
```

Changes in XML Billing Records

After the Flexible Media Routing feature is enabled, the SBC adds the **mediarealm** attribute to the adjacency element in the XML billing records as follows:

<adjacency type="orig" name="Adj1" account="Acc1" vpn="0X12345678" mediarealm="Internet"/>

For more information about the mediarealm attribute, see Appendix C, XML Billing Schema.



Inherit Profiles for Non-IMS Adjacencies

Cisco Unified Border Element (SP Edition) supports Inherit Profiles for adjacencies that are not part of an IP Multimedia Subsystem (IMS) network. This feature allows Cisco Unified Border Element (SP Edition) to operate in non-IMS networks using any of three non-IMS profiles that define an adjacency as Access, Core, or Peering. Cisco Unified Border Element (SP Edition) uses this definition to process packets properly and add the correct information in the outgoing packets.

By configuring each of these different types of adjacency with a profile, you can make efficiency and occupancy gains. For example, Cisco Unified Border Element (SP Edition) does not store registration information from messages received from Peering adjacencies. When a subscriber successfully registers from an Access adjacency, Cisco Unified Border Element (SP Edition) remembers the subscriber's registration details for later use and only stores this information on Access adjacencies.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Feature History for Inherit Profiles for Non-IMS Adjacencies

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR.

Contents

This module contains the following sections:

- Prerequisites, page 21-2
- Information About Inherit Profiles for Non-IMS Adjacencies, page 21-2
- CLI Support for Inherit Profiles for Non-IMS Adjacencies, page 21-3
- Configuration Examples, page 21-4

Prerequisites

The following prerequisites are required to implement Inherit Profiles for Non-IMS Adjacencies:

• Before implementing this feature, the SBC must already be configured. See the procedures described in ?\$paratext[CT_ChapTitle]>?

Information About Inherit Profiles for Non-IMS Adjacencies

Cisco Unified Border Element (SP Edition) can be deployed in various network topologies and plays different roles depending on its location in the network. Each of the deployed roles usually has a specific set of requirements associated with it. These requirements control which headers need to be added, checked, updated, or removed, and which headers, methods, and options are permitted to be passed through.

Cisco Unified Border Element (SP Edition) can be deployed in non-IMS networks and thus takes on different roles in non-IMS networks. For example, Cisco Unified Border Element (SP Edition) can face a registrar network or end user client devices that will attempt to register through the SBC. Alternatively, you can position it on the Network-Network Interface (NNI).

To deploy in non-IMS networks, Cisco Unified Border Element (SP Edition) uses easily-configured inherit profiles that comprise a collection of related configuration appropriate to a particular network role. Inherit profiles may be configured for an application on a per-adjacency basis or at a global level as a default.

Non-IMS Inherit Profiles Types and Behaviors

The following are the non-IMS inherit profiles that can be configured for an adjacency:

- preset-access profile—configures an Access adjacency. The Access adjacency is not part of an IMS network. This adjacency faces user equipment, such as a subscriber's telephone or other SIP device, that attempts to register through the SBC.
- preset-core profile—configures a Core adjacency. This is the default profile. The Core adjacency is not part of an IMS network. This adjacency faces a registrar network and links to the registrar.
- preset-peering profile—configures a Peering adjacency. The Peering adjacency is not part of an IMS network. This adjacency, for example, sitting at the Network-Network Interface, links one registrar to another. The SBC is not required to store subscriber information from messages received from peering adjacencies.

The following are examples of behaviors that are affected by the non-IMS inherit profiles:

- Whether various headers (such as P Charging Vector) are created.
- Which headers, methods, and options are passed through and which are stripped out.
- Whether inbound and outbound calls to a subscriber can be made before that subscriber is registered.
- Whether the SBC rewrites the contact headers during the registration process.

When you configure the SBC with a certain non-IMS profile, calls may be handled differently. For example, when a call is received on a Core adjacency, the SBC checks to see if the endpoint is registered. If the subscriber is registered and is known to be behind a Network Address Translation (NAT), the SBC configures the call to traverse the NAT. If the endpoint is not registered, the SBC applies a routing policy and routes the call to the appropriate adjacency.

Effect of Non-IMS Inherit Profiles on Method Profiles, Header Profiles, and Option Profiles

Use of a non-IMS inherit profile dynamically assigns the following sets of profiles (method profile, header profile, and option profile) to a call based on the non-IMS inherit-profile selected. Table 21-1 shows which non-IMS inherit profile has an effect on which specific method profile, header profile, and option profile.

The effect is not visible in the adjacency configuration for header-profile, method-profile or option profiles, and can be overridden by explicit configuration of header, method, option profiles as needed.

Non-IMS Inherit Profile	Method Profile	Header Profile	Option Profile
preset-access	preset-std-in-mth	preset-std-in-hdr	preset-std-in-opt
	preset-std-out-mth	preset-std-out-hdr	preset-std-out-opt
	Type: Whitelist	Type: Whitelist	Type: Whitelist
	Actions: Passes INFO, Passes UPDATE	Actions: Passes Server, Passes Diversion, Passes Resource-Priority	Actions: Passes Replaces (only)
preset-core	preset-std-in-mth	preset-std-in-hdr	preset-std-in-opt
	preset-std-out-mth	preset-std-out-hdr	preset-std-out-opt
	Type: Whitelist	Actions: Passes Server,	Type: Whitelist
	Actions: Passes INFO, Passes UPDATE	Passes Diversion, Passes Resource-Priority	Actions: Passes Replaces (only)
preset-peering	preset-std-in-mth	preset-std-in-hdr	preset-std-in-opt
	preset-std-out-mth	preset-std-out-hdr	preset-std-out-opt
	Type: Whitelist	Actions: Passes Server,	Type: Whitelist
	Actions: Passes INFO, Passes UPDATE	Passes Diversion, Passes Resource-Priority	Actions: Passes Replaces (only)

Table 21-1 Effect of Non-IMS Inherit Profiles on Method, Header and Option Profiles

CLI Support for Inherit Profiles for Non-IMS Adjacencies

The **inherit profile** command has the following three keywords that allow you to configure a preset-access, preset-core, or preset-peer profile for an adjacency that is not part of an IMS network:

preset-access—Specifies a preset access profile for an adjacency that faces an access device on a User-Network Interface (UNI) location.

preset-core—Specifies a preset core profile for an adjacency that faces a core device on a UNI location. This is the default.

preset-peering—Specifies a preset peering profile for an adjacency that faces a peer device on a Network-Network Interface (NNI) location.

The adjacency-specific command configuration overrides any global configuration of the adjacency that was configured using the **sip inherit profile** command.

The following example shows all the profiles available with the **inherit profile** command:

```
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip sipa
Router(config-sbc-sbe-adj-sip)# inherit profile ?
preset-access preset-access profile
preset-core preset-core profile
preset-ibcf-ext-untrusted preset-ibcf-ext-untrusted profile
preset-ibcf-external preset-ibcf-external profile
preset-ibcf-internal preset-ibcf-internal profile
preset-p-cscf-access preset-p-cscf-access profile
preset-peering preset-standard-non-ims profile
```

Configuration Examples

The following example displays detailed output for adjacency client, including the "Inherit profile:" field that shows that the adjacency has been configured with the non-IMS preset-access profile:

Router# show sbc mySBC sbe adjacencies client detail

SBC Service "mySBC"	
Adjacency client (SIP)	
Status:	Attached
Signaling address:	200.0.0.12:5062, VRF Admin
Signaling-peer:	200.0.0.30:5062
Remote address:	200.0.0.0 255.255.255.0
Force next hop:	No
Account:	
Group:	None
In header profile:	Default
Out header profile:	Default
In method profile:	Default
Out method profile:	Default
In UA option prof:	Default
Out UA option prof:	Default
In proxy opt prof:	Default
Out proxy opt prof:	Default
Priority set name:	None
Local-id:	None
Rewrite REGISTER:	On
Target address:	None
NAT Status:	Auto Detect
Reg-min-expiry:	3000 seconds
Fast-register:	Enabled
Fast-register-int:	30 seconds
Authenticated mode:	None
Authenticated realm:	None
Auth. nonce life time:	300 seconds
IMS visited NetID:	None
Inherit profile:	preset-access
Force next hop:	No
Home network Id:	None
UnEncrypt key data:	None
SIPI passthrough:	No
Rewrite from domain:	Yes
Rewrite to header:	Yes
Media passthrough:	No
Preferred transport:	UDP

Hunting Triggers:	Global Triggers
Redirect mode:	Pass-through
Security:	Untrusted
Outbound-flood-rate:	None
Ping-enabled:	No
Signaling Peer Status:	Not Tested





Cisco Unified Border Element (SP Edition) Registration Features

Cisco Unified Border Element (SP Edition) supports the SIP Fast Registration, SoftSwitch Shielding, Registration Monitoring, Aggregate Registration, Provisioned Delegate Registration, and Contact Username Passthrough features in the unified model.

SIP Fast Registration addresses the problem where SIP messages to the Network Address Translation (NAT) endpoint are unable to penetrate the NAT and firewalls to establish calls. Using SIP Fast Registration, the NAT endpoints transmit SIP REGISTER requests at a high enough frequency to keep the NAT pinhole alive.

The SoftSwitch Shielding feature allows a lower SIP registration rate on the links to registrars (typically softswitches) than on the links to endpoints. Allowing a lower registration rate shields the softswitch from an undesirably high rate of re-registrations.

Cisco Unified Border Element (SP Edition) supports monitoring events subscription for changes of the registration state with the Registration Monitoring functionality.

Aggregate Registration registers all the endpoints connected to it in a single registration. This functionality enables Cisco Unified Border Element (SP Edition) to support devices that implicitly register multiple endpoints through it.

The Provisioned Delegate Registration feature allows the Cisco Unified Border Element (SP Edition) to support client or end user devices that cannot register themselves in a network where SIP calls are passing through a registrar. Cisco Unified Border Element (SP Edition) is able to register on behalf of such client devices. The Provisioned Delegate Registration feature can support Cisco Telepresence systems where the end user applications cannot send the registration message and Cisco Unified Border Element (SP Edition) does it on their behalf.

The Contact Username Passthrough enhancement enables interoperability with softswitches that require the contact username portion of the Contact URI in SIP REGISTER requests to pass through unchanged.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



These features are supported in the unified model.

Γ

Feature History	or Registration	Features
-----------------	-----------------	----------

Release	Modification
Cisco IOS XE Release 2.4	The SIP Fast Registration, SoftSwitch Shielding, Registration Monitoring, Aggregate Registration, and Delegate Registration features were introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 2.5	The Contact Username Passthrough for non-IMS networks and Support for Supported Path Under REGISTER Request features were added.
Cisco IOS XE Release 3.1S	The Per Subscriber Delete feature was added.
Cisco IOS XE Release 3.2S	The adding expires-header to register-message feature was added.
Cisco IOS XE Release 3.3S	The Alternative Contact Rewriting feature was added.

Contents

This chapter contains the following sections:

- Prerequisites, page 22-3
- Restrictions, page 22-3
- Information About SIP Registration, page 22-3
- Adding an Expires-Header to a Register-Message, page 22-4
- Support for Supported Path Under REGISTER Request, page 22-6
- Information About Contact Username Passthrough, page 22-6
- Configuring Contact Username Passthrough, page 22-7
- Information About Alternative Contact Rewriting, page 22-9
- Configuring Alternative Contact Rewriting, page 22-10
- Information About SIP Fast Registration, page 22-12
- Configuring SIP Fast Registration, page 22-15
- Information About SoftSwitch Shielding, page 22-16
- Configuring SoftSwitch Shielding, page 22-17
- Information About Registration Monitoring, page 22-19
- Configuring Registration Monitoring, page 22-20
- Information About Per Subscriber Delete, page 22-21
- Configuring Per Subscriber Delete, page 22-21
- Information About Aggregate Registration, page 22-22
- Configuring Aggregate Registration, page 22-22
- Information About Provisioned Delegate Registration, page 22-24
- Provisional Delegate Registration Commands, page 22-26
- Configuration Examples, page 22-30

Prerequisites

The following prerequisite is required to implement SoftSwitch Shielding, Registration Monitoring, Aggregate Registration, and Provisioned Delegate Registration:

Before implementing these features, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions

SIP Fast Registration has the following restrictions:

- Only UDP is supported.
- REGISTERs with a zero expiry time ("Unregisters") are always forwarded to the registrar and not fast-pathed, if the SBC matches them to a known registration.
- Minimal parsing of REGISTER requests is performed before a decision is taken to send a fast-path response; this minimizes the load on the SBC. A REGISTER request is only fast-pathed if its expiry interval is not zero and it comes from the same IP address and port as a known subscription.
- Endpoints that send their requests from ephemeral (short-lived) ports do not have their registration requests fast-pathed.
- The "FastReg interval" cannot be higher than the "MinExpiry interval." If the "MinExpiry interval" is less than twice the "FastReg interval," fast-pathing is not performed.

Provisioned Delegate Registration has the following restrictions:

- The delegate registration configuration is limited to no more than 1000 subscribers with each subscriber having no more than 5 contacts.
- H.323 adjacencies and SIP to H.323 interworking are not supported in Cisco IOS XE Release 2.4 and earlier.

Information About SIP Registration

Registration is required if the end user has a dynamic IP address, if the provider does not support static hostnames, or if NAT is used.

In a SIP REGISTER message, the Contact: header contains the URI that identifies the subscriber.

When a device registers in a non-IMS network, Cisco Unified Border Element (SP Edition) takes the SIP REGISTER Contact: header and modifies it by replacing the contact username with a hash to produce a unique contact username. This is the default behavior for a typical registration. This is needed because there may be multiple UNI adjacencies in different VLANS, which have similar contacts. (Note that this does not apply to the IMS P-CSCF profile.) Then, the SBC forwards the REGISTER to the registrar containing this new modified Contact: header. Meanwhile, the SBC will also store a record of the original contact and the modified contact in its internal memory.

When the core network desires to ring that subscriber, the SBC will receive an INVITE containing the modified contact information. The Cisco SBC will check its memory to look up the information, and will swap out the header with the original information, and will direct the call to the appropriate SIP adjacency in the correct customer network.

This means that no explicit call routing detail needs to be configured in the call policy for routing calls from the core to subscribers, since the SBC has its internal memory of registrations.

Note that even though a SIP adjacency may be intended to receive only subscriber (registered) traffic, it is still possible for unregistered callers to initiate calls from that same adjacency. This can be considered useful, because emergency callers therefore may not need to register first.

When the call arrives at the softswitch, it can check if the subscriber has registered or not, and if to allow the call or not.

In the case where softswitch interoperability is desired, you may want to pass through the contact username instead of hashing it. Cisco Unified Border Element (SP Edition) provides a Contact Username Passthrough enhancement for non-IMS networks. See Information About Contact Username Passthrough, page 22-6.

Adding an Expires-Header to a Register-Message

Some registrars or endpoints might fail to understand the Expires parameters configured in the contact URI. To overcome this issue, you can configure the SBC to add an Expires header to the register messages.

In SIP, the expiry time of a registration is specified by registering the endpoints using the following:

- · Expires header
- The Expires parameter on the registered contact URI.
- The registrar can choose an expiration period, if no expirations period is specified.

Configuring SBC to Add an Expires-Header

To configure SBC to add an Expires header to the register messages, complete the following steps:

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. expires-header
- 6. softswitch-shield
- 7. exit
- 8. end
- 9. show sbc sbc-name sbe adjacencies adjacency-name Detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	expires-header options	Adds the Expires parameter in a SIP contact header.
	Example:	Use the <i>options</i> argument to specify one of the following strings for adding expires to the header:
	Router(config-sbc-sbe-adj-sip)# expires-header add-not-present	• <i>add-not-present</i> —The SBC provides expiry information in the format provided by an endpoint, or as indicated by other configurations.
		• <i>add-smallest</i> —The value of the Expires header is set to the value of the smallest Expires parameter in any provided contact.
		• <i>add-value</i> —The SBC adds an Expires header to any REGISTER request that is sent out on the specified adjacency that does not contain an expiry value.
Step 6	softswitch-shield	Enables softswitch shielding on the SIP.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# softswitch-shield</pre>	
Step 7	exit	Exits the adj-sip-ping mode, and moves to adj-sip mode.
	Example: Router(config-sbc-sbe-adj-sip-ping)# exit	

	Command or Action	Purpose
Step 8	end	Exits the SBE mode and returns to the privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe)# end	
Step 9	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Lists the configured Expires headers for the specified adjacency.
	Example: Router# show sbc mysbc sbe adjacencies sipGW detail	

Support for Supported Path Under REGISTER Request

Starting with Cisco IOS XE Release 2.5, Cisco Unified Border Element (SP Edition) supports the use of the Path extension header field in the Supported field of a REGISTER Request. The Path field provides a way to accumulate and send a list of proxies between a SIP user agent and a registrar. For information on the Path field, see RFC 3327 Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts.

Information About Contact Username Passthrough

The Contact Username Passthrough feature enables interoperability with softswitches that require the contact username portion of the Contact URI in SIP REGISTER requests to pass through unchanged. In certain situations in non-IMS networks, a softswitch may be unable to operate with Cisco Unified Border Element (SP Edition) rewriting or hashing the contact username portion of the Contact URI in SIP REGISTER requests. In these cases, subscribers may not be able to register through the SBC unless you configure the SBC to pass through the contact username.

In a typical SIP registration process, the default behavior is that Cisco Unified Border Element (SP Edition) rewrites the URIs in the Contact headers of REGISTER requests sent by subscribers for the following reasons:

- To remain on the signaling path for requests sent to this subscriber.
- To disambiguate subscribers that register from different devices with the same private username by replacing the username part of the Contact URI with a unique string.

For example, the SBC receives two REGISTER requests, with the following contact URIs using the same username, "bob":

bob@1.1.1.1

bob@2.2.2.2

In each case, the REGISTER contains a Contact URI with the SBC's address. The SBC replaces or rewrites the username "bob" in each URI with a unique string to disambiguate them.

In Cisco IOS XE Release 2.5 and later, you can choose to configure the SBC to pass through, not rewrite, the contact username on SIP REGISTER requests by ensuring that each contact username associated with a given subscriber uses a different port number. By using unique ports for each contact sent to the registrar, the SBC can uniquely correlate to the registered endpoints without requiring a unique username. This can be configured for each adjacency facing the registrar.

The following is an example of contact username passthrough where the username "bob" is passed through unchanged and the hostport is rewritten with the address of the SBC and a unique port number:

sip:bob@1.1.1.1 -----> sip:bob@192.168.101.1:5060

See the ?\$paranum>Contact Username Passthrough Examples? section on page 22-38 for more configuration examples.



This feature has no effect in IMS deployments where the SBC does not rewrite contact usernames.

Configuring Contact Username Passthrough

You can use the **registration contact username passthrough** and the **signaling-port** commands in the (config-sbc-sbe-adj-sip) configuration mode to configure the Contact Username Passthrough feature.

The **registration contact username** command with the **passthrough** option allows you to specify that the contact username in the SIP REGISTER request should be passed through unchanged when rewriting contacts. This option should be enabled on the registrar-facing adjacency. The **passthrough** option disambiguates subscribers that register from different devices with the same private username by using a unique local port number when multiple contact URIs are registered for the same public ID. The range of valid signaling ports are configured with the signaling-port command on a registrar-facing adjacency.

If you do not specify the *max-port-num* option in the **signaling-port** command on this adjacency, then the SBC is not able to disambiguate subscribers that register from different devices with the same username.

The default is the **rewrite** option which allows the username to be changed when rewriting contacts.



If the contact username is longer than 32 characters, then it is *not* passed through and is replaced with a hash as is the case when the default **rewrite** option is chosen.

The following example configures the SBC to specify that the contact username is passed through unchanged.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. no attach
- 6. registration contact username {passthrough | rewrite}
- 7. signaling-port port-num [max-port-num]
- 8. exit
- 9. end
- 10. show sbc sbe adjacencies

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures the adjacency (facing the registrar), and enters into adjacency sip configuration mode.
	<pre>Example: Router(config-sbc-sbe)# adjacency sip adj1</pre>	
Step 5	no attach	(Optional) Use this command to detach an existing adjacency so it is not active before modifying it.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# no attach</pre>	
Step 6	registration contact username {passthrough rewrite}	Specifies whether the contact username in the SIP REGISTER request is passed through unchanged when rewriting contacts.
	Example: Router(config-sbc-sbe-adj-sip)# registration contact username passthrough	This option must be enabled on the registrar-facing adjacency.
		The passthrough keyword disambiguates subscribers that register from different devices with the same private username by using a unique local port number when multiple contact URIs are registered for the same public ID.
Step 7	<pre>signaling-port port-num [max-port-num] Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5060 5062</pre>	Configures range of valid signaling ports on a registrar-facing adjacency to allow the SBC to disambiguate subscribers that register from different devices with the same username.
		<i>max-port-num</i> is the range from 1 through 65535.
		If both <i>port-num</i> and <i>max-port-num</i> are specified, then the <i>port-num</i> indicates the lower boundary of the range and <i>max-port-num</i> indicates the upper boundary of the range. If no <i>max-port-num</i> is specified, then the adjacency listens only on the single <i>port-num</i> . <i>Max-port-num</i> only needs to be set if a range of local listen ports is required for this adjacency.

	Command or Action	Purpose
Step 8	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 9	end	Exits SBE configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 10	show sbc sbe adjacencies	Displays SBC adjacencies.
	Example: Router# show sbc sbe adjacencies	

Information About Alternative Contact Rewriting

In a User-to-Network Interface (UNI) deployment scenario, the endpoint registers to registrar, softswitch, or proxy through the SBC. The SBC rewrites the received contact header to use its own signaling address and keeps itself in the call signaling flow. The SBC maintains the mapping between the received and forwarded contact information. This ensures that the contacts received from the multiple devices are unique, and provides anonymity to the subscribers.

Prior to Cisco IOS XE Release 3.3S. the SBC would rewrite the contact by replacing the entire contact with an alphanumeric string generated by hashing the received contact information. However, registrars can determine that the multiple registrations are for the same Address of Record (AOR) by comparing an initial section of the user-info in the contact header. For example, they determine that the two contacts for 02083661177-abc@sbc.com and 02083661177-xyz@sbc.com are for the same endpoint.

From Cisco IOS XE Release 3.3S, the SBC rewrites the contact header in the following two methods:

- Hashed value of hexadecimal characters—<*DN>* + "-" + <*hashed_value>*, the <*hashed_value>* is a randomly generated value and is unique for a specific endpoint, so that the softswitch can identify those endpoints and initiate forking. Forking is an multiple calls attempt to the endpoints for a single AoR.
- Username of rewritten contact Uniform Resource Identifier (URI) only includes numeric hashed value.

Delegate Registration

When a delegate subscriber is configured on a preset-access adjacency, the contact header sent to the registrar is rewritten similar to the contact header received on a REGISTER message. Therefore, the Alternative Contact Rewriting feature applies to a delegate registration also. The *original contact* provided to the user on exit and used to generate the rewritten contact is the contact that is configured using the **sip-contact** *contact uri* command under the SBE Subscriber Entry mode.

Restrictions on Alternative Contact Rewriting

The feature has the following restriction:

• If there is no username in a contact URI, 32 digits hashed username is used. However, if the original username is 24 bytes or more in length, the username is rewritten in the <23 digit numeric hash>-<8 digit numeric hash> format.

Configuring Alternative Contact Rewriting

This task explains how to configure the Alternative Contact Rewriting feature.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. registration contact username rewrite [numeric | userid-and-numeric]
- 6. end
- 7. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode. Use the <i>sbc-name</i> argument to define the name of the service.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the adjacency facing the registrar, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip pe42	Note The Alternative Contact Support feature does not support the H.323 adjacencies.

	Command or Action	Purpose
Step 5	registration contact username rewrite [numeric userid-and-numeric]]	Configures the contact username in a SIP REGISTER request so that it can be modified.
	Example:	• rewrite —Allows the contact username in a SIP REGISTER request to be changed or rewritten.
	Router(config-sbc-sbe-adj-sip)# registration contact username rewrite userid-and-numeric	• numeric —Rewrites the contact username in a SIP REGISTER request as an originating hashed numeric value.
		• userid-and-numeric —Rewriting the contact username in a SIP REGISTER request as an originating userid plus hashed numeric value.
Step 6	end	Exits adjacency SIP configuration mode and returns to Exec mode.
	Example: Router(config-sbc-sbe)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays the detailed field output for the specified SIP adjacency.
	Example: Router# show sbc sbe mySBC sbe adjacencies pe42	

The following example show the output of the show sbc sbe adjacencies detail command:

Router# show sbc pe41 sbe adjacencies pe42 detail

```
SBC Service "pe41"
 Adjacency pe42 (SIP)
   Status:
                        Attached
   Signaling address: 88.41.41.41:5060
                      0
   IPsec server port:
   Signaling-peer:
                         88.42.42.42:5060 (Default)
   Signaling-peer status: Not Tested
   Signaling-peer priority: 2147483647
   Signaling-peer switch: always
   Peer status: Not Tested
   Current peer index:
                             0
   Force next hop:
                        No
   Force next hop select: Out-of-dialog
   Admin Domain:
                        None
   Account:
   Group:
                         None
   Rewrite REGISTER:
                       Off
   Register contact username: Rewrite as userid and digits
   Target address: None
   NAT Status:
                        Auto Detect
   Reg-min-expiry:
                       3000 seconds
   Local Jitter Ratio: 0/1000
```

Information About SIP Fast Registration

SIP Fast Registration performs the following functions:

- Prompts endpoints to register frequently with Cisco Unified Border Element (SP Edition), causing NAT/Firewall pinholes to remain open.
- Protects internal network elements from the large number of REGISTER messages arising from endpoint registration.
- Allows Cisco Unified Border Element (SP Edition) to do minimal processing of REGISTER messages, which improves performance. This is particularly important when dealing with significant oversubscription—where there typically may be 10 times more subscribers than active calls.

When Cisco Unified Border Element (SP Edition) faces the customer premise side and the customer on the network edge deploys Network Address Translation (NAT) and firewalls, SIP INVITEs to the NAT endpoint are unable to penetrate the NAT to establish calls. To overcome the problem, the endpoints transmit SIP REGISTER requests at a high enough frequency to keep the NAT pinhole alive. The SIP Fast Registration feature off loads the processing from the registrar for a large number of endpoints and generates SIP REGISTER replies from the data plane (forwarding services provided by Cisco QuantumFlow Processor (QFP)). This also limits the impact on the router's CPU load. The QFP processes the expected SIP re-register messages by short-cutting the REGISTER messages and quickly turning them around.

Typically the registrar responds to the first SIP REGISTER message asking the end-point to send its next SIP REGISTER message within 3600 seconds (configurable). With the SIP Fast Registration feature, Cisco Unified Border Element (SP Edition) intercepts this Reply and informs the end-point to REGISTER every 30 seconds to keep the NAT open. The Route Processor (RP) programs a SIP Fast Registration (SFX) entry with a fast expiry times parameter in seconds. When the fast expiry times parameter expires, the QFP punts the SIP REGISTER message to the RP to update the state before forwarding it to the registrar.

Fast registration is configured per SIP adjacency, on the endpoint-facing adjacency, that is, the adjacency which receives the incoming REGISTER request. After an endpoint has registered using fast registration through an adjacency, all subsequent registration requests from the same endpoint are responded to by Cisco Unified Border Element (SP Edition), without notifying the softswitch, until the registration interval has almost expired.

The following shows a fast registration configuration example:

```
Reg-min-expiry: 300 seconds
Fast-register: Enabled
Fast-register-interval: 60 seconds
Register aggregate: Disabled
...
```

In the example above, Cisco Unified Border Element (SP Edition) performs fast registrations at 60 second intervals, and will send registrations towards the registrar/softswitch at intervals of 480 seconds. This is calculated by a hard-coded algorithm of (3 x fast-register-interval) + (reg-min-expiry), plus taking into account the expiry time in the inbound registration from the endpoint.

In the above example, Softswitch Shielding is not enabled. Thus if incoming expiry time from the endpoint is 400 seconds, which is less than 480 seconds, then the incoming registration interval to the registrar/softswitch is calculated as 480 seconds. However, if the incoming expiry time from the endpoint is 600 seconds, which is larger than 480 seconds, then the incoming registration interval to the registrar/softswitch is calculated as 600 seconds.

On the other hand, when Softswitch Shielding is enabled, then the Softswitch Shielding timer takes precedence and is always used as the incoming registration interval to the registrar/softswitch. See the ?\$paranum>Information About SoftSwitch Shielding? section on page 22-16.

When fast registration is enabled, the incoming registration time should not be less than the fast-register-interval. Otherwise, the SBC will reject the registration with error message 423 (Interval Too Brief). The SBC compares incoming registration time with the interval set in fast-register-interval, instead of the interval in reg-min-expiry. If fast registration is *disabled*, then the incoming registration time should not be less than the reg-min-expiry time. Otherwise, the SBC will reject the registration with response code 423 (Interval Too Brief).

For information on commands, such as **fast-register-interval** and **reg-min-expiry**, see the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model.*

SIP Fast Registration is not enabled by default. You must configure it with the **inherit profile preset-access** command, either as a global configuration or per adjacency.

Once SIP Fast Registration is configured, fast-pathing is on by default on an adjacency. You can then disable Fast Registration using the **fast-register disable** command.

REGISTER messages are rejected by Cisco Unified Border Element (SP Edition) with a 423 Interval Too Brief response code under the following conditions:

• If Fast Registration is enabled and the Expires header in the REGISTER message is less than the inbound adjacency's "**fast-register-interval**."

The **fast-register-interval** command controls the recommended rate at which endpoints send REGISTER requests. The lower this value, the more frequently endpoints re-register, thus keeping a NAT/firewall pinhole open. Therefore, we recommend configuring this to a slightly lower value than the pinhole timeout, if that is known.

• If Fast Registration is not enabled and the Expires header in the REGISTER messages is less than the inbound adjacency's "**reg-min-expiry**."

The **reg-min-expiry** command controls the rate at which REGISTER requests are sent from the SBC to the SIP registrar. The lower this value, the greater the potential register load on the softswitch. If fast-pathing is not enabled for an adjacency, SBC rejects any REGISTER requests with a shorter expiry interval than the **reg-min-expiry** command.

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Figure 22-1 illustrates where network elements are located in a network configured with Fast Registration, SoftSwitch Shielding, and Aggregate Registration.

Figure 22-1 Voice Network Elements in a Fast Registration, SoftSwitch Shielding, and Aggregate Registration Network



Restrictions for SIP Fast Registration

The restrictions for SIP Fast Registration are the following:

- Only UDP is supported.
- REGISTERs with a zero expiry time ("Unregisters") are always forwarded to the registrar and not fast-pathed, if the SBC matches them to a known registration.
- Minimal parsing of REGISTER requests is performed before a decision is taken to send a fast-path response; this minimizes the load on the SBC. A REGISTER request is only fast-pathed if its expiry interval is not zero and it comes from the same IP address and port as a known subscription.
- Endpoints that send their requests from ephemeral (short-lived) ports do not have their registration requests fast-pathed.
- The fast-register-interval cannot be higher than the reg-min-expiry (minimum expiry value). If the minimum expiry value is less than twice the fast-register-interval, fast-pathing is not performed.

Configuring SIP Fast Registration

This task configures a basic SIP Fast Registration on an adjacency.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}
- 6. exit
- 7. end
- 8. show platform hardware qfp active feature sbc sfx

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency {sip h323} adjacency-name	Configures the adjacency (facing the subscriber), and enters into adjacency sip configuration mode.
	<pre>Example: Router(config-sbc-sbe)# adjacency sip adj1</pre>	Note H.323 adjacencies are not supported in Cisco IOS XE Release 2.4 and earlier.
Step 5	<pre>inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}</pre>	Configures an inherit profile for the SIP adjacency. The SIP adjacency must be configured to preset-access for Fast Register. An access adjacency faces user equipment, such as a subscriber's telephone or other SIP device, that attempts to register through the SBC.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-access</pre>	The default is preset-core.

	Command or Action	Purpose
Step 6	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 7	end	Exits SBE configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 8	show platform hardware qfp active feature sbc sfx	Displays the QFP SIP Fast-Register (SFX) counters.
	Example: Router# show platform hardware qfp active feature sbc sfx global	

Information About SoftSwitch Shielding

Cisco Unified Border Element (SP Edition) supports the SoftSwitch Shielding feature that allows a lower SIP registration rate on the links to registrars (typically softswitches) than on the links to endpoints. In a network where endpoints frequently refresh their SIP registrations to a softswitch, allowing a lower registration rate shields the softswitch from an undesirably high rate of re-registrations while ensuring the softswitch still has accurate knowledge of registered endpoints.

For example, if endpoints are sending REGISTER messages to inform the proxy server of the callee address location every 15 minutes, Cisco Unified Border Element (SP Edition) can be configured to only forward REGISTER messages to the softswitch every 12 hours, unless there is a change to the contact being registered. Using SoftSwitch Shielding reduces the load on the softswitch and the network. On the other hand, if an endpoint stops sending REGISTER messages, Cisco Unified Border Element (SP Edition) detects the change within the endpoint's expiry interval and removes the subscriber state, thus preventing calls to or from this endpoint.

The SoftSwitch Shielding feature gives Cisco Unified Border Element (SP Edition) the capability to shield the softswitch from a large portion of the registration processing. You are also able to simultaneously configure the SIP Fast Registration feature and the SoftSwitch Shielding feature. In addition, if the REGISTER message contains an Authorization header, Cisco Unified Border Element (SP Edition) forwards the REGISTER message to the softswitch registrar.

Use the **registration outgoing timer** command to enable SoftSwitch Shielding and set the time interval during which the SBC forwards REGISTER messages to the softswitch before timing out.
Figure 22-2 illustrates a SoftSwitch Shielding call flow.



Figure 22-2

SoftSwitch Shielding Call Flow

Configuring SoftSwitch Shielding

This task configures SoftSwitch Shielding on an adjacency.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency { sip | h323 } adjacency-name
- 5. registration outgoing timer sec
- 6. registration rewrite-register
- 7. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}

- 8. exit
- 9. end
- 10. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the adjacency facing the registrar, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip SoftSwitch	Note H.323 adjacencies are not supported in Cisco IOS XE Release 2.4 and earlier.
Step 5	<pre>registration outgoing timer sec Example: Router(config-sbc-sbe-adj-sip)# registration outgoing timer 36000</pre>	Enables SoftSwitch Shielding and sets the registration timeout timer for the time interval during which the SBC forwards outgoing REGISTER messages to the softswitch before timing out. <i>sec</i> —value is 1 to 2147483647. The default of zero disables SoftSwitch Shielding.
		In this example, the time interval is set to every 10 hours or 36000 seconds.
Step 6	registration rewrite-register	Configures the SIP register request rewriting on an adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# registration rewrite-register</pre>	
Step 7	<pre>inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}</pre>	Configures a global inherit profile for the SIP adjacency. An adjacency facing the registrar typically has a preset-core profile. The default is preset-core.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-core</pre>	

	Command or Action	Purpose
Step 8	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 9	end	Exits SBE configuration mode and returns to Exec mode.
	Example: Router(config-sbc-sbe)# end	
Step 10	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the detailed field output for the specified SIP adjacency, including the "Register Out Timer:" field that shows the time interval in seconds when the SBC forwards
	Example:	the next REGISTER messages to the softswitch.
	Router# show sbc sbe mySBC sbe adjacencies SoftSwitch detail	

Information About Registration Monitoring

Cisco Unified Border Element (SP Edition) supports creating event subscriptions for changes of registration state. Event subscriptions are generally network-initiated de-registrations. This is a requirement of the IP Multimedia Subsystem (IMS) specifications for Proxy-Call Session Control Function (P-CSCF), a SIP proxy server. For more information, refer to 3rd Generation Partnership Project (3GPP) TS 24.229 v7.5.1.

This support is configured on a per-adjacency basis through the registration monitor field of the Adjacency Table. If this field is set, then Cisco Unified Border Element (SP Edition) creates an event subscription with the registrar for each registered subscriber situated on the adjacency.

The registrar uses the event subscription to provide active indications of changes to the state of the registration. Based on these indications, Cisco Unified Border Element (SP Edition) adds, removes, or updates subscriber state, as appropriate. For more information on event subscriptions, refer to RFC 3680.

Cisco Unified Border Element (SP Edition) does not clean up fast register configuration in the event of a network-initiated de-registration. In this case, the user equipment (UE) is not able to re-register with the registrar until the fast register time period expires.

Cisco Unified Border Element (SP Edition) sets the duration of the monitoring subscription to be the maximum expired interval of the subscriber's contacts plus a configurable constant. The default monitoring subscription duration is 32 seconds.

Cisco Unified Border Element (SP Edition) only re-subscribes to the Serving-Call Session Control Function's (S-CSCF) monitoring state when the UE sends a re-register through the SBC. Cisco Unified Border Element (SP Edition) does not follow the 3GPP model of refreshing subscriptions 600 seconds before they expire. The SBC implementation reduces the load on the P-CSCF and S-CSCF, both SIP servers, while ensuring that the subscription lifetime exceeds the registration lifetime, which ensures that network-initiated de-registrations are always detected.

Use the **registration monitor** command to enable monitoring of event subscriptions for registration state changes.

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Configuring Registration Monitoring

This task configures how to monitor event subscriptions as a result of registration state changes.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. registration monitor
- 6. exit
- 7. end
- 8. show sbc *sbc-name* sbe adjacencies *adjacency-name* detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the adjacency facing the registrar, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip Cary-IP-PBX	
Step 5	registration monitor	Enables monitoring of event subscriptions as a result of registration state changes.
	Example: Router(config-sbc-sbe-adj-sip)# registration monitor	
Step 6	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

	Command or Action	Purpose
Step 7	end	Exits SBE configuration mode and returns to Exec mode.
	Example: Router(config-sbc-sbe)# end	
Step 8	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the detailed field output for the specified SIP adjacency, including the "Registration Monitor:" field that shows Registration Monitoring is "Enabled."
	Example: Router# show sbc sbe mySBC sbe adjacencies Cary-IP-PBX detail	

Information About Per Subscriber Delete

The Per Subscriber Delete feature provides a mechanism for manually deleting individual subscribers, and associated registered contacts or other subscriber state if any, from the database. This feature works on both manually created subscriber entries and dynamically created entries during the standard registration process. It does not cause the SBC to signal either the subscriber or the registrar; it only removes the internal state from the SBC associated with that subscriber.

Use the **clear sbc sbe sip subscriber aor** command to clear the stuck registrations on Cisco ASR 1000 Series Routers.

Configuring Per Subscriber Delete

This section shows how to clear stuck registrations.

SUMMARY STEPS

- 1. show sbc *sbc-name* sbe sip subscribers
- 2. clear sbc sbc-name sbe sip subscriber aor address-of-record

DETAILED STEPS

	Command or Action	Purpose
Step 1	<pre>show sbc sbc-name sbe sip subscribers Example: Pouter# show sbc asr sbe sip subscribers</pre>	Displays details of all the SIP endpoints that have registered with the SBC. Information about the Address of Record (AOR) for each subscriber is also displayed.
Sten 2	clear shc shc-name she sin subscriber aor	Clears the stuck registrations on
otep 2	address-of-record	Cisco ASR 1000 Series Routers.
	Example:	
	Router# clear sbc asr sbe sip subscriber aor sip:alice@open-ims.test	

Information About Aggregate Registration

A registrar is typically a registration server in a SIP network, but outside of the Cisco Unified Border Element (SP Edition) device. The registrar accepts and processes registration requests that register one or more IP addresses to a specific URI, usually a "sip:" address. Because SIP endpoints need to know each others IP address, the registrar acts as a location service. More than one user agent can register at the same IP address. When a call is placed to that IP address, all the registered user agents will ring.

Cisco Unified Border Element (SP Edition) supports Aggregate Registration where a single registration is implemented that causes the registrar to implicitly register multiple IP addresses. The SBC performs aggregate registration for endpoints connected to it. Thus Cisco Unified Border Element (SP Edition) can support devices that implicitly registers multiple endpoints through it. This way of registering all endpoints connected to it in a single registration can be compared to what is commonly done by an IP-PBX device.

The Aggregate Registration feature allows single registration on behalf of multiple endpoints and implicit registration of single endpoints behind the Internet Protocol Private Branch eXchange (IP-PBX).

Aggregate registration is configured on a per-adjacency basis and is configured under an adjacency. All end user clients attached to the adjacency can perform aggregate registration.

When an adjacency has aggregate registration support enabled, the SBC behaves as follows:

- On receiving a REGISTER message, the SBC caches the top Via header and stores it with the normal registration details.
- On receiving an INVITE or out-of-dialog request on the adjacency, Cisco Unified Border Element (SP Edition) attempts to look up the registration using the top Via header, not the Contact and From headers. This ensures that the SBC routes the call to the registrar correctly.
- On receiving an INVITE or out-of-dialog request to the adjacency, Cisco Unified Border Element (SP Edition) overwrites the Request URI as follows:
 - The username is overwritten with the username in the P-Called-Party-Id header if present, or the To header if not.
 - The hostname is overwritten with the hostname that was present on the Contact header that the PBX registered.

Use the registration aggregate command to enable Aggregate Registration support from an adjacency.

Configuring Aggregate Registration

This task configures Aggregate Registration on an adjacency.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. registration rewrite-register

- 6. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}
- 7. registration aggregate
- 8. header-name [contact [add [tls-param]] | from{passthrough} | to{passthrough}]
- 9. request-line request-uri rewrite
- 10. exit
- 11. end
- 12. show sbc sbc-name sbe adjacencies adjacency-name detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency {sip h323} adjacency-name	Configures the adjacency facing the registrar, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip Cary-IP-PBX	Note H.323 adjacencies are not supported in Cisco IOS XE Release 2.4 and earlier.
Step 5	registration rewrite-register	Configures the SIP register request rewriting on an adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# registration rewrite-register	
Step 6	<pre>inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}</pre>	Configures a global inherit profile for a SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-access</pre>	

	Command or Action	Purpose
Step 7	registration aggregate	Enables Aggregate Registration support from the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# registration aggregate	Note This step and the next two steps in the correct order (header-name and request-line request-uri rewrite) are required to enable aggregate registration call routing to work completely.
Step 8	header-name [contact [add [tls-param]] from {passthrough} to {passthrough}]	Configures the passthrough header for non-REGISTER requests.
	Example: Router(config-sbc-sbe-adj-sip)# header-name to passthrough	
Step 9	<pre>request-line request-uri rewrite Example: Router(config-sbc-sbe-adj-sip)# request-line request-uri rewrite</pre>	Allows outgoing calls to the endpoint registered with Aggregate Registration. The SBC rewrites the Request-URI as <user>@<hostname>, before sending a request to the registered subscriber (IP-PBX) on this adjacency.</hostname></user>
Step 10	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 11	end	Exits SBE configuration mode and returns to Exec mode.
	Example: Router(config-sbc-sbe)# end	
Step 12	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the detailed field output for the specified SIP adjacency, including the "Register Aggregate:" field that shows Aggregate Registration is "Enabled."
	Example: Router# show sbc sbe mySBC sbe adjacencies Cary-IP-PBX detail	

Information About Provisioned Delegate Registration

In a SIP network, some third-party client or end user devices are unable to register themselves with a registrar. A registrar is a server that resides outside the SBC device. These end users or clients are generally the applications running on systems used by people. The application may be a softphone application running on your PC or a messaging device in your IP phone. For example, the softphone application generates a request when you try to call another person over the network and sends the request to a server. The generated request is a register message or registration.

End users register their locations or addresses to a registrar server. By registering or sending a special message to a registrar server, the registrar server maintains updated locations of end users in a SIP network. The client then sends the request to a proxy server because when the request is generated, the address of the recipient or callee is not known. The registration is sent to inform a proxy server of the location of the callee address.

Cisco Unified Border Element (SP Edition) can be configured to register any client devices that cannot register themselves. Using the Provisioned Delegate Registration feature, you can set up delegate registration for individual client devices. The client device can make and receive calls as though it had registered normally. Additionally, you can specify individual parameters for each client device, such as registering the client device to a specified registrar server.

Provisioned Delegate Registration is done by provisioning Cisco Unified Border Element (SP Edition) with enough information about a client device so that it can originate a registration for the device itself. Cisco Unified Border Element (SP Edition) can perform end user registration for up to several hundred to a thousand end user clients or delegate clients.

Provisioned Delegate Registration supports the following functionalities:

- Cisco Unified Border Element (SP Edition) can be configured to register with a SIP registrar on behalf of another SIP network entity the delegate client.
- Users can configure the following information per delegate client:
 - The registrar where the end user client is registered.
 - The registrar facing adjacency if the user wants to bypass normal routing.
 - The adjacency facing the delegate client.
 - The expiration time of the registration.
 - The refresh time of the registration.
 - The contact Uniform Resource Identifier (URI) information of the delegate client. Depending on the adjacency configuration, the URI may be rewritten on messages going to the registrar to force calls to the delegate client to be routed through the SBC.
 - The next hop to the delegate client (for calls coming from the registrar).
 - The address of record (AoR) of the delegate client.
 - The number of attempts and frequency of registration retries when a failure is received.
- Calls from the delegate client can be forwarded to one of the following (in order of preference):
 - the entity identified in the Service-Route header if present on the registration response, or
 - the registrar.
- Calls to the delegate client must have the Request URI rewritten to indicate the delegate client, and forwarded out of the adjacency facing it.

Restrictions

The following is a restriction of the Provisioned Delegate Registration feature:

• The delegate registration configuration is limited to no more than 1000 subscribers with each subscriber having no more than 5 contacts.

Provisioned Delegate Registration Call Flow Description

When an end user client is brought up, Cisco Unified Border Element (SP Edition) builds a REGISTER message on behalf of the client, and sends it to the specified registrar. The REGISTER message contains all the contact Uniform Resource Identifier (URI) information that have been configured on the end user client.

If the registrar responds positively to the REGISTER message, Cisco Unified Border Element (SP Edition) stores this fact. Calls to and from the end user client is treated by the SBC exactly as if the client has registered itself.

If the registrar responds negatively to the REGISTER message. For example, if Cisco Unified Border Element (SP Edition) receives error response 423—Interval too brief, the SBC retries building a REGISTER message after the configured retry interval. Cisco Unified Border Element (SP Edition) repeats this process the configured number of times. If the end user client has still failed to be registered, a log is made and the subscriber operating status is changed to OPER_ACT_FAILED.

Before the registration time of the end user client expires, Cisco Unified Border Element (SP Edition) performs the registration processing again to refresh the callee/recipient address location through registration.

See ?\$paranum>Provisional Delegate Registration Commands? section on page 22-26 for more information on configuration steps and commands.

Configuring Delegate Registration Profile

Cisco Unified Border Element (SP Edition) requires a configured subscriber for each end user client upon whose behalf the SBC performs registration. The user may configure retry counts, retry intervals, duration, and the refresh buffer time for each configured subscriber, also called "delegated subscriber." Several subscribers may all share the same nondefault values for the fields in the Delegate Registrations (amb_mw_sudb_subscriber) table. Instead of requiring configuration for each subscriber separately, Cisco Unified Border Element (SP Edition) allows the user to configure a subscriber profile that can be applied to one or more subscribers.

Use the **delegate-profile** *profile-name* command to configure a profile for a delegate registration subscriber.

Provisional Delegate Registration Commands

When the AdminStatus of the Delegate Registrations table is set to AdminStatusUp, Cisco Unified Border Element (SP Edition) attempts to register with the registrar. If the registration is successful, the delegate/client device is treated the same as all other subscribers. Cisco Unified Border Element (SP Edition) registers the device for the length of time specified. Cisco Unified Border Element (SP Edition) renews the registration before it expires, with the specified configurable buffer time.

If a registration (or re-registration) fails, Cisco Unified Border Element (SP Edition) retries registration after the configured delay time. Cisco Unified Border Element (SP Edition) retries the specified number of times. If registration still fails, Cisco Unified Border Element (SP Edition) logs the failure and sets the subscriber operating status to failed.

The following commands are used to configure Provisional Delegate Registration:

- Use the **delegate-profile** command to configure a delegate client registration profile that can be applied to a delegate subscriber. After a delegate profile is configured, profile parameters that specify **duration**, **retry-count**, **retry-interval**, and **refresh-buffer** may optionally be configured.
- Use the **subscriber** *aor* command to define the address of record for the subscriber and define the unique subscriber for whom you want to configure delegate registration. The subscriber must have one or more SIP contacts/URIs associated with it.

- Use the **sip-contact** *uri* command to configure a SIP contact URI for a subscriber. The contact information is used to provision Cisco Unified Border Element (SP Edition) with client device information, so the SBC can register the device. For every delegate registration configured with the **delegate-registration** *hostname* command, one or more SIP contacts/URIs must be configured in the SIP Contacts table (amb_mw_sudb_local_id).
- Use the **delegate-registration** *hostname* command to configure a delegate registration for a delegate client.
- Use the **profile** command to apply a delegate registration profile to a delegate registration subscriber
- Use the **show sbc** *sbc name* **sbe sip subscribers** command to display subscribers for whom Provisioned Delegate Registration has been provisioned.
- Use the **show sbc** *sbc name* **sbe sip delegate-profiles** command to display subscriber profiles for whom Provisioned Delegate Registration has been configured.

Configuring Provisional Delegate Registration

This task configures in order: a profile for a delegate registration subscriber; delegate registration for a specified subscriber associated with an individual client device; delegate registration for a specified client/delegate device; and displays subscribers and subscriber profiles for whom delegate registration have been configured.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. delegate-profile profile name
- 5. duration dur time in secs
- 6. retry-count #times to retry
- 7. retry-interval retry time in secs
- 8. refresh-buffer timeout in secs
- 9. exit
- 10. subscriber aor
- 11. sip-contact uri
- 12. adjacency adjacency name
- 13. exit
- 14. delegate registration hostname
- 15. adjacency adjacency name
- **16**. **profile** my-profile
- 17. activate
- 18. end
- **19**. **show sbc** *sbc name* **sbe sip subscribers delegate**
- 20. show sbc sbc name sbe sip delegate-profiles

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>delegate-profile profile name Example: Router(config-sbc-sbe)# delegate-profile my profile</pre>	Configures a delegate/client registration profile that can be applied to a delegate registration subscriber. Enters into subscriber delegate profile configuration mode where profile parameters can be configured.
		The profile name is a string of 24 characters maximum length.
Step 5	<pre>duration dur time in secs Example: Router(config-sbc-sbe-subscriber-delegate-prof) # duration 100</pre>	Configures the expiration time when the delegate client is due to expire, that is, the length of time in seconds during which the SBC tries to perform delegate registration before stopping.
		The default duration time is 1800 seconds. The range is 1 to 2,147,483 seconds.
Step 6	<pre>retry-count #times to retry</pre>	Configures the number of times the SBC repeats the delegate registration processing after the retry interval ends.
	<pre>Example: Router(config-sbc-sbe-subscriber-delegate-prof) # retry-count 5</pre>	The default is 3 times. The range is 0 to 255 times.
Step 7	retry-interval retry time in secs	Configures the length of time the SBC waits before it retries delegate registration.
	<pre>Example: Router(config-sbc-sbe-subscriber-delegate-prof) # retry-interval 60</pre>	The default is 30 seconds. The range is 1 to 2,147,483 seconds.
Step 8	refresh-buffer timeout in secs	Configures the length of time by which the SBC attempts to renew or refresh the address location with a delegate registration before the specified expiration time (duration).
	Router(config-sbc-sbe-subscriber-delegate-prof) # refresh-buffer 200	The default is 30 seconds. The range is 1 to 2,147,483 seconds.
Step 9	exit	Exits Subscriber Delegate Profile configuration mode and enters SBE configuration mode.
	Example: Router(config-sbc-sbe-del-prof)# exit	

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	Command or Action	Purpose
Step 10	<pre>subscriber aor Example: Router(config-sbc-sbe)# subscriber sip:bob@isp.example</pre>	Configures a delegate registration for a specified subscriber associated with an individual client device. Enters into subscriber-entry configuration mode where SIP contact info can be configured for the delegate registration.
Step 11	sip-contact uri Example:	Configures the SIP contact information for a specified URI IP address location or address of record. Enters into subscriber-contact (SIP) configuration mode.
	Router(config-sbc-sbe-subscriber-entry)# sip-contact sip:steve@10.1.1.2	The URI is a string of 62 maximum character length.
Step 12	adjacency adjacency name	Configures the mandatory local subscriber adjacency name of the configured SIP contact.
	Example: Router(config-sbc-sbe-subscriber-contact)# adjacency CallMgrB	
Step 13	<pre>exit Example: Router(config=shc=she=subscriber=contact)# exit</pre>	Exits subscriber-contact (SIP) configuration mode and enters into subscriber-entry configuration mode to configure delegate registration.
Step 14	delegate-registration hostname Example: Router(config-sbc-sbe-subscriber-entry)#	Configures a delegate registration for a specified client device or delegate client, and enters into subscriber-delegate configuration mode where registration parameters can be set.
Step 15	adjacency adjacency name	Configures the adjacency facing the registrar.
	Example: Router(config-sbc-sbe-subscriber-delegate)# adjacency CallMgrA	
Step 16	<pre>profile profile name Example: Router(config-sbc-sbe-subscriber-delegate)# profile my profile</pre>	Applies the delegate registration profile, created previously with the delegate-profile command, to a delegate registration subscriber.
Step 17	activate	(Required) Activates the delegate registration.
	Example: Router(config-sbc-sbe-subscriber-delegate)# activate	
Step 18	end	Exits subscriber-delegate configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-subscriber-delegate)# end	

	Command or Action	Purpose
Step 19	show sbc sbc name sbe sip subscribers delegate	Displays subscribers for whom delegate registration has been configured. The delegate keyword displays the associated URI contact information for subscribers.
	Router# show sbc mySBC sbe sip subscribers delegate	
Step 20	show sbc sbc name sbe sip delegate-profiles	Displays subscriber profiles for subscribers for whom delegate registration has been configured.
	Example: Router# show sbc mySBC sbe sip	

Configuration Examples

This section has the following configuration examples:

- SIP Fast Registration Example, page 22-30
- SoftSwitch Shielding and Aggregate Registration Configuration Examples, page 22-31
- Registration Monitoring Examples, page 22-35
- Provisional Delegate Registration Examples, page 22-36
- Contact Username Passthrough Examples, page 22-38
- Alternative Contact Rewriting Example, page 22-39
- Registering with Softswitch via Cisco SRP Integrated Access Device (IAD) Examples, page 22-39

SIP Fast Registration Example

Use the **show platform hardware qfp active feature sbc sfx** command to display the QFP SIP Fast-Register (SFX) counters. Information about how SIP fast-register (SFX) messages are processed, including which SIP REGISTER request packets are punted to the Route Processor (RP) or dropped, may help explain why call rates are low and why the RP CPU load is high.

The following example shows information about the parsing of SIP fast-register (SFX) messages in the Cisco QuantumFlow Processor (QFP):

Router# show platform hardware qfp active feature sbc sfx global

SBC QFP SIP Fast Register Data Plane Information _____ SIP 200 OK Replies generated = 0 SIP REGISTER punts : No table entry = 0 Fast Timer expiry = 0 = 0 Expires=0 SIP Syntax Error = 0 QFP Out of Resources = 0 QFP Internal Error = 0 SIP REGISTER drops : QFP Internal Error = 0 UDP Length Error = 0 UDP Checksum Error = 0

SoftSwitch Shielding and Aggregate Registration Configuration Examples

The following is a configuration example showing that Aggregate Registration and SoftSwitch Shielding are configured:

```
sbc test
she
  sip header-profile myheader
   header P-Called-Party-ID entry 1
    action pass
  adjacency sip sippa
                            =====> Adjacency facing IP-PBX
   header-profile inbound myheader
   header-profile outbound myheader
   inherit profile preset-access
   preferred-transport udp
   signaling-address ipv4 99.99.103.150
   signaling-port 5080
   remote-address ipv4 100.100.1.64 255.255.255.255
   signaling-peer 100.100.1.64
   signaling-peer-port 5080
   registration rewrite-register
   account sipp-a
   registration aggregate
   fast-register disable
   header-name to passthrough
   request-line request-uri rewrite
   attach
  adjacency sip sippb
                           =====> Adjacency facing Registrar
   nat force-off
   header-profile inbound myheader
   header-profile outbound myheader
   inherit profile preset-core
   preferred-transport udp
   signaling-address ipv4 99.99.103.150
   signaling-port 5082
   remote-address ipv4 100.100.1.64 255.255.255.255
   signaling-peer 100.100.1.64
   signaling-peer-port 5082
   account sipp-b
   registration target address 100.100.1.64
   registration target port 5084
   fast-register disable
   attach
  cac-policy-set 1
   first-cac-table mytable
   first-cac-scope src-adjacency
   cac-table mytable
    table-type limit adjacency
    entrv 1
     match-value sippa
     max-num-calls 10
     action cac-complete
   complete
  active-cac-policy-set 1
  call-policy-set 1
   first-call-routing-table src-acc-table
   first-reg-routing-table src-acc-table
   rtg-src-adjacency-table src-acc-table
    entry 1
     action complete
     dst-adjacency sippb
     match-adjacency sippa
```

L

```
entry 2
      action complete
      dst-adjacency sippa
      match-adjacency sippb
   complete
   call-policy-set 2
   active-call-policy-set 1
 1
 vdbe global
 unexpected-source-alerting
media-address ipv4 99.99.103.156
media-timeout 9999
activate
!
Softswitch shielding config
_____
sbc test
sbe
   adjacency sip sippa
    signaling-address ipv4 99.99.103.150
    signaling-port 5080
   remote-address ipv4 100.100.1.64 255.255.255.255
   signaling-peer 100.100.1.64
    signaling-peer-port 5080
    registration rewrite-register
    account sipp-a
    attach
   adjacency sip sippb
    signaling-address ipv4 99.99.103.150
    signaling-port 5082
   remote-address ipv4 100.100.1.64 255.255.255.255
    signaling-peer 100.100.1.64
    signaling-peer-port 5082
    account sipp-b
    registration outgoing timer 86400
    registration target address 100.100.1.64
   registration target port 5084
    attach
   call-policy-set 1
    first-call-routing-table src-acc-table
    first-reg-routing-table src-acc-table
   rtg-src-adjacency-table src-acc-table
     entry 1
      action complete
      dst-adjacency sippb
      match-adjacency sippa
     entry 2
      action complete
      dst-adjacency sippa
      match-adjacency sippb
   complete
   active-call-policy-set 1
 !
media-address ipv4 99.99.103.156
media-timeout 9999
activate
ļ
Router# show sbc test sbe adjacencies sippb detail
SBC Service "test"
  Adjacency sippb (SIP)
    Status:
                           Attached
```

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99.99.103.150:5082

Signaling address:

Signaling-peer:	100.100.1.64:5082
Force next hop:	No
Account:	sipp-b
Group:	None
In header profile:	Default
Out header profile:	Default
In method profile:	Default
Out method profile:	Default
In UA option prof:	Default
Out UA option prof:	Default
In proxy opt prof:	Default
Out proxy opt prof:	Default
Priority set name:	None
Local-id:	None
Rewrite REGISTER:	Off
Target address:	100.100.1.64:5084
NAT Status:	Auto Detect
Reg-min-expiry:	3000 seconds
Fast-register:	Enabled
Fast-register-int:	30 seconds
Register aggregate:	Disabled
Register Out Interval:	86400 seconds
Authenticated mode:	None
Authenticated realm:	None
Auth. nonce life time:	300 seconds
IMS visited NetID:	None
Inherit profile:	Default
Force next hop:	No
Home network Id:	None
UnEncrypt key data:	None
SIPI passthrough: N	10
Rewrite from domain:	Yes
Rewrite to header:	Yes
Media passthrough:	No
Client authentication:	No
Hunting Triggers:	Global Triggers
Redirect mode:	Pass-through
Security:	Untrusted-Unencrypted
Signaling Peer Status:	Not Tested
Rewrite Request-uri:	Disabled
Registration Monitor:	Disabled
DTMF SIP NOTIFY Relay:	Enabled
DTMF SIP NOTIFY Interva	al: 2000
DTMF SIP default durat	lon: 200
DTMF Preferred Method:	SIP NOTIFY

The following example configures SoftSwitch Shielding on adjacency "SoftSwitch:"

```
sbc mySbc
sbe
adjacency sip SoftSwitch
registration outgoing timer <sec>
registration rewrite-register
inherit profile preset-core
```

The following example shows detailed output for adjacency SoftSwitch, including the "Register Out Timer:" field that shows the time interval in seconds when the SBC forwards the next REGISTER messages to the softswitch.

```
Router# show sbc mySbc sbe adjacencies SoftSwitch detail
```

```
SBC Service "mySbc"
Adjacency SoftSwitch (SIP)
Status: Attached
```

Signaling address: 100.100.100.100:5060, VRF Admin Signaling-peer: 10.10.51.10:5060 Force next hop: No Account: Group: None In header profile: Default Out header profile: Default In method profile: Default Out method profile: Default In UA option prof: Default Out UA option prof: Default Default In proxy opt prof: Out proxy opt prof: Default Priority set name: None Local-id: None Rewrite REGISTER: Off Target address: None Register Out Timer: 36000 seconds Register Aggregate: Disabled NAT Status: Auto Detect Reg-min-expiry: 30 seconds Enabled Fast-register: Fast-register-int: 30 seconds Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default Force next hop: No Home network Id: None UnEncrypt key data: None SIPI passthrough: No Rewrite from domain: Yes Rewrite to header: Yes Media passthrough: No Preferred transport: UDP Hunting Triggers: Global Triggers Redirect mode: Pass-through Security: Untrusted Outbound-flood-rate: None Ping-enabled: No Signaling Peer Status: Not Tested

The following example enables Aggregate Registration on adjacency Cary-IP-PBX, which has a preset access profile specified because it faces an access device on a UNI network. The last three commands in the configuration, entered in the correct order, enable the aggregate registration call routing to work.

```
sbc mySbc
sbe
adjacency sip Cary-IP-PBX
registration rewrite-register
inherit profile preset-access
registration aggregate
header-name to passthrough
request-line request-uri rewrite
```

The following example displays detailed output for adjacency Cary-IP-PBX, including the "Register Aggregate:" field that shows Aggregate Registration is "Enabled."

Router# show sbc mySbc sbe adjacencies Cary-IP-PBX detail

```
SBC Service "mySBC"
Adjacency Cary-IP-PBX (SIP)
Status: Attached
```

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Signaling address: 100.100.100.100:5060, VRF Admin Signaling-peer: 10.10.51.10:5060 Force next hop: No Account: Group: None In header profile: Default Out header profile: Default In method profile: Default Out method profile: Default In UA option prof: Default Out UA option prof: Default In proxy opt prof: Default Out proxy opt prof: Default Priority set name: None Local-id: None Off Rewrite REGISTER: Target address: None Register Out Timer: 1800 seconds Register Aggregate: Enabled NAT Status: Auto Detect Reg-min-expiry: 30 seconds Enabled Fast-register: Fast-register-int: 30 seconds Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default Force next hop: No Home network Id: None UnEncrypt key data: None SIPI passthrough: No Rewrite from domain: Yes Rewrite to header: Yes Media passthrough: No Preferred transport: UDP Hunting Triggers: Global Triggers Redirect mode: Pass-through Security: Untrusted Outbound-flood-rate: None Ping-enabled: No Signaling Peer Status: Not Tested Rewrite Request-uri: Enabled Registration Monitor: Disabled

Registration Monitoring Examples

The following example shows how monitoring of event subscriptions as a result of registration state changes is enabled:

```
sbc Raleigh-SBC
sbe
adjacency sip Cary-IP-PBX
registration monitor
```

The following example displays detailed output for adjacency Cary-IP-PBX, including the "Registration Monitor:" field that shows Registration Monitoring is "Enabled:"

Router# show sbc mySBC sbe adjacencies Cary-IP-PBX detail

```
SBC Service "mySbc"
Adjacency Cary-IP-PBX (SIP)
```

Status: Signaling address: 100.100.100.100.100. Signaling-peer: 10.10.51.10:5060 100.100.100.100:5060, VRF Admin Account: Group: None In header profile: Default Out header profile: Default In method profile: Default Out method profile: Default In UA option prof: Default Out UA option prof: Default Default In proxy opt prof: Out proxy opt prof: Default Priority set name: None Local-id: None Rewrite REGISTER: Off Target address: None Register Out Timer: Register Aggregate: 1800 seconds Enabled NAT Status: Auto Detect 30 seconds Reg-min-expiry: Fast-register: Enabled Fast-register-int: 30 seconds Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default Force next hop: No Home network Id: None UnEncrypt key data: None SIPI passthrough: No Rewrite from domain: Yes Rewrite to header: Yes Media passthrough: No Preferred transport: UDP Hunting Triggers: Global Triggers Redirect mode: Pass-through Security: Untrusted Outbound-flood-rate: None Ping-enabled: No Signaling Peer Status: Not Tested Rewrite Request-uri: Disabled Registration Monitor: Enabled

Provisional Delegate Registration Examples

The following example configures a delegate registration profile that can be applied to a delegate registration subscriber.

```
sbc mySbc sbe
delegate-profile my-profile
duration 1000
retry-count 5
retry-interval 60
refresh-buffer 200
```

The following example configures a SIP contact for a subscriber, for whom a subscriber detail table exists, and for whom, after the SIP contact is configured, delegate registration can be configured:

sbc mySbc

```
sbe
subscriber sip:bob@isp.example
sip-contact sip:steve@10.1.1.2
adjacency CallMgrB
exit
```

The following example configures a delegate registration for a specified client device address location, after the SIP contact information has been configured:

```
sbc mySbc
sbe
subscriber sip:bob@isp.example
sip-contact sip:steve@10.1.1.2
adjacency CallMgrB ======>> client adjacency
exit
delegate-registration sip:registrar@1.1.1.1
adjacency CallMgrA =====>> registrar adjacency
activate
```

The following show example displays subscribers for which delegate registration have been configured. The **delegate** keyword displays the associated URI contact information for subscribers.

Router# show sbc mySBC sbe sip subscribers delegate

AOR:	<pre>sip:steve1.cisco.com</pre>
Subscriber Location[s]:	<pre>sip:contact@cisco.com -> CallMgrC</pre>
	<pre>sip:contact2@cisco.com -> CallMgrD</pre>
Registrar adj:	CallMgrA
Registrar:	sip:myreg@172.18.52.148
Register Duration:	1800
Register Retries:	3
Retry Interval:	30
Refresh Buffer:	30
Time left:	0 days

The following show example displays subscriber profiles for subscribers for whom delegate registration has been configured.

```
Router# show sbc mySBC sbe sip delegate-profiles
    0
               2
                     34
                           5
                                 6
                                      7
         1
Delegate Profiles:
_____
 Profile
              = steve
 Duration (secs)
              = 1800
 Retry Count
              = 3
 Retry Interval (secs) = 30
 Refresh Buffer (secs) = 30
```

Contact Username Passthrough Examples

The following is an example with a single contact showing that the username part of the contact is passed through unchanged:

```
adjacency sip SIPP1Reg
group SIPP1Reg
inherit profile preset-core
signaling-address ipv4 192.168.101.1
statistics-setting summary
signaling-port 5060 5062
remote-address ipv4 192.168.101.12 255.255.255.255
signaling-peer 192.168.101.12
signaling-peer-port 7068
registration target address 192.168.101.12
registration target port 7069
registration contact username passthrough
attach
```

The following is an example flow of multiple registrations for the same subscriber; the example illustrates how a sequence of REGISTER requests registering multiple contacts behaves. This example assumes all headers are omitted from the requests, apart from Contact headers, and that the registrar-facing adjacency has a signaling-port range from 5060 to 5063 (this means 4 local ports are available).

Step 1 A REGISTER is received registering two contact addresses for the number 5551234.

```
REGISTER sip:5551234@1.2.3.4 SIP/2.0
Contact: <sip:bob@1.1.1.1>
Contact: <sip:robert@1.1.1.1>
```

Step 2 The SBC forwards this REGISTER to the registrar having rewritten the contact address and port.

```
REGISTER sip:5551234@1.2.3.4 SIP/2.0
Contact: <sip:bob@192.168.101.1:5060>
Contact: <sip:robert@192.168.101.1:5061>
```

Step 3 Another REGISTER is received for the number 5551234, registering another endpoint with a duplicate username of "bob."

```
REGISTER sip:5551234@1.2.3.4 SIP/2.0
Contact: <sip:bob@2.2.2.2>
```

Step 4 The SBC forwards this to the registrar, again passing the username through unchanged.

REGISTER sip:5551234@1.2.3.4 SIP/2.0 Contact: <sip:bob@192.168.101.1:5062>

Step 5 A third endpoint registers for the same number. This endpoint provides a very long contact name in the contact field.

```
REGISTER sip:5551234@1.2.3.4 SIP/2.0
Contact: <sip:this_is_an_extremely_long_contact_username@2.2.2.2>
```

Step 6 The SBC forwards this request to the registrar and rewrites the username because it is over the maximum passthrough length (32).

```
REGISTER sip:5551234@1.2.3.4 SIP/2.0
Contact: <sip: 6e83bca53a48bd629a153a93ff8f4af1@192.168.101.1:5063>
```

Alternative Contact Rewriting Example

The following example shows how to configure the Alternative Contact Rewriting feature on the SBC:

```
sbc test
sbe
   adjacency sip core-side-1
    force-signaling-peer
   nat force-off
    inherit profile preset-core
    signaling-address ipv4 9.9.9.1
    remote-address ipv4 10.0.49.78 255.255.255.255
    signaling-peer 10.0.49.78
   registration target address 10.0.49.78
registration target port 5060
registration contact username rewrite userid-and-numeric
attach
   adjacency sip core-side-2
    force-signaling-peer
    nat force-off
   inherit profile preset-core
    signaling-address ipv4 9.9.9.2
   remote-address ipv4 10.0.49.76 255.255.255.255
    signaling-peer 10.0.49.76
    registration target address 10.0.49.76
registration target port 5060
registration contact username rewrite numeric
attach
```

Registering with Softswitch via Cisco SRP Integrated Access Device (IAD) Examples

The following example shows how to register a subscriber with Softswitch, using the Cisco SRP Integrated Access Device (IAD).

This example uses IP realms in the adjacency. See IP Realm Support for information on IP Realms.

```
sbc interop
she
  adjacency sip srp
   nat force-on
   inherit profile preset-access
   preferred-transport udp
   signaling-address ipv4 10.3.127.1
   statistics method summary
   signaling-peer 0.0.0.0
   registration rewrite-register
   realm customer.com
   attach
  adjacency sip meta
   nat force-off
   inherit profile preset-core
   preferred-transport udp
   signaling-address ipv4 99.109.206.106
   statistics method summary
   signaling-peer sig.trav.demo.softswitch.com
   registration target address sig.trav.demo.softswitch.com
   registration target port 5060
```

Γ

```
fast-register disable
   header-name To passthrough
   header-name From passthrough
   realm sig.trav.demo.softswitch.com
   attach
  call-policy-set 1
   first-call-routing-table crtab1
   first-reg-routing-table crtab1
   rtg-src-adjacency-table crtab1
    entry 1
     action complete
     dst-adjacency meta
     match-adjacency srp
    entry 2
     action complete
     dst-adjacency srp
     match-adjacency meta
   complete
  active-call-policy-set 1
!
1
1
media-address ipv4 10.3.127.1 realm customer.com
media-address ipv4 99.109.206.106 realm sig.trav.demo.softswitch.com
activate
!
```



SIP Message Manipulation

You can configure the Cisco Unified Border Element (SP Edition) to selectively examine and manipulate incoming SIP messages on an adjacency.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to *Cisco Unified Border Element* (SP Edition) Command Reference: Unified Model at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification	
Cisco IOS XE Release 2.4	The SIP Header Profile, SIP Method Profile, Parameter Profile, Response Code Mapping, SIP Header Manipulation, and Provisional Response filtering features were introduced on Cisco IOS XR along with support for the unified model.	
Cisco IOS XE Release 2.5	The following features were introduced on the Cisco ASR 1000 Series Routers:	
	• Ability to Insert Firewall Parameter in SIP Contact Header.	
	• Enhanced SIP header manipulation functionality on the Cisco ASR 1000 Series Routers.	
	• P-KT-UE-IP header (type of private header) support as part of SIP header manipulation functionality.	
Cisco IOS XE Release 2.6	The following SIP header manipulation functions were enabled with new CLIs on the Cisco ASR 1000 Series Routers:	
	Parse User Name Parameters	
	Suppress Expires Header	
	Configuring Customer P-Asserted-Identity	
Cisco IOS XE Release 3.1S	The following features were added on the Cisco ASR 1000 Series Routers:	
	SIP Destination ID	
	• SIP Source ID	

Feature History for SIP Header Manipulation on Cisco Unified Border Element (SP Edition)

Cisco IOS XE Release 3.2S	co IOS XE Release 3.2S SBC supports call-policy routing of calls using the hostname in the Request URI. The calls are now routed even in the absence of userna in the Request URI.	
	The Event Header in Publish Method feature was added on the Cisco ASR 1000 Series Routers.	
Cisco IOS XE Release 3.3S	The SIP Message Editing feature was added.	
Cisco IOS XE Release 3.4S	The SDP Editing Using Script-Based Editors feature was added.	

Contents

This chapter contains the following sections:

- SIP Message Editing Using Profiles, page 23-3
- SIP Message Editing Using Editors, page 23-64
- SDP Editing Using Script-Based Editors, page 23-84

SIP Message Editing Using Profiles

This section contains the following information on SIP profiles:

- Information About SIP Profiles, page 23-3
- Method Profiles, page 23-4
- Response Code Mapping, page 23-12
- Header Profiles, page 23-16
- Provisional Response Filtering, page 23-33
- Parameter Profiles, page 23-36
- Ability to Insert Firewall Parameter in the SIP Contact Header, page 23-42
- Configuration Examples for SIP Profiles, page 23-46

Note

From Release 3.3S, the concept of *editors* has been introduced. An editor is the enhanced version of its corresponding profile. From ?\$paranum>SIP Message Editing Using Editors? section on page 23-64, all occurrences of *profile* have been replaced by *editor*. For example, a method profile is called a method editor.

Information About SIP Profiles

Cisco Unified Border Element (SP Edition) can manipulate the following SIP profiles:

- · Method profiles
- Header profiles
- Parameter profiles

Method profiles allow the association of header profiles and parameter profiles to method elements contained in the method profile. You can use actions with method profiles to allow the whitelist to contain blacklisted headers and the blacklist to contain whitelisted headers as well as to reject non-vital methods. This allows any profile to contain mixed actions per-profile.

Header profiles allow complex header manipulation to occur, over and above the existing whitelist and blacklist functionality using actions based on conditional expressions.

Header profiles additionally allow the association of parameter profiles in header elements contained in the profile.

You can use variables to store header content; you can then optionally reconstruct the headers using previously stored variables. You can also match headers based on regular expression matching. You can use conditional matching to match against adjacency settings, transport addresses, and a number of boolean match criteria. You can also use header profiles to reference and make limited modifications to the Request Line.

A header profile can conditionally match any part of a header, but can only replace the entire header. SIP parameter profiles extend this capability to allow changes to be made to individual SIP Request Uniform Resource Identifier (URI) parameters associated with a header.

Parameter profiles allow the removal, replacement, or addition of specific URI parameters within certain vital headers.

You can also associate parameter profiles with methods in method profiles for the purpose of request-line processing per method only.

You can configure multiple store rules, request-lines, and header entries, each with unique actions and/or conditions under which the action is applied. Figure 23-1 shows the hierarchical association of adjacency, method profiles, header profiles, and parameter profiles. The dotted line shows the deprecated method for parameter profile association to method profiles.





Method Profiles

SIP methods can be blacklisted and whitelisted dynamically at run-time during receipt of a message (ingress) and at transmission of a message (egress).

A configured method profile allows two types of method profiles for non-vital requests. These can be blacklist (drop) or whitelist (pass). The whitelist action is considered to be the default type for a method if 'blacklist' is not present in the command line.

The method profile will contain a list of methods which are either passed on (whitelist) or dropped (blacklist). A single profile can then be associated with each of the inbound or outbound call sides.

Method profiles can be associated with pre-defined header profiles. In addition, pre-defined parameter profiles can be associated with the Request-line per method.

Method profiles are not allowed to blacklist or whitelist vital methods; however, header profiles and parameter profiles can be associated with vital methods.

Status code mapping can be associated with any method type declared in a method profile such that any response identified with this method can be changed. For example, a 503 response to an INVITE could potentially be changed to a 500 response if appropriate mapping is declared against the INVITE method.

This section contains the following topics:

- Restrictions for Configuring Method Profiles, page 23-5
- Information About Method Profiles, page 23-5
- Configuring Method Profiles, page 23-7
- Unconfiguring Method Profiles, page 23-9
- Applying Method Profiles, page 23-11

Restrictions for Configuring Method Profiles

Review the following restrictions for method profiles:

- Any given profile must be exclusively a whitelist or a blacklist.
- Two profiles are applied to process any given SIP message: one inbound and, if permitted through that, one outbound.
- Profiles check only SIP methods in the Request Uniform Resource Identifier (URI)
- SIP requests that are blacklisted and non-essential are rejected as a result of a method profile's rules. SIP responses are always forwarded.
- Any method unknown to Cisco Unified Border Element (SP Edition) which is forwarded as a result of a profile's rules does not affect creating or deleting a SIP dialog.
- Methods that are essential to the operation of Cisco Unified Border Element (SP Edition) cannot be blacklisted and are implicitly added to any whitelist.
- Profiles cannot be deleted while they are in active use by at least one adjacency.
- In case of non-Information Management System (IMS) preset, there is a default method profile (sip method-profile default). If configured, the default method profile is attached to the adjacencies for which no explicit user-defined method profiles are configured for both inbound and outbound. The sip method profile default is an empty white-list by itself.

Information About Method Profiles

After you configure a profile, you can assign it for a default application. Any SIP adjacency can apply it to signaling for that adjacency.

Note

Profiles are an optional part of the configuration—they do not have to be specified for Cisco Unified Border Element (SP Edition) to operate correctly. The default behavior is that requests with one of the essential methods are processed, and all other requests are rejected.

You can add or remove methods from profiles at any time. Each method can optionally be assigned one of three actions with the **action** command:

- Either **pass** or **reject** the method.
- Use the **as-profile** action to select the default profile blacklist or whitelist.

Profiles cannot be deleted while at least one adjacency is using them. You can see which adjacencies are using a profile by entering the following show commands:

show sbc sbc-name sbe sip method-profile [profile-name]
or
show sbc sbc-name sbe sip essential-methods

The following methods are part of the essential method set:

- ACK
- BYE
- CANCEL
- INVITE
- NOTIFY
- PRACK
- REFER
- REGISTER
- SUBSCRIBE

To modify parameters in the request-line, associate a parameter profile with a method profile.

Cisco IOS XE Release 2.4 and later contains the following functionalities:

- Predefined header profiles can be associated with outgoing method profiles.
- Predefined parameter profiles can be associated with the request-line per method.



Header profiles and parameter profiles can be associated with essential methods even though method profiles are not allowed to blacklist/whitelist essential methods.

• Response code mapping can be associated with any method type declared in a method profile so that any response identified with the method can be changed. For example, a 503 response to an INVITE could potentially be changed to a 500 response if appropriate mapping is declared against the INVITE method.

Configuring Method Profiles

This procedure shows how to configure method profiles.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. sip method-profile profile-name
- 5. description description
- 6. blacklist
- 7. pass-body
- 8. method name
- 9. action {as-profile | pass | reject}
- 10. end
- **11.** show sbc *sbc-name* sbe sip method-profile [*profile-name*]
- 12. show sbc *sbc-name* sbe sip essential-methods

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the submode for configuring the method profile.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip method-profile profile-name</pre>	Configures a method profile and enters SIP method profile configuration mode.
	Example: Router(config-sbc-sbe)# sip method-profile profile1	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies that do not have a specific profile configured.
Step 5	description description	Adds a description for the specified profile.
	Example: Router(config-sbc-sbe-sip-mth)# description mysbc profile1	The no form of this command removes the description. This description is displayed when the show command is used for this profile and is displayed for each profile when displaying a summary of all profiles.
Step 6	blacklist	Configures a profile to be a blacklist. The no form of this command configures the profile to be a whitelist.
	Example: Router(config-sbc-sbe-sip-mth)# blacklist	Note By default, profiles are whitelists.
Step 7	pass-body	Permits message bodies to be passed through for non-vital methods accepted by this profile.
	Example: Router(config-sbc-sbe-sip-mth)# pass-body	The no form of this command strips the message body out of any non-vital SIP messages matched by this profile.
		Note Non-vital method is same as non-essential method.
Step 8	method name	Adds a method with the specified name to the profile. Enters the SBE method profile element configuration mode.
	Example: Router(config-sbc-sbe-sip-mth)# method test	This field can be 1 to 32 characters (inclusive) in length and is case-insensitive.
		The no form of this command deletes the method with that name from the profile.

	Command or Action	Purpose
Step 9	action {as-profile pass reject}	Specifies the action to be performed on the parameter.
		• as-profile —Drops the method.
	Example:	• pass —Passes the method.
	as-profile	• reject —Rejects the method.
Step 10	end	Exits SBE method profile element configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-mth-ele)# end	
Step 11	<pre>show sbc sbc-name sbe sip method-profile [profile-name]</pre>	Displays details for the method profile with the designated name.
		Use profile-name default to view the default profile.
	Example: Router# show sbc mysbc sbe sip-method-profile profile1	Displays a list of all configured method profiles if no profile-name is specified.
Step 12	show sbc sbc-name sbe sip essential-methods	Displays a list of the essential methods.
	Example: Router# show sbc mysbc sbe sip essential-methods	

Unconfiguring Method Profiles

The following example shows the proper sequence for unconfiguring a method profile applied to an adjacency. References to the profile must first be removed from all adjacencies. In this example, only one adjacency refers to the profile.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. no method-profile inbound profile-name
- 6. exit
- 7. no sip method-profile profile name
- 8. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the submode for configuring the method profile.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipadj1	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	no method-profile inbound profile-name	Unconfigures profile1 that was used for inbound signaling on adjacency test.
	Example: Router(config-sbc-sbe-adj-sip)# no method-profile inbound profile1	
Step 6	exit	Exits SBE SIP adjacency configuration mode and enters SBE configuration mode.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# exit</pre>	
Step 7	no sip method-profile profile name	The no form of this command deletes the method with that name from the profile.
	Example: Router(config-sbc-sbe)# no sip method-profile profile1	
Step 8	end	Exits the SBE mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	

Applying Method Profiles

This procedure shows how to apply method profiles.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. method-profile inbound *profile-name*
- 6. end
- 7. **show sbc** *sbc-name* **sbe sip method-profile** *name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip test	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	method-profile inbound profile-name	Sets profile1 to be used for inbound signaling on adjacency test.
	Example: Router(config-sbc-sbe-adj-sip)# method-profile inbound profile1	Note When attaching a method profile to an adjacency, the adjacency must be in the "no attach" state.

	Command or Action	Purpose
Step 6	end	Exits the header profile mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 7	show sbc <i>sbc-name</i> sbe sip method-profile <i>name</i>	Displays the header profile information.
	Example:	
	Router# show sbc mysbc sbe sip method-profile one	

Response Code Mapping

Response code mapping provides an ability to manipulate the SIP response codes when the messages traverse the Cisco Unified Border Element (SP Edition). The mapping table is applied to inbound messages received at a SIP adjacency or to responses sent out of a SIP adjacency. The mapping is user-configurable on a per SIP method basis so that each SIP method can be mapped differently. lists the mapping limitations on SIP response code.

Response Codes	Mapping
100	No mapping allowed
1xx	Maps to 1yy (not 100)
2xx	Maps to 2yy
3xx	Maps to 3yy
4xx	Maps to 4yy, 5yy, or 6yy
5xx	Maps to 4yy, 5yy, or 6yy
бхх	Maps to 4yy, 5yy, or 6yy

Response code mapping allows you to:

- Map a particular response code to a specific response code. For example, you can map 401 to 400, but not to 300. You can map 102 to 101, but not 100.
- Map a group of response codes (defined using a wildcard) to a specific response code. For example, you can map 40X to 400, or map all of 4XX to 400.
- Specify exceptions to the wildcard. For example, mapping 2XX to 201, and mapping 200 to 200.

You can use the **map-status-code** command to add one of more mappings.

Where configuration causes the response code to be mapped to one that is not defined in RFC 3261, Cisco Unified Border Element (SP Edition) applies the reason phrase "Unrecognized status code."

This section contains the following topics:

- Restrictions for Response Code Mapping, page 23-13
- Configuring Response Code Mapping, page 23-13
- Applying Response Code Mapping, page 23-15
Restrictions for Response Code Mapping

The following restrictions apply to Response Code Mapping:

- Response code mapping only covers mapping of SIP response codes. H.323 calls cannot have their response codes mapped.
- Certain messages are processed only by the SIP Transaction Manager; mapping of these messages is not possible. For example, badly formatted messages that cannot be interpreted are responded to directly by the SIP Transaction Manager.
- There is no provision for the mapping of SIP reason phrases. The reason phrase will always match the reason code as defined in RFC 3261. A generic reason phrase is applied when the requested reason code has no corresponding definition in RFC 3261. This phrase is a compile time constant.
- Changing the response code could result in an invalid message (for example, mapping the response code could produce a message with mandatory headers missing). There is no provision to ensure that messages contain headers required by the new response code.
- A maximum of 128 mappings is permitted in each direction per adjacency (128 inbound and 128 outbound mappings).

Configuring Response Code Mapping

This procedure shows how to configure response code mapping.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip method-profile *profile-name*
- 5. method name
- 6. map-status-code
- 7. range statuscoderange value statuscodevalue
- 8. end
- 9. show sbc *sbc-name* sbe sip method-profile [*profile-name*]
- **10.** show sbc *sbc-name* sbe sip essential-methods

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the submode for configuring the method profile.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip method-profile profile-name</pre>	Configures a method profile.
	Example: Router(config-sbc-sbe)# sip method-profile profile1	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies that do not have a specific profile configured.
Step 5	method name	Adds a method with the specified name to the profile.
	Example:	This field can be 1 to 32 characters (inclusive) in length and is case-insensitive.
	Router(config-sbc-sbe-sip-mth)# method test	The no form of this command deletes the method with that name from the profile.
Step 6	map-status-code	Enters the SIP method profile element configuration mode.
	Example: Router(config-sbc-sbe-sip-mth-ele)# map-status-code	
Step 7	range statuscoderange value statuscodevalue	Maps a range of response codes to a response code.
	Example: Router(config-sbc-sbe-sip-mth-ele-map)# range 5XX value 500	
Step 8	end	Exits the method profile mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-mth-prf)# end	

	Command or Action	Purpose
Step 9	<pre>show sbc sbc-name sbe sip method-profile [profile-name]</pre>	Displays details for the method profile with the designated name.
	Example: Router# show sbc mysbc sbe sip-method-profile profile1	Use profile- <i>name</i> default to view the default profile. Displays a list of all configured method profiles if no profile-name is specified.
Step 10	show sbc sbc-name sbe sip essential-methods	Displays a list of the essential methods.
	Example: Router# show sbc mysbc sbe sip essential-methods	

Applying Response Code Mapping

Apply response code mapping by associating it with an adjacency.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. method-profile inbound profile-name
- 6. end
- 7. show sbc *sbc-name* sbe sip method-profile *name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip test	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	method-profile inbound profile-name	Sets profile1 to be used for inbound signaling on adjacency test.
	Example: Router(config-sbc-sbe-adj-sip)# method-profile inbound profile1	Note When attaching a method profile to an adjacency, the adjacency must be in the "no attach" state.
Step 6	end	Exits the header profile mode and returns to Privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-adj-sip)# end	
Step 7	show sbc <i>sbc-name</i> sbe sip method-profile <i>name</i>	Displays the header profile information.
	Example: Router# show sbc mysbc sbe sip method-profile one	

Header Profiles

Header profiles processing occurs in a two-stage process. In the first stage, the following steps occur:

- 1. Select next header from the message.
- 2. Look through the header profile for rules affecting the selected header.
- 3. In configured order, try to apply each rule to the header.
- 4. If the action is to add a header, then ignore this rule and move on to the next.
- 5. If the match condition is FALSE then move onto the next rule, do not evaluate any parameter profile.
- **6.** Apply the action or parameter profile described in the element. If this is to remove the header, then move on to the next header in the message.

The second stage adds new headers to the message. Because it occurs after the first stage, there is a well-defined group of headers in the message. The steps are:

- 1. Take each rule that adds a header to the message.
- **2.** If the action is to add the first instance of the header only and there is already a header with that name in the message, then move onto the next addition rule.



- **Note** If another action has replaced the name of header then it is the replaced name that is used to test whether a new header should be added. That is, any header-name replacements performed in stage 1 are used in this stage of header-name comparisons, and not the original header-names from the arriving message.
- 3. Add the header if the match condition evaluates to TRUE.
- **4.** Apply any rules defined for that header in user-configured order with this name. Only apply rules that are ordered after the add header rule, if the header was added.

This section contains the following topics:

- Restrictions for Configuring Header Profiles, page 23-17
- Information About Header Profiles, page 23-17
- Header Manipulation, page 23-18
- Header Profile Configuration Information, page 23-25
- Configuring Header Profiles, page 23-25
- Applying Header Profiles, page 23-27

Restrictions for Configuring Header Profiles

Review the following restrictions for header profiles:

- Any given profile must be exclusively a whitelist or a blacklist.
- Two profiles are applied to process any given SIP message: one inbound and, if permitted through that, one outbound.
- SIP headers that are essential to the operation of Cisco Unified Border Element (SP Edition) cannot be blacklisted and are implicitly added to any whitelist.
- Profiles can not be removed while they are in active use by an adjacency.
- For provisional filtering, provisional responses may not be blocked where the sender has required reliable provisional responses (SIP 100rel). This is to ensure that Cisco Unified Border Element (SP Edition) does not interfere with the call setup (as per RFC3262) by dropping the provisional response.
- Header profile conditional matching can be performed against any part of the message. The matches can be exact matches or even sub-strings of any given field.
- The conditions may be associated with a specific header referenced by the header profile header definition, but can also reference other non-vital parts of the message in order to evaluate the conditional expression; thus the condition could be associated with header P-Asserted-Identity while checking against the contents of the Call-Info header.

Information About Header Profiles

After you configure a profile, you can assign it for a default application. Any SIP adjacency can apply it to signaling for that adjacency.

You can add or remove headers from profiles at any time. Headers configured on a profile must contain characters that are valid for a SIP header.

Profiles cannot be deleted while any adjacency is using them. You can see which adjacencies are using a profile by entering the following show command:

show sbc sbc-name sbe sip method-profile [profile-name]
or

show sbc sbc-name sbe sip essential-methods

The following are the essential SIP headers, which must not be configured on any profile:

- Allow
- Authorization
- Call-ID

- Contact
- Content-Length
- Content-Type
- CSeq
- Event
- Expires
- From
- Max-Forwards
- Min-Expires
- Proxy-Authenticate
- Proxy-Authorization
- Proxy-Require
- Record-Route
- Referred-By
- Referred-To
- Replaces
- Require
- Route
- Subscription-State
- Supported
- To
- Via
- WWW-Authenticate

Note

Profiles are an optional part of the configuration. If no profile is applicable to a given SIP signal, then the essential headers are processed and all other headers are not forwarded.

Header Manipulation

You can modify non-essential headers in SIP messages using header and parameter profiles. The following information summarizes the supported actions:

- Pass the header unchanged (whitelist functionality).
- Conditionally pass the header unchanged.
- Remove the header (blacklist functionality).
- Conditionally remove the header.
- Replace the name of the header. The replacement name cannot be that of a vital header.
- Conditionally replace the header content (appearing after the ":").
- Add a new instance of a header to a message regardless of whether or not the header already exists.
- Add the first instance of the header to the message, if a header with this name does not already exist.

- A combination of the above actions can be specified as a set or group of actions to be performed within a profile.
- The header profiles can be used in method profiles to allow header actions only associated with specific requests types.
- Parameter profiles can be associated with headers in header profiles.
- Header content can be stored in variables and later expanded during replace-value actions.
- Privacy headers are treated as unknown headers, which by default would be blacklisted (stripped). However, the SBC can be configured to pass through SIP Privacy headers.
- Regular expression matching can be performed on headers.

You can match against any part of a header but only replace the entire header. A parameter profile extends this capability to change individual SIP URI parameters associated with a header. Header profiles can only modify non-vital header information. To display the vital header information, use the **show sbc test sbe sip essential-method**, **show sbc test sbe sip essential-headers**, or **show sbc test sbe sip essential-parameters** commands.

Parameter profiles can be specified to match the following parts of the message.

- Request URI
- To
- From
- Contact

To modify the parameters in the Request-line, associate a parameter profile with a method profile. To modify the parameters in the Contact, To, or From headers, associate a parameter profile in the header profile.

Event Header in Publish Method

As per RFC3903, the SIP PUBLISH request must contain an Event header. In releases earlier than Cisco IOS XE Release 3.2S, the SBC could pass through the PUBLISH method using the existing message manipulation framework, but could not pass through the Event header. The effect of this was that attempts to use the PUBLISH services (containing an Event header) through the SBC were blocked

From Cisco IOS XE Release 3.2S, the SBC can pass through the PUBLISH method containing Event header using the existing message manipulation framework. Preset header manipulations accessed by inherit-profiles are modified to pass-on the Event header.

The Event Header in Publish Method feature does not affect the behaviors for SUBSCRIBE, REFER, and NOTIFY methods. Event headers are passed through unchanged. For all the other methods, the Event header is treated generically.

Header Profile Conditional Matching

To allow header manipulation, a set of conditions can be specified in order to dictate the rules under which the header actions will be applied. Conditional matching allows comparisons to be performed against any part of the message. The matches can be exact matches or even sub-strings of any given field.

The conditions can be associated with a specific header referenced by the header profile header definition, but equally can also reference other non-vital parts of the message in order to evaluate the conditional expression.

Note

Absence of a condition (conditional expression) implies the condition for the action is always true.

Each condition represents a part of the message to be manipulated, and the operation to be performed. A condition can be defined in the following ways:

condition comparison-type operator comparison-value

or

condition boolean-operator operator {true | false}

```
Example:
condition header-value contains "Cisco"
condition is-request eq true
```

Table 23-1 lists the comparison types.

Comparison Type	Description
status-code	Response code value
header-value	Current header content
header-name name header-value	Content of a different header
variables	Match on variable content
adjacency	Match on adjacency settings
transport	Match on transport addresses or ports
header-uri	Match on parts of the URI (username)
request-uri	Match on parts of the request-URI (username)
word	Match on static strings

Table 23-1 Comparison Types

Table 23-2 lists the operators.

Table 23-2 Operators

Operator	Description
[not] eq	Equals or not equal
[not] contains	Contains or does not contain
[not] regex-match	Regular expression matching (BRE)
store-as	Store rules only

Table 23-3 lists the boolean operators.

Table 23-3 Boolean Operators

Boolean Operator	Description
is-sip-uri	Does the header contain a sip: URI
is-tel-uri	Does the header contain a tel: URI

is-request	Is the message a request
is-100rel-required	Is the call performing 100rel
is-defined	Test if a variable is defined

Table 23-3 Boolean Operators

The following restrictions apply for conditional matching:

- Multiple conditional expressions against the same header can be added each containing unique actions and conditions to build complex manipulations
- Each condition must be entered one at a time. To add a subsequent condition to an existing condition, the condition must begin with "and" or "or". If the condition does not contain "and" or "or", it effectively overwrites any conditions already defined.
- If no profile-type is explicitly expressed in the header profile command line definition then the assumed header profile type will be "whitelist".
- Multiple headers of the same type can be declared in any one profile defining either different action types or conditions.
- Character "*" can be used as a wildcard header, although only one wildcard header entry can be configured per profile.
- Duplicate header names with differing actions or conditions can be identified with the "entry <integer>" parameter in the command line. This can be used for the purposes of editing or deletion of a specific action related to a header. If no "entry" in the command line then it is assumed that the first entry related to the header of this header type is being configured.

Store Rules Declaration

The data extracted from headers can be stored into variables. The store rules are defined which are executed prior to any header element actions. Store rules are specialized header elements of the format:

```
Store-Rule:<entry>
```

The store rules contain conditions which allow storage in one of the following two ways:

1. A condition can contain a "store-as" keyword to directly store a string or complete header value into a variable.

condition comparison-type store-as variable-name

Example:

condition header-value store-as var1

The content of header-value will be stored into var1.

2. A regular expression can be applied to a header using keyword "regex-match". If the regular expression contains one or more (up to five max) sets of escaped parentheses '\(\)' around specific parts of the regular expression, then if the regular expression successfully matches, the values of each parts of the match grouped by the parentheses are extracted and stored into variables defined in the regex-match keyword arguments.

condition *comparison-type* **regex-match** [**store-as** *variable-name....(up to 5)*]

Example:

```
condition header-name P-Asserted-Identiy header-value regex-match
sip:\(.*\)@[Cc]isco.com store-as var1
```

For the complete list of comparison types, operators, and boolean operators, refer Table 23-1, Table 23-2, and Table 23-3.

Extracted variables can later be used in the actions which require values such as replace-value, add-first-header/add-header. Variables are expanded by use of "\${var}" format within the replacement string.

Request Line Modification

You can perform limited modification to the request-line with action replace-value in header profiles.

The use of the request-line forming part of the header profiles is the preferred method for changes (including parameter profiles) to the request-line.

The format of the value used in action replace-value is:

sip:user@host[:port]

The variables that are already extracted to the store rules can be used in the construction of the Request Line.

Example:

"sip:\${user}@\${host}"

Request-line is a specialized header element of the format:

Request-URI:<entry>

Note

Changes to the request-line must meet the SIP RFC 3261 formatting rules, and any host declared in the replacement must be a valid host to the SBC. User configuration cannot pre-screen the configured changes due to the possibility of variables being present in the configured replacement value. It is only at run-time when the actual request-line can be determined, and errors in request-line construction can result in call failures. Extreme care must be taken when using this feature to prevent call failures.

Parse User Name Parameters

You can configure the SBC to search and parse SIP and SIPS URIs for user name parameters in messages received on an adjacency. If the SIP and SIPS URIs contain any user name parameters, those parameters are treated as regular URI parameters. This is applicable to SIP and SIPS URIs within the Request URI, and the To and From headers for INVITE requests and out-of-dialog requests.

The following is an example of a URI with a username parameter:

"sip:username;cic=1234@host.com;user=phone". Here, 'cic=1234' is treated as a URI parameter, such as 'user=phone', and the username is taken to be 'username', instead of 'username;cic=1234'

Use the command **uri username parameters parse** to enable parsing.

Suppress Expires Header

You can configure the SBC to suppress the Expires Header in the outgoing INVITE requests. Use the command **header-name expires suppress** to remove the Expires Header.

Configuring Customer P-Asserted-Identity

You can configure the SBC to specify a value for the P-Asserted-Identity on the outgoing SIP message. The header is added to all requests and responses except ACK, CANCEL, INFO, PRACK, REGISTER and UPDATE.

Use the **header-name p-asserted-id [header-value [***header-value***]** | **assert]** command to specify a value for the P-Asserted-Identity.

SIP Destination ID



This feature is applicable only to the INVITE and non-REGISTER out-of-dialogue requests.

When routing a call, the destination address or called party identity is typically derived from the Request URI. However, there are other headers where this information could potentially be derived from, such as To: or P-Called-Party-ID.

You can define an ordered list of headers that can be used to derive the called party address. The headers can include any non-essential SIP header, or To:, and Request URI. A maximum of ten headers can be configured in a header list. The header with priority 1 is analyzed first, the header with priority 2 is analyzed next, and the header with priority 10 is analyzed last.

The following sections describe how this feature works on incoming and outgoing requests.

Incoming Requests

For incoming requests:

- By default, the SBC extracts the called party identity from either the P-Called-Party-ID: header or from the Request URI.
- If the SBC finds multiple instances of a given header in a received SIP message, the first instance is used for called party identity extraction. If the SBC encounters a syntax error while extracting the identity, the SBC creates a log, and moves to the next header in the priority list.
- If a header is not present in the SIP request, or if a header in the header list contains a SIP URI without a username, the SBC moves to the next header in the header list.
- After all headers have been tried without success, the SBC extracts the called party identity from the Request URI.
- The header list may include the Request URI to enable the SBC to look for the called party identity from the Request URI when it gets to a point where the Request URI is prioritized in the list. If the list contains only the Request URI, the SBC looks at only the Request URI.

Outgoing Requests

By default, the SBC reinserts both the domain and the username from the called party identity back into the SIP header from which the identifier originally came on the inbound side.

Outgoing Request URI:

• If the called party identity was originally extracted from the Request URI, the Request URI is reconstructed using the called party identity.

• If the called party identity was originally extracted from another header, the username and domain in the Request URI from the SIP message received are preserved. This is done before any SIP header filtering or other editing function (for example, IP/FQDN URI translation) is applied to the Request URI.

Outgoing To Header or Passed Through Arbitrary Header:

- If the called party identity was originally extracted from a header (rather than the Request URI) and that header has been passed through using the inbound adjacency's header manipulation functionality, the SBC inserts the domain and username back into the header, thereby preserving the scheme, URI parameters, and header parameters that were in the original message. Failures due to corruption of header because of the inbound header filtering configuration are logged by the SBC, but other failures are ignored.
- The called party identity may have been edited by the SBC (for example, as part of Number Manipulation) before being reinserted into the outgoing message. This is done only for the first instance of the header in the outbound SIP request before any outbound header filtering or any other editing is applied to the header. There is no restriction on header filtering. You may configure the header editing rules that may subsequently remove or change the header containing the called party identity.
- We recommend that you configure action pass on the inbound header filter profile for all the headers specified in the header list. These headers can then be filtered by the outbound header filter profile.

To configure the destination address header list, use the **dst-address** and **header-priority** commands.

See the ?\$paranum>Configuring an Ordered List of Headers for Deriving the SIP Destination Address? section on page 23-28 for details on configuring header-priority for deriving SIP source ID.

The SBC can be configured to perform conditional matching based on these derived values. See the ?\$paranum>Header Profile Conditional Matching? section on page 23-19 for more details.

SIP Source ID

When routing a call, the source number can be analyzed and modified using a call policy. The source address is typically derived from the From: header. There are, however, other headers from where this information could potentially be derived from, such as P-Preferred-Identity, P-Asserted-Identity, Remote-Party-ID.

You can define an ordered list of headers that can be used to derive the called party address. The headers can include any non-essential sip header and the From header. The SIP Source ID feature also enables you to derive the source number from an ordered set of headers for the calls that were either redirected or diverted. A maximum of ten headers can be configured in the header list. The header with priority 1 is analyzed first, header with priority 2 is analyzed next and the header with priority 10 is analyzed last.

To configure the source address header list, use the **src-address** command and the **header-priority** command.

See the ?\$paranum>Configuring an Ordered List of Headers for Deriving SIP Source Address? section on page 23-30 for details on configuring header-priority for deriving SIP source ID.

SIP Source ID for Diverted Calls

For diverted calls, you can use the address of the party that diverted the call to derive the source address for source analysis. All the diverted calls contain a Diversion: header that contains the details of the party that diverted the call. The SBC can be configured to enter a list of headers for the diverted calls, from which the source number can be derived.

For Cisco IOS XE Release 3.1.0S, this list can only contain one Diversion: header.

To configure the source address header list, use the **div-address** command and the **header-priority** command.

See the ?\$paranum>Configuring an Ordered List of Headers for Deriving SIP Source Address of Diverted Calls? section on page 23-31 section for details on configuring header-priority for deriving SIP source ID for diverted calls.

The SBC can be configured to perform conditional matching based on these derived values. See the ?\$paranum>Header Profile Conditional Matching? section on page 23-19 for more details on conditional matching.

Header Profile Configuration Information

Consideration needs to be given as to the effect of an action or set of actions in conjunction with the default profile behavior (whitelist/blacklist).

An empty blacklist will effectively try to pass on any non-vital header.

An empty whitelist will effectively drop all non-vital headers.

The behavior becomes more complex when conditions are associated with headers.

It is important to consider what actions are defined on the in-bound side. If an empty whitelist header profile is associated with the in-bound side, then no non-vital headers will be visible at all to the outbound side, and therefore, actions applied to the out-bound sides profile may appear not to work. You may need to consider adding actions to 'pass' a specific header on the in-bound side by adding the header to a whitelist (with action as-profile or pass) or adding the header with action 'pass' in a blacklist.

For example, if a header profile is defined as a whitelist (default behavior), and a header action to modify the header-value is inserted with a condition, then the action will be processed if the condition is TRUE and the header modified, but will be ignored if the condition is FALSE.

Because the header is inserted into the whitelist it might well be assumed that it would be passed on unmodified if the condition is FALSE, however, if the condition is FALSE, the action (entry) is ignored, and therefore it is as if the header is not present in the whitelist so the header will not be passed on.

To overcome this, a second entry with action 'pass' can be entered; thus if the headers condition is TRUE, the content with be modified, but if the condition is false, it will be ignored and continue to process any other entries. The second entry has an action 'pass' and will cause the header to be passed on.

Configuring Header Profiles

This procedure shows how to configure header profiles.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile profile-name
- 5. blacklist
- 6. description text

L

- 7. header name [entry number]
- 8. action {add-first-header | add-header | as-profile | drop-msg | pass | replace-name | replace-value | strip }
- **9.** condition [comparison-type | boolean-operator | operator | comparison-value]
- 10. end
- **11. show sbc** *sbc-name* **sbe sip header-profile** [*profile-name*]
- 12. show sbc *sbc-name* sbe sip essential-headers

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Evampla	
	Router# configure terminal	
Step 2	sbc sbc-name	Enters the submode for configuring the header profile.
		Use the <i>sbc-name</i> argument to define the name of the
	Example:	service.
	Router(config) # sbc mysbc	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example:	
	Router(config-sbc)# sbe	
Step 4	<pre>sip header-profile profile-name</pre>	Configures a header profile.
	Example: Router(config-sbc-sbe)# sip header-profile profile1	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies which do not have a specific profile configured.
Step 5	blacklist	Configures a profile to be a blacklist.
	Example:	The no form of this command configures the profile to be a whitelist.
	Router(config-sbc-sbe-sip-hdr)# blacklist	Note By default, profiles are whitelists.
Step 6	description text	Adds a description for the specified profile.
		The no form of this command removes the description.
	Example: Router(config-sbc-sbe-sip-hdr)# description blacklist profile	This description is displayed when the show command is used for this profile and is displayed for each profile when displaying a summary of all profiles.
Step 7	header name [entry number]	header <i>name</i> —Configures the SIP header that will be modified. Enters SBC SBE SIP-HDR-ELE configuration mode.
	Example: Router(config-sbc-sbe-sip-hdr)# header Organization entry 1	entry number—Specifies which action entry to work on.

	Command or Action	Purpose
Step 8	action {add-first-header add-header	Specifies the type of action to be applied to the header.
	as-profile drop-msg pass replace-name replace-value strip}	In the example, the action specified is to conditionally replace the header content with a replace value of
	Example: Router(config-sbc-sbe-sip-hdr-ele)# action replace-value XYZcompany	XYZcompany.
Step 9	condition [comparison-type boolean-operator operator comparison-value]	Specifies the condition to match before taking an action to a SIP message profile. If the condition is met, the action specified in step 8 is performed.
	Example:	Enters SIP header profile configuration mode.
	Router (config-sbc-sbe-sip-hdr-ele-act)# condition header-value ABCcompany	In the example, the value of the <i>condition header-value</i> is ABCcompany, which is matched and thus the value ABCcompany is replaced with XYZcompany.
Step 10	end	Exits the SBC SBE SIP-HDR-ELE configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# end	
Step 11	<pre>show sbc sbc-name sbe sip header-profile [profile-name]</pre>	Displays details for the header profile with the designated name.
		Use the profile-name default to view the default profile.
	Example: Router# show sbc mysbc sbe sip header-profile profile1	Displays a list of all configured method profiles if no profile-name is specified.
Step 12	show sbc <i>sbc-name</i> sbe sip essential-headers	Displays a list of the essential headers.
	Example: Router# show sbc mysbc sbe sip essential-headers	

Applying Header Profiles

This procedure shows how to apply header profiles.

SUMMARY STEPS

- 1. configure
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. header-profile inbound profile-name
- 6. end
- 7. show sbc sbc-name sbe sip header-profile name

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	header-profile inbound profile-name	Sets the inbound header profile to be used for inbound signaling on adjacency sipGW.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# header-profile inbound profile1</pre>	Note When attaching a header profile to an adjacency, the adjacency must be in the "no attach" state.
Step 6	end	Exits the SBE SIP adjacency mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 7	show sbc <i>sbc-name</i> sbe sip header-profile <i>name</i>	Displays the header profile information.
	Example: Router# show sbc sbc-name sbe sip header-profile name	

Configuring an Ordered List of Headers for Deriving the SIP Destination Address

This task configures a list of headers for deriving SIP destination address.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile profile-id
- 5. dst-address

- 6. header-prio 1 header-name header-name
- 7. header-prio 2 header-name header-name
- 8. header-prio 3 header-name header-name
- 9. end
- **10. show sbc** *sbc-name* **sbe sip header-profile** *profile-id*

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Enables entry into the mode of an SBC service.
		Use the <i>sbc-name</i> argument to define the name of the SBC.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enables entry into the mode of an SBE entity within an SBC service.
	Example:	
	Router(config-sbc)# sbc mySbc sbe	
Step 4	sip header-profile	Creates the SIP header profile.
	Example: Router(config-sbc-sbe)# sip header-profile Hprof1	
Step 5	dst-address	Enables entry into the mode to configure destination address.
	Example: Router(config-sbc-sbe-sip-hdr)# dst-address	
Step 6	header-prio 1 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example:	
	Router(config-sbc-sbe-sip-hdr-dst)# header-prio 1 header-name P-Called-Party-ID	
Step 7	header-prio 2 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example: Router(config-sbc-sbe-sip-hdr-dst)# header-prio 2 header-name To	
Step 8	header-prio 3 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example: Router(config-sbc-sbe-sip-hdr-dst)# header-prio 3 header-name Request-uri	

	Command or Action	Purpose
Step 9	end	Enables exit from the destination address configuration mode, and return to the privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-sip-hdr-dst)# end	
Step 10	<pre>show sbc sbc-name sbe sip header-profile profile-id</pre>	Shows the configuration details of the header profile.
	Example:	
	Router# show sbc mySbc sbe sip header-profile Hprof1	

Configuring an Ordered List of Headers for Deriving SIP Source Address

This task configures a list of headers for deriving SIP source address.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile profile-id
- 5. src-address
- 6. header-prio 1 header-name header-name
- 7. header-prio 2 header-name header-name
- 8. header-prio 3 header-name header-name
- 9. end
- 10. show sbc *sbc-name* sbe sip header-profile *profile-id*

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Enables entry into the mode of an SBC service. Use the <i>sbc-name</i> argument to define the name of the SBC.
	Example: Router(config)# sbc mySbc	

	Command or Action	Purpose
Step 3	sbe	Enables entry into the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe mySbc sbe	
Step 4	sip header-profile	Creates SIP header profile.
	Example: Router(config-sbc-sbe)# sip header-profile Hprof1	
Step 5	src-address	Enables entry into the mode to configure source address.
	Example: Router(config-sbc-sbe-sip-hdr)# src-address	
Step 6	header-prio 1 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example: Router(config-sbc-sbe-sip-hdr-src)# header-prio 1 header-name P-Asserted-Identity	
Step 7	header-prio 2 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example: Router(config-sbc-sbe-sip-hdr-src)# header-prio 2 header-name P-Preferred-Identity	
Step 8	header-prio 3 header-name header-name	Configures the header priority, and specifies the header to be used.
	<pre>Example: Router(config-sbc-sbe-sip-hdr-src)# header-prio 3 header-name From</pre>	
Step 9	end	Enables exit from the source address configuration mode and return to the privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-hdr-src)# end	
Step 10	<pre>show sbc sbc-name sbe sip header-profile profile-id</pre>	Shows the configuration details of the header profile.
	Example: Router# show sbc mySbc sbe sip header-profile Hprof1	

Configuring an Ordered List of Headers for Deriving SIP Source Address of Diverted Calls

This task configures a list of headers for deriving SIP source address of diverted calls.

SUMMARY STEPS

1. configure terminal

- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile profile-id
- 5. div-address
- 6. header-prio 1 header-name header-name
- 7. end
- 8. show sbc *sbc-name* sbe sip header-profile *profile-id*

	Command or Action	Purpose
Step 1	configure	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc sbc-name	Enables entry into the mode of an SBC service.
		Use the <i>sbc-name</i> argument to define the name of the sbc.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enables entry into the mode of an SBE entity within an SBC service.
	Example:	
	Router(config-sbc)# sbe mySbc sbe	
Step 4	sip header-profile	Creates SIP header profile.
	Example: Router(config-sbc-sbe)#sip header-profile Hprof1	
Step 5	div-address	Enables entry into the mode to configure source address for diverted calls.
	Example: Router(config-sbc-sbe-sip-hdr)# div-address	
Step 6	header-prio 1 header-name header-name	Configures the header priority, and specifies the header to be used.
	Example: Router(config-sbc-sbe-sip-hdr-src-div)# header-prio 1 header-name Diversion	

	Command or Action	Purpose
Step 7	end	Enables exit from the source address configuration mode and return to privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-sip-hdr-src)# end	
Step 8	<pre>show sbc sbc-name sbe sip header-profile profile-id</pre>	Shows the configuration details of the header profile.
	Example: Router# show sbc mySbc sbe sip header-profile	
	Hprof1	

Following is an example for the **show** command output after the header list—for destination address, source address, and diversion address—is configured on SBC:

```
ASR-1002#show sbc mine sbe sip header-profile Hprof1
  Header profile "Hprof1"
   Description:
                Whitelist
    Type:
    dst-address: (inbound only)
       header-prio 1 header-name P-Called-ID
       header-prio 1 header-name To
       header-prio 1 header-name Request-uri
    src-address: (inbound only)
        header-prio 1 header-name Remote-Party-ID
        header-prio 2 header-name P-Preferred-Identity
       header-prio 3 header-name From
   div-address (inbound only)
       header-prio 1 Diversion
   store-rules:
       No store-rule entries found.
    request-line:
       No request-line entries found.
   headers:
      test
        entry 1
         description:
         action add-first-header value "cisco"
           condition is-request eq true
   Not in use with any adjacencies
   Not in use with any method-profile
ASR-1002#
```

Provisional Response Filtering

Provisional response filtering makes it possible to block 1XX responses (except 100) sent by endpoints. When configuring provisional response filtering, keep the following in mind:

- Provisional responses may not be blocked where the sender has required reliable provisional responses (SIP 100rel).
- Dropping responses where 100_rel is required is not recommended. It may prevent call setup since RFC3262 states subsequent responses should not be sent.

Note

A call attempted with the "Required: 100Rel" header in the INVITE will fail when the adjacency is configured with a header profile to drop 183 messages.

This section contains the following topics:

- Provisional Response Filtering Information, page 23-34
- Configuring Provisional Response Filtering, page 23-34
- Applying Provisional Response Filtering, page 23-35

Provisional Response Filtering Information

Provisional response filtering is achieved by the use of the **action drop-msg** command. The action must be associated with the wildcard header action *. A condition should be added to match on the specific response code that must be dropped.

۵, Note

The header action * can only be used one time in a profile.

Configuring Provisional Response Filtering

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile profile-name
- 5. header *
- 6. action drop-msg
- 7. condition status-code
- 8. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure	
Step 2	sbc <i>sbc</i> - <i>name</i>	Enters the submode for configuring the header profile.
	Example: Router(config)# sbc mysbc	service.

	Command or Action	Purpose
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip header-profile profile-name</pre>	Configures a header profile.
	Example: Router(config-sbc-sbe)# sip header-profile profile1	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies which do not have a specific profile configured.
Step 5	header *	Configures a profile to be a blacklist.
	Example: Router(config-sbc-sbe-sip-hdr)# header *	The no form of this command configures the profile to be a whitelist.Note By default, profiles are whitelists.
		Note In order to filter provisional responses always use the asterisk (*) as the header name with the header command as shown in the command example.
Step 6	action drop-msg	Configures the action to take on an element type in a header.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# action drop-msg	
Step 7	condition status-code	Specifies a condition to match before taking an action to a SIP message profile.
	Example: Router(config-sbc-sbe-sip-hdr-ele-act)# condition status-code eq 183	
Step 8	end	Returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-hdr-ele-act)# end	

Applying Provisional Response Filtering

This procedure shows how to apply provisional response filtering.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. header-profile inbound profile-name
- 6. end
- 7. show sbc *sbc-name* sbe sip header-profile *name*

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	header-profile inbound profile-name	Sets the inbound header profile.
	Example: Router(config-sbc-sbe-adj-sip)# header-profile inbound profile1	
Step 6	end	Exits the SBE SIP adjacency mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 7	show sbc sbc-name sbe sip header-profile name	Shows details of the specified SIP header profile.
	Example: Router# show sbc MySbc sbe sip header-profile profile1	

Parameter Profiles

Parameter profiles allow you to specify specific URI parameter names and allow the removal, replacement, or the addition of specific non-vital URI parameters within certain headers.

The header profile allows potential conditional matching against SIP URI parameters forming part of a limited set of headers. It only allows complete replacement of the header and or content.

The parameter profile will allow actions to be performed only on the SIP URI parameters and not header parameters

This section contains the following topics:

- Restrictions for Configuring Parameter Profiles, page 23-37
- Information About Parameter Profiles, page 23-37
- Configuring Parameter Profiles, page 23-38
- Applying a Parameter Profile to a Header Profile, page 23-39
- Associating with an Adjacency, page 23-41

Restrictions for Configuring Parameter Profiles

Review the following restrictions for parameter profiles:

- A parameter profile is only permitted to act on parameters associated with SIP URIs and not header parameters.
- To prevent call processing failures, actions cannot be performed against vital (essential) parameters.
- Parameter profiles work only on the outbound side.
- Some of the existing adjacency settings may impact the way parameter actions are affected. For example, consider the adjacency setting Rewrite to Header is set as follows:

```
sbc test
   sbe
      adjacency sip <adj name>
      passthrough [to/from]
```

This setting can cause the To: and or From: headers to be passed from inbound to outbound side.

The default setting on an adjacency, however, is FALSE (no "passthrough [to/From]" appears in the show run against the adjacency)' which means that the To: and From: headers are effectively always re-written on the outbound side by default. The impact of this is that parameter profiles actions applied to the inbound sides To: and/or From: headers will be lost on the outbound side unless 'passthrough [to/from]' is set in the configuration. Thus the action **add-not-present** can look like it always adds a parameter on the outbound side, even when the parameter is present on the in-bound side.

- If a parameter profile adds a parameter to the request-line, and the To: header does not have setting 'passthrough to' set against the adjacency, then the re-writing of the To: header which is typically based on the Request Line, will cause the parameter to also appear in the To: header.
- The content of the Request-line may affect the behavior of parameter profiles attached to method profiles. If the request-line that arrives on the in-bound side of the call directly addresses the address of Cisco Unified Border Element (SP Edition), then effectively any call that originates on the out-bound side requires a new Request Line to be generated. This means that parameters arriving on the in-bound side are effectively lost and can cause the action add-not-present to look like it always adds a parameter.

If however, the Request Line address the final destination, then the Request Line is effectively passed across to the outbound side and modified as needed. Parameters in this case are visible on the out-bound side.

Information About Parameter Profiles

Parameter profiles form a set of actions that can be performed against any one header or request-line.

Parameter profiles can only be specified against the following parts of the message:

- Request URI
- To
- From
- Contact

To modify parameters in Contact, To, or From headers, associate a parameter profile in the header profile.

To modify parameters in the request-line, associate a parameter profile with a method profile.



Parameter profiles can be associated with essential methods even though method profiles are not allowed to blacklist/whitelist essential methods.

Configuring Parameter Profiles

Perform this task to configure parameter profiles.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip parameter-profile { profile-name }
- 5. **parameter** {*parameter name*}
- 6. action {add-not-present| add-or-replace | strip}
- 7. end
- 8. show sbc sbc-name sbe sip-parameter-profile [profile name]
- 9. show sbc sbc name sbe sip essential-parameters

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	

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	Command or Action	Purpose
Step 4	<pre>sip parameter-profile {profile-name}</pre>	Configures a parameter profile and enters SBE SIP header configuration mode.
	Example: Router(config-sbc-sbe)# sip parameter-profile parmprof1	
Step 5	<pre>parameter {parameter name}</pre>	Adds a parameter with a specified name to the parameter profile.
	Example: Router(config-sbc-sbe-sip-prm)# parameter user	
Step 6	action {add-not-present add-or-replace strip}	Specifies the action to be performed on the parameter.
	Example: Router(config-sbc-sbe-sip-prm-ele)# action add-not-present value phone	
Step 7	end	Exits the SBE parameter profile parameter configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-prm-ele)# end	
Step 8	<pre>show sbc sbc-name sbe sip-parameter-profile [profile name]</pre>	Displays details for the parameter profile with the designated name.
	Example: Router# show sbc mysbc sbe sip parameter-profile profile1	Use the name default to view the default profile.
Step 9	show sbc sbc name sbe sip essential-headers	Displays a list of the essential headers.
	Example: Router# show sbc mysbc sbe sip essential-headers	

Applying a Parameter Profile to a Header Profile

Perform this task to apply parameter profiles to a header profile.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip header-profile header-profile-name
- 5. header header-name
- 6. parameter-profile parameter-profile-name
- 7. end
- 8. **show sbc** *sbc-name* **sbe sip header-profile** {*profile-name*}

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the configuration mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the configuration mode of the signaling border element (SBE) function of the SBC.
	Example:	
0	Router(config-sbc)# sbe	
Step 4	sip header-profile header-profile-name	Enters the configuration mode for a header profile.
	<pre>Example: Router(config-sbc-sbe-sip)# sip header-profile profile1</pre>	
Step 5	header header-name	Enters the header subcommand mode, where you specify the header type to match.
	Example: Router(config-sbc-sbe-sip-hdr)# header P-Asserted-Identity	
Step 6	<pre>parameter-profile parameter-profile-name</pre>	Configures the parameter profile to apply when the header type is matched.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile parmprof1	
Step 7	end	Exits the SIP header profile header configuration mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# end	
Step 8	show sbc <i>sbc-name</i> sbe sip header-profile <i>name</i>	Displays the header profile information.
	Example: Router# show sbc sbc-name sbe sip header-profile name	

Associating with an Adjacency

Perform the following steps to associate a header profile with an adjacency.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. header-profile inbound profile-name
- 6. end
- 7. show sbc sbc-name sbe sip header-profile name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	header-profile inbound profile-name	Sets profile1 to be used for inbound signaling on adjacency sipGW.
	Example: Router(config-sbc-sbe-adj-sip)# header-profile inbound profile1	

	Command or Action	Purpose
Step 6	end	Exits the header profile mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe-sip-hdr-prf)# end	
Step 7	show sbc <i>sbc-name</i> sbe sip header-profile <i>name</i>	Displays the header profile information.
	Example:	
	Router# show sbc sbc-name sbe sip header-profile name	

Ability to Insert Firewall Parameter in the SIP Contact Header

This feature enables Cisco Unified Border Element (SP Edition) to insert the calling party's network information (IP address) into SIP headers.

You can use this feature to insert the public IP address for user equipment (UE) that is behind the Network Address Translation (NAT) devices into the SIP contact header as a "firewall" parameter. Inserting a firewall parameter in the header is needed because public IP address information in SIP messages is required in order to properly charge the related parties.

A sample modified contact header in SIP message is the following:

Contact:<sip:ea7cf5084c04f49e77644dbe53fd5f1d@10.140.90.6;transport=udp;firewall=10.0.48.41>; Expires=600

See ?\$paranum>Ability to Insert Firewall Parameter in SIP Contact Header Examples? section on page 23-64 for examples on inserting IP address information into SIP contact headers.

Configuring Ability to Insert Firewall Parameter in the SIP Contact Header

Perform these tasks to configure this feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip parameter-profile profile-name
- **5. parameter** {*parameter name*}
- 6. action {add-not-present [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| add-or-replace [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| strip}
- 7. exit
- 8. sip parameter-profile profile-name
- **9. parameter** {*parameter name*}
- **10.** action {add-not-present [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| add-or-replace [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| strip}
- 11. exit

- **12**. **sip header-profile** *profile-name*
- 13. action {add-not-present [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| add-or-replace [value] {*private-ip-address* | *public-ip-address* | *access-user-data*}| strip}
- 14. exit
- **15.** header header-name
- **16.** entry entry_num {action [add-header | as-profile | drop-msg | pass | replace-name | replace-value | strip] | parameter-profile name}
- 17. parameter-profile name
- **18.** sip header-profile *profile-name*
- **19. header** *header-name*
- **20. entry** *entry_num* {*action* [*add-header* | *as-profile* | *drop-msg* | *pass* | *replace-name* | *replace-value* | *strip*] | *parameter-profile name*}
- 21. parameter-profile name

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the configuration mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the configuration mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip parameter-profile {profile-name}</pre>	Configures a parameter profile and enters SBE SIP header configuration mode.
	Example: Router(config-sbc-sbe)# sip parameter-profile proxy-param	
Step 5	<pre>parameter {parameter name}</pre>	Adds a parameter with a specified name to the parameter profile and enters SIP parameter profile parameter
	Example: Router(config-sbc-sbe-sip-prm)# parameter firewall	configuration mode.

	Command or Action	Purpose
Step 6	<pre>action {add-not-present [value] {private-ip-address public-ip-address access-user-data} add-or-replace [value] {private-ip-address public-ip-address access-user-data} strip}</pre>	Configures the action to take on a parameter.
	Example: Router(config-sbc-sbe-sip-prm-ele)# action-strip	
Step 7	exit	Exits SBE parameter profile parameter configuration mode and enters SBE configuration mode.
	Example: Router(config-sbc-sbe-sip-prm-ele)# exit	
Step 8	<pre>sip parameter-profile {profile-name}</pre>	Configures a parameter profile. Enters into SIP parameter profile configuration mode.
	Example: Router(config-sbc-sbe)# sip parameter-profile access-param	
Step 9	<pre>parameter {parameter name}</pre>	Adds a parameter with a specified name to the parameter profile. Enters SIP parameter profile configuration mode.
	Example: Router(config-sbc-sbe-sip-prm)# parameter firewall	
Step 10	<pre>action {add-not-present [value] {private-ip-address public-ip-address access-user-data} add-or-replace [value] {private-ip-address public-ip-address access-user-data} strip}</pre>	Configures the action to take on a parameter.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# action add-or-replace value public-ip-address	
Step 11	exit	Exits to SBE configuration mode.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# exit	
Step 12	sip header-profile profile-name	Configures a header profile. Enters SIP header profile header configuration mode.
	Example: Router(config-sbc-sbe)# sip header-profile proxy	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies which do not have a specific profile configured.

	Command or Action	Purpose
Step 13	<pre>action {add-not-present [value] {private-ip-address public-ip-address access-user-data} add-or-replace [value] {private-ip-address public-ip-address access-user-data} strip}</pre>	Configures the action to take on a parameter.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# action add-or-replace value public-ip-address	
Step 14	exit	Exits SBE header profile header configuration mode and enters into SIP header configuration mode.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# exit	
Step 15	header name	Configures the profile to contain the header test1. Enters SIP header profile header configuration mode.
	Example: Router(config-sbc-sbe-sip-hdr)# header test1	
Step 16	<pre>entry entry_num {action [add-header as-profile drop-msg pass replace-name replace-value strip] parameter-profile name}</pre>	Configures an entry in a profile.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# entry 1	
Step 17	<pre>parameter-profile parameter-profile-name</pre>	Configures the parameter profile to apply when the header type is matched.
	Example: Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile proxy-param	
Step 18	<pre>sip header-profile profile-name</pre>	Configures a header profile. Enters SIP header configuration mode.
	<pre>Example: Router(config-sbc-sbe)# sip header-profile test1</pre>	If you enter the <i>profile-name</i> default , the default profile is configured. This profile is used for all adjacencies which do not have a specific profile configured.
Step 19	header name	Configures the profile to contain the header test1. Enters SBE header profile header configuration mode.
	Example: Router(config-sbc-sbe-sip-hdr)# header test1	

	Command or Action	Purpose
Step 20	<pre>entry entry_num {action [add-header as-profile drop-msg pass replace-name replace-value strip] parameter-profile name}</pre>	Configures an entry in a profile.
	<pre>Example: Router(config-sbc-sbe-sip-hdr-ele)# entry 1 action as-profile</pre>	
Step 21	<pre>parameter-profile parameter-profile-name</pre>	Configures the parameter profile to apply when the header type is matched.
	Example:	
	Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile access-param	

Configuration Examples for SIP Profiles

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- Applying Method Profiles Example, page 23-48
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- Associating Predefined Parameter Profiles Example, page 23-49
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Method Profile Examples

The following example shows the commands and output generated when you configure method profiles.

```
Router# configure terminal
Router(config)# sbc umsbc-node3
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip method-profile test1 ==> Configures new method profile
with name test1
Router(config-sbc-sbe-sip-mth)# method abcd ==> Adds a method abcd to method profile test1
by default, abcd is whitelisted if applied
to the adjacency
Router(config-sbc-sbe-sip-mth)# blacklist ==> Blacklists abcd and allow methods other
than abcd on the adjacency
Router:Nov 13 17:43:11.124: config[65761]: %MGBL-CONFIG-6-DB_COMMIT: Configuration
committed by user 'username'. Use 'show configuration commit changes 100000296' to view
the changes.
Router(config-sbc-sbe-sip-mth)# end
```

Router:Nov 13 17:43:14.866: config[65761]: %MGBL-SYS-5-CONFIG_I : Configured from console by username

This example shows the output for all method profiles.

This command describes the available method profiles which can be used by the adjacencies. By default, the "default" method profile is configured implicitly and applied to both inbound and outbound directions of all the adjacencies. The default method profile is always active unless it is overwritten by a user-configured method profile. "In use" explains whether the method profile is used by any adjacency or not. When the value is Yes, the "default" method profile is applied to all the adjacencies and is in use. However "test1" has been configured, but not applied to any of the adjacencies. Once you apply the test1 method profile to any adjacency, test1 shows Yes in the "In use" field.

Router# show sbc test sbe sip method-profile

Method profiles for SBC service "test1"

Name	In use
test1	No
mprof1	No
default	Yes
preset-acc-in-mth	No
preset-std-in-mth	No
preset-acc-out-mth	No
preset-core-in-mth	No
preset-std-out-mth	No
preset-core-out-mth	No
preset-ipsec-in-mth	No
preset-ipsec-out-mth	No
preset-ibcf-ext-in-mth	No
preset-ibcf-int-in-mth	No
preset-ibcf-utr-in-mth	No
preset-ibcf-int-in-mth	No
preset-ibcf-utr-in-mth	No
preset-ibcf-ext-out-mth	No
preset-ibcf-int-out-mth	No
preset-ibcf-utr-out-mth	No

This example shows the output for the method profiles test1.

```
Router# show sbc test sbe sip method-profile test1
Method profile "test1"
Description:
Type: Whitelist
Methods:
INVITE
action as-profile
map-status-code
range 50X value 500
range 60X value 600
Not in use with any adjacencies
```

Applying Method Profiles Example

The following examples show the commands and output generated when you are applying a method profile to Cisco Unified Border Element (SP Edition).

The **method-profile inbound test1** command applies method profile "test1" on the inbound direction. It means that for all incoming messages, check for the method type "abcd." If the "abcd" method arrives, blacklist it and generate error code 405 Method Not Allowed. All other methods are allowed.

```
Router# configure terminal
Router(config) # sbc umsbc-node3
Router(config-sbc)# sbe
Router(config-sbc-sbe) # adjacency sip sipp-10
Router(config-sbc-sbe-adj-sip)# method-profile inbound test1
Router:Nov 13 17:44:28.609 : config[65761]: %MGBL-CONFIG-6-DB_COMMIT : Configuration
committed by user 'username'. Use 'show configuration commit changes 1000000297' to view
the changes.
Router(config-sbc-sbe-adj-sip) # end
Router:Nov 13 17:44:31.637 : config[65761]: %MGBL-SYS-5-CONFIG_I : Configured from console
by username
Router# show sbc umsbc-node3 sbe sip method-profile
Method profiles for SBC service "umsbc-node3"
 Name
                              In use
  test1
                               Yes
 testb
                               No
Router# show sbc umsbc-node3 sbe sip method-profile test1
 Method profile "test1"
   Type: Blacklist
   Methods:
     abcd
   In use by:
      Adjacency: sipp-10 (in)
```

Associating Predefined Header Profiles Example

This example shows how to ensure that the parameter myparm=myvalue is added to the request-line of an INVITE:

First, configure a parameter profile for myparm:

Router# configure terminal Router(config)# sbc test
```
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip parameter-profile parmprof1
Router(config-sbc-sbe-sip-prm)# parameter myparm
Router(config-sbc-sbe-sip-prm-ele)# action add-not-present value myvalue
```

Then configure and associate with a method profile:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip method-profile mthdprof1
Router(config-sbc-sbe-sip-mth)# method INVITE
Router(config-sbc-sbe-sip-prm-ele)# parameter-profile parmprof1
```

Finally, associate with an adjacency

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# method-profile outbound mthdprof1
```

At the inbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone;myparm=myvalue SIP/2.0

Associating Predefined Parameter Profiles Example

The following example shows how to ensure P-Asserted-Identity is always passed in an INVITE if it contains user=phone.

First, configure a header profile which references a P-Asserted-Identity header:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-profile hdrprof1
Router(config-sbc-sbe-sip-hdr)# header P-Asserted-Identity
Router(config-sbc-sbe-sip-hdr-ele)# action pass
Router(config-sbc-sbe-sip-hdr-ele-act)# condition header-value contains user=phone
```

Then create and associate the header profile with a method profile:

```
Router(config-sbc-sbe)# sip method-profile mthdprof1
Router(config-sbc-sbe-sip-mth)# method INVITE
Router(config-sbc-sbe-sip-prm-ele)# header-profile hdrprof1
```

Finally, associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# method-profile outbound mthdprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
P-Asserted-Identity: "rob" <sip:1234567@cisco.com;user=phone>
```

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
P-Asserted-Identity: "rob" <sip:1234567@cisco.com;user=phone>
```

Associating Response Code Mapping Example

The following example shows how to create a status-code map so that all 5XX responses to an INVITE are mapped to 500.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip method-profile mthdprof1
Router(config-sbc-sbe-sip-mth)# method INVITE
Router(config-sbc-sbe-sip-mth-ele)# map-status-code
Router(config-sbc-sbe-sip-mth-ele-map)# range 5XX value 500
```

Finally, associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# method-profile outbound mthdprof
```

At the inbound side:

SIP/2.0 501 Not Implemented

At the outbound side:

abcd

SIP/2.0 500 Internal Server Error

Configuring Header Profiles Example

The following example shows the commands and output generated when you configure the header profiles.

```
Router(config)# sbc umsbc-node3 sbe
Router(config-sbc-sbe)# sip header-profile EXAMPLE
Router(config-sbc-sbe-sip-hdr)# blacklist
Router(config-sbc-sbe-sip-hdr)# header abcd
Router# show sbc sbc4 sbe sip header-profile EXAMPLE
Header profile EXAMPLE
Type: Whitelist
Headers:
```

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```
Cisco-Guid
   Entry 1:
     action add-first-header
 User-Agent:
   Entry 1:
     action as-profile
 Remote-Party-ID
   Entry 1:
     action strip
       condition header-value contains user=phone
    Entry 2:
     parameter-profile adduser
 P-Asserted-Identity
    Entry 1:
      action strip
       condition header-value contains user=phone
 Organisation
    Entrv 1:
     action replace-value value Cisco-Systems
        condition header-value contains MCI
In use by:
  Adjacency: callgen100sip (in, out)
```

Applying Header Profiles Example

The following example shows the commands and output generated when you are applying a header profile to Cisco Unified Border Element (SP Edition).

```
Router# configure terminal
Router(config) # sbc umsbc-node3 sbe
Router(config-sbc-sbe) # adjacency sip sipp-10
Router(config-sbc-sbe-adj-sip)# header-profile inbound test1
Router(config-sbc-sbe-adj-sip)# header-profile outbound test1
Router# show sbc umsbc-node3 sbe sip header-profile test1
  Header profile "test1"
   Type:
            Blacklist
   Headers:
      abcd
    In use by:
       Adjacency: sipp-10 (in, out)
show running-config
sbc umsbc-node3
she
  activate
sip header-profile test1
   blacklist
   header abcd
  Т
  adjacency sip sipp-10
  header-profile inbound test1
   header-profile outbound test1
   signaling-address ipv4 88.88.109.8
   signaling-port 5060
   remote-address ipv4 10.10.105.222 255.255.255.255
   security trusted-encrypted
   signaling-peer 10.10.105.222
   signaling-peer-port 5060
```

account sip-customer

Header Manipulation Examples

Example—Removing P-Asserted-Identity Header

The following example shows how to remove the header in any message if the header P-Asserted-Identity contains *user=phone*.

First, access the header:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header P-Asserted-Identity
Router(config-sbc-sbe-hdr-ele)# action strip
Router(config-sbc-sbe-hdr-ele-act)# condition header-value contains user=phone
```

Next, associate the header with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

P-Asserted-Identity: "rob" <sip:1234567@cisco.com;user=phone>

At the outbound side:

No P-Asserted-Identity header present

Add this condition in addition to a previous existing condition:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header P-Asserted-Identity
Router(config-sbc-sbe-hdr-ele)# entry 2
Router(config-sbc-sbe-hdr-ele)# action strip
Router(config-sbc-sbe-hdr-ele-act)# condition header-value contains user=phone
```

Finally, associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
P-Asserted-Identity: "rob" <sip:1234567@cisco.com;user=phone>
```

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
<No P-Asserted-Identity header present>
```

Example—Removing Header Based on Condition in Another Header

The next example shows how to remove a header based on a condition in another header in the message. First, strip the P-Asserted-Identity header, but only if Call-Info: contains "telephone-event."

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header P-Asserted-Identity
Router(config-sbc-sbe-hdr-ele)# action strip
Router(config-sbc-sbe-hdr-ele-act)# condition header-name Call-Info header-value contains
telephone-event
```

Then associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
P-Asserted-Identity: "rob" <sip:1234567@cisco.com;user=phone>
...
Call-Info: <sip:8985@10.131.132.6>;method="NOTIFY;Event=telephone-event;Duration=1000"
```

The result at the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
<No P-Asserted-Identity header present>
```

Example—Removing Organization Header from All Responses

The next example removes an Organization header from all Responses:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Organization
Router(config-sbc-sbe-hdr-ele)# action strip
Router(config-sbc-sbe-hdr-ele-act)# condition status-code eq 200
```

Associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
SIP/2.0 200 OK
...
Allow: INVITE, ACK, PRACK, SUBSCRIBE, BYE, CANCEL, NOTIFY, INFO, REFER, UPDATE
At the outbound side:
```

SIP/2.0 200 OK
...
<No allow header present>

Example—Transforming a Header into Another Header

This example transforms one header into another header (Diversion into Hist-Info).

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Diversion
Router(config-sbc-sbe-hdr-ele)# action replace-name value Hist-Info
```

Associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Diversion: <sip:1234567@cisco.com>;reason=unconditional;counter=1;privacy=off

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Hist-Info: <sip:1234567@cisco.com>;reason=unconditional;counter=1;privacy=off
```

Example—Outgoing Messages Contain a Specific Header

This example ensures all outgoing messages contain a specific header (Organization: Cisco.com).

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Organization
Router(config-sbc-sbe-hdr-ele)# action add-first-header value cisco.com
```

Associate the header profile with an adjacency:

Router# configure terminal

```
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
```

Router(config-sbc-sbe-sip)# header-profile outbound headprof1

At the inbound side:

```
INVITE sip:1234567@cisco,com;user=phone SIP/2.0
...
<no Organization header present>
```

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Organization: cisco.com

Example—Blacklisting a Header

This example blacklists a header (all instances are removed for any method/response).

Note

Provide the set of the

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr-ele)# blacklist
Router(config-sbc-sbe-sip-hdr)# header Organization
```

Or:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe-hdr)# sip header-profile headprof1
Router(config-sbc-sbe-hdr-ele)# blacklist
Router(config-sbc-sbe-sip-hdr)# header Organization
Router(config-sbc-sbe-sip-hdr)# action as-profile
```

Associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco,com;user=phone SIP/2.0
...
Organization: cisco.com
```

At the outbound side:

INVITE sip:1234567@cisco,com;user=phone SIP/2.0
...
<no Organization: header present>

Example—Whitelisting a Header

This example whitelists a header (pass in all methods/responses).

Note

This can only be specified against a whitelist type of profile which is a default profile and same as "no blacklist."

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Organization
Or:
```

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Organization
Router(config-sbc-sbe-hdr-ele)# action as-profile
```

Associate the header profile with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Organization: cisco.com
```

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Organization: cisco.com
```

Example—Passing a Date Header

This example passes a header (Date) conditionally in a 200 response.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header Date
Router(config-sbc-sbe-hdr-ele)# action pass
Router(config-sbc-sbe-hdr-ele-act)# condition status-code eg 200
```

Associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

Ensure no other responses contain a Date: header SIP/2.0 200 OK ... Date: Mon, 01 Jan 2008 GMT

At the outbound side:-

SIP/2.0 200 OK

. . .

Date: Mon, 01 Jan 2008 GMT

Also try all responses containing a Date: header and ensure the 200 OK only contains one

Example—Stripping Organization Headers in INVITE

This example strips all 'Organization' headers in an INVITE. To do this, a header profile is created and then associated it with a method profile.

Note

Header profiles can be associated with vital (essential) methods.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headerprof1
Router(config-sbc-sbe-hdr)# blacklist
Router(config-sbc-sbe-hdr-ele)# header Organization
```

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe) sip method-profile methodprof1
Router(config-sbc-sbe-sip-mth) blacklist
Router(config-sbc-sbe-sip-mth) method INVITE
Router(config-sbc-sbe-sip-mth-ele) header-profile headerprof1
```

Associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# method-profile outbound methodprof1
```

At the inbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
Organization: cisco.com

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
<no Organization: header present>
```

Example—Applying Parameter Profile

This example applies a parameter profile to add user=phone into the request-line of an INVITE.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip parameter-profile test
Router(config-sbc-sbe-sip-prm)# parameter user
Router(config-sbc-sbe-sip-prm-ele)# action add-not-present value phone
```

Associate with a method profile:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe) sip method-profile test
Router(config-sbc-sbe-sip-mth) method INVITE
Router(config-sbc-sbe-sip-mth-ele) parameter-profile test
```

Associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# method-profile inbound headprof1
```

At the inbound side:

INVITE sip:1234567@cisco.com SIP/2.0

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0

Example—Stripping P-Called-Party-Identity

This example shows how to strip the P-Called-Party-Identity and modify the To: header based on its content:

```
Router# configure terminal
Router(config) # sbc test
Router(config-sbc) # sbe
Router(config-sbc-sbe) # sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# store-rule entry 1
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the P-Called-Party-Identity"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition header-name P-Called-Party-Identity
header-value store-as pcpid
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router (config-sbc-sbe-sip-hdr) # header P-Called-Party-Identity entry 1
Router(config-sbc-sbe-sip-hdr-ele)# action strip
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr-ele)# exit
Router(config-sbc-sbe-sip-hdr) # header To entry 1
Router(config-sbc-sbe-sip-hdr-ele)# action replace-value value "${pcpid}"
Router(config-sbc-sbe-sip-hdr-ele-act)# description "replace the To value"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable pcpid is-defined eq true
```

Associate with an outbound adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

Replacing Outbound Request Line Example

This example shows how to replace the outbound request-line with host 172.1.1.1 if user = begins with 1234:

```
Router# configure terminal
Router(config) # sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe) # sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# store-rule entry 1
Router (config-sbc-sbe-sip-hdr-ele-act) # condition request-uri is-sip-uri eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and request-uri sip-uri-user store-as
user
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# request-line entry 1
Router(config-sbc-sbe-sip-hdr-ele)# action replace-value value "sip:${user}@172.1.1.1"
Router (config-sbc-sbe-sip-hdr-ele-act) # description "convert RPID param into Privacy
header value"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition is-request eq true
Router (config-sbc-sbe-sip-hdr-ele-act) # condition and request-uri is-sip-uri eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and request-uri sip-uri-user regex-match
`^1234"
```

Associate with an outbound adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

Example—P-KT-UE-IP Header Support

The P-KT-UE-IP header is a type of private header that is supported as a type of SIP header manipulation. The examples in this section show how to remove any existing P-KT-UE-IP headers from all received messages and then replace them with a single P-KT-UE-IP header for INVITE and OOD requests. In the examples, the call is placed from adj1 to adj2.

The following shows how to configure a header profile with two entries. The first entry strips the "P-KT-UE-IP" header and the second entry adds the "P-KT-UE-IP" with a value set to the 18-character string \${msg.rmt_ip_addr}.

```
Router(config-sbc-sbe)# sip header-profile kt
Router(config-sbc-sbe-sip-hdr)# store-rule entry 1
Router(config-sbc-sbe-sip-hdr-ele)# condition adjacency signaling-peer store-as address
Router(config-sbc-sbe-sip-hdr-ele)# exit
Router(config-sbc-sbe-sip-hdr-ele)# entry 1 action strip
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr-ele)# entry 2 action add-header value "${address}"
```

The following applies the above header profile to the incoming adjacency as an inbound header profile.

```
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# header-profile inbound kt
```

The following configures a header profile to allow passthrough of the "P-KT-UE-IP" header.

Router(config-sbc-sbe)# sip header-profile kt-pass Router(config-sbc-sbe-sip-hdr)# header P-KT-UE-IP Router(config-sbc-sbe-sip-hdr-ele)# action pass

The following applies the above header profile to the outgoing adjacency as an outbound header profile.

```
Router(config-sbc-sbe)# adjacency sip adj2
Router(config-sbc-sbe-adj-sip)# header-profile outbound kt-pass
```

Response Filtering Example

The following example drops SIP 183 provisional responses from a header profile based on matching the header * associated with inbound and outbound adjacencies.

First, create a header profile headprof1 to match on header * and drop the message:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-hdr)# header *
Router(config-sbc-sbe-hdr-ele)# action drop-msg
Router(config-sbc-sbehdr-ele-act)# condition status-code eq 183
```

Associate the profile headprof1 to the inbound side of an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adjacencyA
Router(config-sbc-sbe-adj-sip)# header-profile inbound headerprof1
```

Associate the profile headprof1 to the inbound and outbound sides of another adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adjacencyB
Router(config-sbc-sbe-adj-sip)# header-profile inbound headerprof1
```

Router# configure terminal Router(config)# sbc test Router(config-sbc)# sbe Router(config-sbc-sbe)# adjacency sip adjacencyB Router(config-sbc-sbe-adj-sip)# header-profile outbound headerprof1

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# header To
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile parmprof1
```

Parameter Profile Examples

This example shows how to add a user=phone parameter into the To: header if one has not already been specified in a header.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip parameter-profile parmprof1
Router(config-sbc-sbe-sip-prm)# parameter user
Router(config-sbc-sbe-sip-prm-ele)# action add-not-present value phone
```

Now add to a header profile:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# header To
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile parmprof1
```

Now associate with an adjacency:

Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com>;tag=1234;
```

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;user=phone>;tag=1234

This example removes the 'user' parameter ('user=phone','user=fax' ...) from the To: header.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip parameter-profile parmprof1
Router(config-sbc-sbe-sip-prm)# parameter user
Router(config-sbc-sbe-sip-prm-ele)# action strip
```

Add to a header profile:

Finally, associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
```

To: "rob" <sip:1234567@cisco.com;user=phone;tag=1234;

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com>;tag=1234

This example shows how to replace 'user=phone' parameter with user=fax or to add user=fax if a user parameter is not present in the header.

Router# configure terminal

```
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip parameter-profile parmprof1
Router(config-sbc-sbe-sip-prm)# parameter user
Router(config-sbc-sbe-sip-prm-ele)# action add-or-replace value fax
```

Add to a header profile:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# header To
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile parmprof1
```

Finally, associate with an adjacency:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
```

To: "rob" <sip:1234567@cisco.com;user=phone;tag=1234;

At the outbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;user=fax>;tag=1234
```

Or

At the inbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;tag=1234;</pre>

At the outbound side:

INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;user=fax>;tag=1234

The next example adds 'user=phone' parameter if it is not already present in the header.

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip parameter-profile parmprof1
Router(config-sbc-sbe-sip-prm)# parameter user
Router(config-sbc-sbe-sip-prm-ele)# action add-not-present value phone
```

Add parameter profile to a header profile:

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# sip header-profile headprof1
Router(config-sbc-sbe-sip-hdr)# header To
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile parmprof1
```

Finally, associate with an adjacency

```
Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-profile outbound headprof1
```

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;user=fax;tag=1234;</pre>
```

At the outbound side:

No parameter added as a user parameter already exists INVITE sip:1234567@cisco.com;user=phone SIP/2.0 ... To: "rob" <sip:1234567@cisco.com>;tag=1234

Or

At the inbound side:

```
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...
To: "rob" <sip:1234567@cisco.com;tag=1234;
At the outbound side:
INVITE sip:1234567@cisco.com;user=phone SIP/2.0
...</pre>
```

```
To: "rob" <sip:1234567@cisco.com;user=phone>;tag=1234
```

Ability to Insert Firewall Parameter in SIP Contact Header Examples

This example adds a SIP parameter profile to remove or append the parameter called firewall:

```
Router(config-sbc-sbe)# sip parameter-profile proxy-param
Router(config-sbc-sbe-sip-prm)# parameter firewall
Router(config-sbc-sbe-sip-prm-ele)# action strip
Router(config-sbc-sbe-sip-prm-ele)# sip parameter-profile access-param
Router(config-sbc-sbe-sip-prm)# parameter firewall
Router(config-sbc-sbe-sip-prm-ele)# action add-or-replace value public-ip-address
```

This example adds a SIP header profile and associates the parameter profile with the header profile

```
Router(config-sbc-sbe-sip-prm-ele)# sip header-profile proxy
Router(config-sbc-sbe-sip-hdr)# header contact entry 1
Router(config-sbc-sbe-sip-hdr-ele)# action as-profile
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile proxy-param
Router(config-sbc-sbe-sip-hdr-ele)# sip header-profile access
Router(config-sbc-sbe-sip-hdr)# header contact
Router(config-sbc-sbe-sip-hdr-ele)# entry 1 action as-profile
Router(config-sbc-sbe-sip-hdr-ele)# parameter-profile access-param
```

This example adds a SIP header profile to a SIP adjacency:

```
adjacency sip sip-proxy
header-profile inbound proxy
header-profile outbound access
adjacency sip sip-user
header-profile inbound access
header-profile outbound proxy
```

SIP Message Editing Using Editors

Note

This section describes body, header, method, option, and parameter editors. The ?\$paranum>SDP Editing Using Script-Based Editors? section on page 23-84 describes script-based editors for modifying the SDP content in SIP messages. You can apply any combination of both types of editors on the SBC for editing SIP messages.

In Release 2.4S, profiles were introduced to enable the SBC to conditionally modify SIP messages. You could configure a profile to modify the body, header, method, option, or parameter of SIP messages that met the matching criteria you specified. This approach was flexible but posed the following limitations:

- Matching criteria could not be set for the vital parts of a message because there was a probability of the call failing if the vital parts of the message were modified.
- With certain limited exceptions, the vital parts of a message could not be modified because the original content of these vital parts was not available at the point at which the profiles were applied.

From Release 3.3S, the concept of *editors* has been introduced. An editor refers to any kind of SBC configuration that is used for conditionally editing SIP messages. Profiles that were introduced in earlier releases are now renamed as editors. For example, body profiles are now known as body editors, header profiles are known as header editors, and so on.

Editors can be associated with an adjacency and linked together so that they can be applied in a specified sequence at run time. In addition, you can test editors by applying them on a test message (a SIP INVITE). You can use the output of the test to determine whether the editors meet your requirements.

In Cisco IOS XE Release 3.3S, the following additional enhancements have been introduced in the SIP Message Editing feature:

• To and From multimode fiber optic edits

Prior to Cisco IOS XE Release 3.3S, the To and From outbound headers of only out-of-dialog messages and dialog-creating messages could be edited. After an edit was performed on a dialog-creating message, the edit was automatically propagated across all the new messages sent on the dialog. From Cisco IOS XE Release 3.3S, edits on the To and From headers can also be performed on in-dialog messages. There is no automatic propagation of these edits. This requires you to ensure that the edits are consistently performed for all messages sent on the dialog.

• Resource Priority header inspection

Prior to Cisco IOS XE Release 3.3S, the Resource Priority header inspection function examined a message before any inbound MMF editing was performed. From Cisco IOS XE Release 3.3S, the Resource Priority header inspection function examines a message after inbound editing has been performed.

• 100rel_required match condition variable

Prior to Cisco IOS XE Release 3.3S, the *100rel_required* match condition variable was a call property that was updated when new information about 100rel support came in from each call leg. From Cisco IOS XE Release 3.3S, this variable is an indicator of whether the received message is marked as *Required: 100rel*.

• Failure responses

Prior to Cisco IOS XE Release 3.3S, failures encountered during message editing resulted in the SBC sending a rejection for the unedited message. From Cisco IOS XE Release 3.3S, the response contains the state of the message at the point of failure. For example, headers added during editing are mentioned in the failure response.

The following sections provide information about implementing SIP message editing using body, header, method, option, and parameter editors:

- Restrictions for SIP Message Editing, page 23-65
- Guidelines for Naming Editors, page 23-66
- Configuring Editors, page 23-66
- Configuration Examples for SIP Message Editors, page 23-76

Restrictions for SIP Message Editing

The SIP Message Editing feature does not support the following actions:

- Editing To and From header tags
- Applying the pass and strip actions on To and From header tags
- Outbound editing of Via headers
- Changing the method types of INVITE, CANCEL, and ACK messages

Guidelines for Naming Editors

Apply the following guidelines while naming an editor:

- Ensure that each editor has a unique name. Apply this guideline across editors. For example, ensure that the name of a header editor is not the same as the name of a method editor.
- Note that an editor and a profile should have the same name to ensure an easy migration path.

Configuring Editors

This task describes how to configure editors on the SBC.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip editor-type {editor | profile}
- 5. sip body-editor editor-name
- 6. exit
- 7. sip method-editor {editor-name | default}
- 8. exit
- 9. sip option-editor {editor-name | default}
- 10. exit
- **11.** sip parameter-editor {*editor-name* | default}
- 12. exit
- **13**. **sip header-editor** {*editor-name* | **default**}
- 14. exit
- 15. adjacency sip adjacency-name
- **16.** editor-type {editor | profile}
- **17.** header-editor {inbound | outbound} {editor-name | default}
- **18.** method-editor {inbound | outbound} {editor-name | default}
- **19. option-editor** [**ua** | **proxy**] {**inbound** | **outbound**} {*editor-name* | **default**}
- **20. body-editor** {**inbound** | **outbound**} {*editor-name*}
- 21. editor-list {after-send | before-receive}

- 22. editor *order-number editor-name* [condition [body contains sdp]]
- 23. end
- 24. show sbc *sbc-name* sbe editors
- 25. show sbc sbc-name sbe sip header-editor [editor-name]
- 26. show sbc sbc-name sbe sip body-editor [editor-name]
- 27. show sbc sbc-name sbe sip method-editor [editor-name]
- **28.** show sbc *sbc-name* sbe sip option-editor [*editor-name*]
- **29.** show sbc sbc-name sbe sip parameter-editor [editor-name]

DETAILED STEPS

	Command or Action	Purpose				
Step 1	configure terminal	Enters the global configuration mode.				
	Example: Router# configure terminal					
Step 2	sbc sbc-name	Enters the SBC service mode.				
	Example: Router(config)# sbc mysbc	• <i>sbc-name</i> —Name of the SBC.				
Step 3	sbe	Enters the SBE configuration mode of the SBC.				
	Example: Router(config-sbc)# sbe					
Step 4	<pre>sip editor-type {editor profile}</pre>	Sets the default type of editor to be applied on any adjacency that has not been explicitly set.				
	Example: Router(config-sbc-sbe)# sip editor-type editor	• editor—Sets the default for using the method, header, option, parameter, or body editor.				
		• profile —Sets the default for using the method, header, option, parameter, or body profile.				
Step 5	sip body-editor editor-name	Creates a body editor to filter non-SDP message bodies from incoming and outgoing SIP messages.				
	Example: Router(config-sbc-sbe)# sip body-editor BodyEditor1	• <i>editor-name</i> —Specifies the name of the body editor.				
		Enters the SIP body configuration mode. Use the following commands under this mode to configure the body editor:				
		• body —Adds a body type to this editor.				
		• description —Sets the description for this editor.				
		The body command enters the SIP body editor element configuration mode, where the following commands can be used:				
		• action —Specifies the action to be performed on the body.				
		• hunt-on-reject—Specifies trigger hunting.				
Step 6	exit	Exits the SIP body configuration mode and enters the SBE configuration mode.				
	Example: Router(config-sbc-sbe-mep-bdy)# exit					

	Command or Action	Purpose				
Step 7	sip method-editor {editor-name default} Configures a method editor.					
	Example:	 <i>editor-name</i>—Specifies the name of the method editor. default—Configures the default method editor. This editor is used for all the adjacencies that do not have a specific editor configured. Enters the SIP method configuration mode. Use the following commands under this mode to configure the method editor: blacklist—Sets this editor to be blacklist. 				
	Router(config-sbc-sbe)# sip method-editor MethodEditor1					
		• description —Sets the description for this editor.				
		• method —Adds a method to this editor.				
		The method command enters the SIP method editor element configuration mode, where the following commands can be used:				
		• action —Specifies the action performed on the method.				
		• body-editor —Adds a body editor to act on the method.				
		• header-editor —Adds a header editor to act on the method.				
		• map-status-code —Allows mapping of the response codes received for a method.				
Step 8	exit	Exits the SIP method configuration mode and enters the SBE configuration mode.				
	Example: Router(config-sbc-sbe-mep-mth)# exit					
Step 9	<pre>sip option-editor {editor-name default}</pre>	Configures an option editor.				
	Example:	• <i>editor-name</i> —Specifies the name of the option editor.				
	Router(config-sbc-sbe)# sip option-editor OptionEditor1	• default —Configures the default option editor.				
		Enters the SIP option configuration mode. Use the following commands under this mode to configure the option editor:				
		• blacklist —Sets this editor to be blacklist.				
		• description —Sets the description for this editor.				
		• option —Adds an option to this editor.				
Step 10	exit	Exits the SIP option configuration mode and enters the SBE configuration mode.				
	Example: Router(config-sbc-sbe-mep-opt)# exit					

	Command or Action	Purpose				
Step 11	sip parameter-editor editor-name	Configures a parameter editor.				
	Example:	• <i>editor-name</i> —Specifies the name of the parameter editor.				
	Router(config-sbc-sbe)# sip parameter-editor ParameterEditor1	Enters the SIP parameter configuration mode. Use the following commands under this mode to configure the parameter editor:				
		• blacklist —Sets this editor to be blacklist.				
		• description —Sets the description for this editor.				
		• parameter —Adds an parameter to this editor.				
		The parameter command enters the SIP parameter editor element configuration mode, from where you can configure the action to be taken on an element type in the parameter editor using the action command.				
Step 12	exit	Exits the SIP parameter configuration mode and enters the SBE configuration mode.				
	Example:					
	Router(config-sbc-sbe-mep-prm)# exit					

	Command or Action	Purpose			
Step 13	<pre>sip header-editor {editor-name default}</pre>	Configures a header editor.			
	Example:	• <i>editor-name</i> —Specifies the name of the header editor.			
	Router(config-sbc-sbe)# sip header-editor HeaderEditor1	• default —Configures the default header editor.			
	E fı tl	Enters the SIP header configuration mode. Use the following commands under this mode to configure the header editor:			
		• blacklist —Sets this editor to be blacklist.			
		• description —Sets the description for this editor.			
		• div-address —Specifies a priority list of headers from which the diverted-by number is to be derived (inbound only). Enters the SIP header editor diversion header configuration mode, from where you can use the following command:			
		 header-prio—Specifies a priority-ordered list for extracting the diverted-by address. 			
		• dst-address —Specifies a priority list of headers from which the called party address is to be derived (inbound only). Enters the SIP header editor destination header configuration mode, from where you can use the following command:			
		 header-prio—Specifies a priority ordered list for extracting the destination address. 			
		• header —Adds a header to this editor. Enters the SIP header editor header configuration mode, from where you can use the following commands:			
		 action—Specifies the type of action. Enters the SIP header editor header action mode, from where you can use the condition command to specify one or more conditions for the action to be effective and the parameter-editor command to specify the parameter editor. 			
		 parameter-editor—Specifies the parameter editor. 			

	Command or Action	 Purpose request-line—Allow actions to modify the Request Line (outbound side only). Enters the SIP header editor header configuration mode, from where you can use the following commands: action—Specifies the type of action. Enters the SIP header editor header action mode, from where you can use the condition command to specify one or more conditions for the action to be effective and the parameter-editor. parameter-editor—Specifies the parameter editor. 			
		• request-line —Allow actions to modify the Request Line (outbound side only). Enters the SIP header editor header configuration mode, from where you can use the following commands:			
		 action—Specifies the type of action. Enters the SIP header editor header action mode, from where you can use the condition command to specify one or more conditions for the action to be effective and the parameter-editor command to specify the parameter editor. 			
		 parameter-editor—Specifies the parameter editor. 			
		• src-address —Specifies a priority list of headers from which the calling party address is to be derived (inbound only). Enters the SIP header editor calling party configuration mode, from where you can use the following command:			
		 header-prio—Specifies a priority ordered list for extracting the source address. 			
		• store-rule —Creates a store rule to extract variables from headers. Enters the SIP header editor header action configuration mode, from where you can use the following commands:			
		 condition—Specifies one or more conditions for the action to be effective. 			
		 description—Sets the description for this action. 			
Step 14	exit	Exits the SIP header configuration mode and enters the SBE configuration mode.			
	Example: Router(config-sbc-sbe-mep-hdr)# exit				
Step 15	adjacency sip adjacency-name	Enters the SBE SIP adjacency configuration mode.			
	Example: Router(config-sbc-sbe)# adjacency sip SIPP	• <i>adjacency-name</i> —Name of the service.			
Step 16	editor-type {editor profile}	Specifies the editor type for the SIP adjacency to apply.			
	Example: Router(config-sbc-sbe-sip)# editor-type editor	• editor —Uses the method, header, option, parameter, or body editor.			
		• profile —Uses the method, header, option, parameter, or body profile.			

	Command or Action	Purpose			
Step 17	<pre>header-editor {inbound outbound} {editor-name default}</pre>	Sets a specified header editor for inbound and outbound signaling on the SBE SIP adjacency.			
		• inbound —Sets the inbound SIP header editor.			
	Example: Router(config-sbc-sbe-sip)# header-editor inbound	• outbound —Sets the outbound SIP header editor.			
	HeaderEditor1	• <i>editor-name</i> —Name of the header editor to be set for inbound or outbound signaling on the adjacency.			
		• default —Sets the header editor to the default settings.			
Step 18	method-editor {inbound outbound} {editor-name	Configures the method editor.			
	default }	• inbound —Sets the inbound SIP method editor.			
	Example: Router(config-sbc-sbe-sip)# method-editor inbound	• outbound —Sets the outbound SIP method editor.			
	HeaderEditor1	• <i>editor-name</i> —Name of the method editor to be set for inbound or outbound signaling on the adjacency.			
		• default —Sets the method editor to the default settings.			
Step 19	<pre>option-editor [ua proxy] [inbound outbound] [editor-name default]</pre>	Sets the adjacency to use the specified editor for white or blacklisting options.			
		• ua —Sets the SIP ua option editors.			
	Example: Router(config-sbc-sbe-adj-sip)# option-editor ua	• proxy —Sets the SIP proxy option editors.			
	inbound OptionHeader1	• inbound —Sets the inbound SIP option editors.			
		• outbound —Sets the outbound SIP option editors.			
		• <i>editor-name</i> —Name of editor to use.			
		• default —Sets the method editor to the default settings.			
Step 20	<pre>body-editor {inbound outbound} {editor-name}</pre>	Associates a body editor to the SIP adjacency so that the body editor acts on incoming and outgoing SIP messages.			
	Example: Router(config-sbc-sbe-adj-sip)# body-editor inbound BodyEditor1	• inbound —Associates the body editor to act on inbound messages on the SIP adjacency.			
		• outbound —Associates the body editor to act on outbound messages on the SIP adjacency.			
		Note When the message is passed through the SBC, the body editor is applied in both the inbound and outbound directions on the respective adjacencies on which the message is routed.			
		• <i>editor-name</i> —Specifies a name for the body editor. The maximum length is 30 characters.			

	Command or Action	Purpose				
Step 21	<pre>editor-list {after-send before-receive}</pre>	Configures a list of registered editors.				
	Example: Router(config-sbc-sbe-adj-sip)# editor-list after-send	 after-send—Specifies that the outgoing message must be edited after it is processed by the adjacency and just before it is forwarded from the adjacency. before-receive—Specifies that the incoming 				
		message must be edited just after it is received on the adjacency and before the adjacency begins processing it.				
		Enters the SIP editor configuration mode.				
Step 22	<pre>editor order-number editor-name [condition [body contains sdp]] Example:</pre>	Configures an editor in the editor list. For each editor that you want to apply in a sequence, run this command to specify the order of the editor in the editor list.				
	Router(config-sbc-sbe-adj-sip-ed)# editor 1 bodyeditor1	 Note You can add any combination of script-based editors and body, header, method, option, and parameter editors in the editor list. order-number-Order in which the editor must 				
		be applied. The range is from 1 to 2147483647.				
		• <i>editor-name</i> —Specifies the name of the editor that you want to apply to messages that are processed by the adjacency.				
		• condition —Specifies that there are one or more conditions for the editor to be applied.				
		• body contains sdp —Specifies that the message body must be SDP-based content. The editor is applied only if this condition is met. Include body contains sdp in the command for script-based editors.				
Step 23	end	Exits the SIP editor configuration mode, and enters the privileged EXEC mode.				
	<pre>Example: Router(config-sbc-sbe-adj-sip)# end</pre>					
Step 24	4 show sbc sbc-name sbe editors Lists all the configured editors.					
	Example: Router# show sbc mysbc sbe editors					

	Command or Action	Purpose			
Step 25	show sbc sbc-name sbe sip body-editor [editor-name]	Displays the details of all body editors, or displays details pertaining to the specified body editor.			
	Example: Router# show sbc mysbc sbe sip body-editor	• <i>sbc-name</i> —Specifies the name of the SBC service.			
	BodyEditor1	• <i>editor-name</i> —Specifies the name of the editor and displays details about the specified editor. If omitted, the command shows information about all the SIP body editors.			
Step 26	<pre>show sbc sbc-name sbe sip header-editor [editor-name]</pre>	Displays the details of all header editors, or displays details pertaining to the specified header editor.			
	Example:	• <i>sbc-name</i> —Specifies the name of the SBC service.			
	Router# show sbc mysbc sbe sip header-editor HeaderEditor1	• <i>editor-name</i> —Specifies the name of the editor and displays details about the specified editor. If omitted, the command shows information about all the SIP header editors.			
Step 27	<pre>show sbc sbc-name sbe sip method-editor [editor-name]</pre>	Displays the details of all method editors, or displays details pertaining to the specified method editor.			
	Example:	• <i>sbc-name</i> —Specifies the name of the SBC service.			
	Router# show sbc mysbc sbe sip method-editor MethodEditor1	• <i>editor-name</i> —Specifies the name of the editor and displays details about the specified editor. If omitted, the command shows information about all the SIP method editors.			
Step 28	<pre>show sbc sbc-name sbe sip option-editor [editor-name]</pre>	Displays the details of all option editors, or displays details pertaining to the specified option editor.			
	Example:	• <i>sbc-name</i> —Specifies the name of the SBC service.			
	Router# show sbc mysbc sbe sip option-editor OptionEditor1	• <i>editor-name</i> —Specifies the name of the editor and displays details about the specified editor. If omitted, the command shows information about all the SIP option editors.			
Step 29	show sbc sbc-name sbe sip parameter-editor [editor-name]	Displays the details of all parameter editors, or displays details pertaining to the specified parameter editor.			
	Example: Router# show sbc mysbc sbe sip parameter-editor	• <i>sbc-name</i> —Specifies the name of the SBC service.			
	rarameterButtorr	• <i>editor-name</i> —Specifies the name of the editor and displays details about the specified editor. If omitted, the command shows information about all the SIP parameter editors.			

Configuration Examples for SIP Message Editors

This section contains the following examples:

- Method Editor Example, page 23-76
- Header Editor Example, page 23-78
- Body Editor Example, page 23-81
- Option Editor Example, page 23-83
- Parameter Editor Example, page 23-83

Method Editor Example

The following example shows how to configure the test1 method editor and the abcd method type on the SBC2 SBC.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip method-editor test1
Router(config-sbc-sbe-mep-mth)# method abcd
Router(config-sbc-sbe-mep-mth)# blacklist
```

The following example shows how the **show sbc sbe sip method-editor** command is used to display details of the meditor1 method editor and the test1 method editor before they have been applied to an adjacency.

```
Router# show sbc SBC2 sbe sip method-editor meditor1
  method-editor "meditor1"
   Description:
            Whitelist
   Type:
   Methods:
      INVITE
        action as-editor
        map-status-code
         range 5XX value 500
         range 6XX value 600
   Not in use with any adjacencies
Router# show sbc SBC2 sbe sip method-editor test1
 method-editor "test1"
   Description:
    Type: Blacklist
   Methods:
      abcd
       action as-editor
   Not in use with any adjacencies
```

The following example shows how the **show sbc sbe sip method-editor** command is used to display a list of all configured method editors:

Router# show sbc SBC2 sbe sip method-editormethod-editors for SBC service "SBC2"NameIn use==============test1Nomeditor1Nopreset-acc-in-mthNo

preset-std-in-mth	No
preset-acc-out-mth	No
preset-core-in-mth	No
preset-std-out-mth	No
preset-core-out-mth	No
preset-ipsec-in-mth	No
preset-ipsec-out-mth	No
default	No
preset-ibcf-ext-in-mth	No
preset-ibcf-int-in-mth	No
preset-ibcf-utr-in-mth	No
preset-ibcf-ext-out-mth	No
preset-ibcf-int-out-mth	No
preset-ibcf-utr-out-mth	No
preset-std-block-in-mth	No
preset-std-block-out-mth	No

Example—Applying the Method Editor

The **method-editor inbound test1** command applies the test1 method editor on the inbound direction. Therefore, for all incoming messages, the method type abcd is checked. When the abcd method arrives, it is blacklisted and the error code *405 Method Not Allowed* is generated. All the other methods are allowed.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip trans-uac
Router(config-sbc-sbe-adj-sip)# no attach
Router(config-sbc-sbe-adj-sip)# method-editor inbound test1
Router(config-sbc-sbe-adj-sip)# attach
```

The following example shows how the **show sbc sbe sip method-editor** command is used to display details of the test1 method editor after it has been applied to an adjacency.

Router# **show sbc SBC2 sbe sip method-editor** method-editors for SBC service "SBC2"

Name	In	use
	===	====
test1	Υe	es
meditor1	No)

Router# show sbc SBC2 sbe sip method-editor test1
method-editor "test1"
Description:
Type: Blacklist
Methods:
 abcd
 action as-editor
In use by adjacency:trans-uac (in)

Header Editor Example

This section contains the following examples:

- Example—Configuring and Applying the Header Editor, page 23-78
- Example—Using Directory Number Prefix to Set Privacy, page 23-79
- Example—Converting Remote-Party-ID or P-Preferred-Identity, page 23-80

Example—Configuring and Applying the Header Editor

The following example shows how to configure the EXAMPLE header editor and the abcd header type on the SBC2 SBC.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-editor EXAMPLE
Router(config-sbc-sbe-mep-hdr)# blacklist
Router(config-sbc-sbe-mep-hdr)# header abcd
```

The following example shows how the **show sbc sbe sip header-editor** command is used to display details of the EXAMPLE header editor:

```
Router# show sbc SBC2 sbe sip header-editor EXAMPLE
  header-editor "EXAMPLE"
   Description:
   Type:
             Blacklist
   store-rules:
       No store-rule entries found.
   request-line:
       No request-line entries found.
   headers:
     abcd
       entrv 1
         description:
         action as-editor
   Not in use with any adjacencies
   Not in use with any method-editor
```

The **header-editor inbound EXAMPLE** command and the **header-editor outbound EXAMPLE** command applies the EXAMPLE header editor on the inbound and outbound direction. Therefore, for all incoming and outgoing messages, the header type abcd is checked. When the abcd header arrives or leaves, it is blacklisted and the error code 405 Method Not Allowed is generated. All the other headers are allowed.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip trans-uac
Router(config-sbc-sbe-adj-sip)# no attach
Router(config-sbc-sbe-adj-sip)# header-editor inbound EXAMPLE
Router(config-sbc-sbe-adj-sip)# header-editor outbound EXAMPLE
Router(config-sbc-sbe-adj-sip)# attach
```

The following example shows how the **show sbc sbe sip header-editor** command is used to display details of the EXAMPLE header editor after it has been applied to an adjacency.

```
Router# show sbc SBC2 sbe sip header-editor EXAMPLE
```

```
header-editor "EXAMPLE"
Description:
Type: Blacklist
store-rules:
    No store-rule entries found.
request-line:
    No request-line entries found.
headers:
    abcd
    entry 1
        description:
        action as-editor
In use by adjacency:trans-uac (in, out)
Not in use with any method-editor
```

Example—Using Directory Number Prefix to Set Privacy

This example shows how to use a directory number prefix to set privacy:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config) # sbc test
Router(config-sbc) # sbe
Router(config-sbc-sbe) # sip header-editor headprof1
Router(config-sbc-sbe-sip-hdr) # store-rule entry 1
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the called party number from
то"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name To is-tel-uri eq true
Router (config-sbc-sbe-sip-hdr-ele-act) # condition and header-name To tel-uri-user store-as
called-dn
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 2
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the called party number from
то"
Router(config-sbc-sip-hdr-ele-act)# condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name To is-sip-uri eq true
Router (config-sbc-sbe-sip-hdr-ele-act) # condition and header-name To sip-uri-user store-as
called-dn
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 3
Router(config-sbc-sbe-sip-hdr-ele-act)# description "set $privacy based on DN"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable privacy is-defined eq false
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable called_dn is-defined eq
true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable called_dn regex-match
`^184″
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and "none" store-as privacy
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 4
Router(config-sbc-sbe-sip-hdr-ele-act)# description "set $privacy based on DN"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable privacy is-defined eq false
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable called_dn is-defined eq
true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable called_dn regex-match
`^186″
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and "user" store-as privacy
```

Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# header Privacy entry 1
Router(config-sbc-sbe-sip-hdr-ele)# action strip
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr-ele)# exit
Router(config-sbc-sbe-sip-hdr-ele)# action add-first-header value "\${privacy}"
Router(config-sbc-sbe-sip-hdr-ele-act)# description "create a privacy header if we have
privacy info"

Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable privacy is-defined eq true

Associate with an inbound adjacency:

Router# configure terminal
Router(config)# sbc test
Router(config-sbc) sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-sip)# header-editor inbound headprof1

Example—Converting Remote-Party-ID or P-Preferred-Identity

This example converts Remote-Party-ID or From into P-Preferred-Identity. If the message is a request and Remote-Party-ID is present then it stores the username into a variable username. If the From header contains a sip: URI or Tel: URI, and Remote-Part-ID was not present then it stores the username into the variable username. Strips all P-Preferred-Identity, Remote-Party-ID's and P-Preferred-Identity headers and inserts a single P-Preferred-Identity header containing the stored username and a Privacy header based on info received:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config) # sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip header-editor headprof1
Router(config-sbc-sbe-sip-hdr)# store-rule entry 1
Router (config-sbc-sbe-sip-hdr-ele-act) # description "store the RPID username in $username"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name Remote-Party-ID
header-value extract user store-as username
Router(config-sbc-sbe-sip-hdr-ele-act) # exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 2
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the privacy parameter in
$rpid-privacy"
Router(config-sbc-sip-hdr-ele-act) # condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name Remote-Party-ID
header-value extract parameter privacy store-as rpid_privacy
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 3
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the From sip uri in $username"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable username is-defined eq
false
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name From header-uri
is-sip-uri eg true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name From header-uri
sip-uri-user store-as username
Router(config-sbc-sbe-sip-hdr-ele-act)# exit
Router(config-sbc-sbe-sip-hdr)# store-rule entry 4
Router(config-sbc-sbe-sip-hdr-ele-act)# description "store the From tel uri in $username"
Router(config-sbc-sbe-sip-hdr-ele-act)# condition is-request eq true
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable username is-defined eq
false
Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name From header-uri
is-tel-uri eq true
```

Router(config-sbc-sbe-sip-hdr-ele-act)# condition and header-name From header-uri tel-uri-user store-as username Router(config-sbc-sbe-sip-hdr-ele-act) # exit Router(config-sbc-sbe-sip-hdr)# store-rule entry 5 Router (config-sbc-sbe-sip-hdr)# description "convert RPID param into Privacy header value" Router(config-sbc-sbe-sip-hdr)# condition variable rpid_privacy is-defined eq true Router(config-sbc-sbe-sip-hdr)# condition and variable rpid_privacy eq "off" Router(config-sbc-sbe-sip-hdr)# condition and "none" store-as privacy Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr)# store-rule entry 6 Router (config-sbc-sbe-sip-hdr-ele-act) # description "convert RPID param into Privacy header value" Router(config-sbc-sbe-sip-hdr-ele-act) # condition variable rpid_privacy is-defined eq true Router(config-sbc-sbe-sip-hdr-ele-act)# condition and variable rpid_privacy eq "id" Router(config-sbc-sbe-sip-hdr-ele-act)# condition and "user" store-as privacy Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr)# header P-Preferred-Identity entry 1 Router(config-sbc-sbe-sip-hdr-ele)# action strip Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr-ele)# exit Router(config-sbc-sbe-sip-hdr)# header P-Preferred-Identity entry 2 Router(config-sbc-sbe-sip-hdr-ele)# action add-first-header value "<sip:\${username}@mydomain.com;user=phone>" Router(config-sbc-sbe-sip-hdr-ele-act)# description "create a P-Preferred-Identity header" Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable username is-defined eq true Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr-ele)# exit Router(config-sbc-sbe-sip-hdr) # header P-Asserted-Identity entry 1 Router(config-sbc-sbe-sip-hdr-ele)# action strip Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr-ele)# exit Router(config-sbc-sbe-sip-hdr)# header Remote-Party-ID entry 1 Router(config-sbc-sbe-sip-hdr-ele)# action strip Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr-ele)# exit Router(config-sbc-sbe-sip-hdr)# header Privacy entry 1 Router(config-sbc-sbe-sip-hdr-ele)# action strip Router(config-sbc-sbe-sip-hdr-ele-act)# exit Router(config-sbc-sbe-sip-hdr-ele)# exit Router(config-sbc-sbe-sip-hdr)# header Privacy entry 2 Router(config-sbc-sbe-sip-hdr-ele)# action add-first-header value "\${privacy}" Router(config-sbc-sbe-sip-hdr-ele-act)# description "create a privacy header if we have privacy info" Router(config-sbc-sbe-sip-hdr-ele-act)# condition variable privacy is-defined eq true

Associate with an inbound adjacency:

Router# configure terminal Enter configuration commands, one per line. End with CNTL/Z. Router(config)# sbc test Router(config-sbc) sbe Router(config-sbc-sbe)# adjacency sip adj1 Router(config-sbc-sbe-sip)# header-editor inbound headprof1

Body Editor Example

The following example shows how to configure the beditor body editor on the SBC2 SBC:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip body-editor beditor1
```

```
Router(config-sbc-sbe-mep-bdy)# body dtmf-relay/mixed
Router(config-sbc-sbe-mep-bdy-ele)# action reject
```

The following example shows how the **show sbc sbe sip body-editor** command is used to display details of the beditor1 body editor:

Router# show sbc SBC2 sbe sip body-editor beditor1

```
body-editor "beditor1"
Description:
Bodies:
    dtmf-relay/mixed
    action reject
    hunt-on-reject false
Not in use with any adjacencies
Not in use with any method-editor
```

Example—Applying Body Editor

The **body-editor inbound beditor1** command and the **body-editor outbound beditor1** command applies the beditor1 body editor on the inbound and outbound direction.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip trans-uac
Router(config-sbc-sbe-adj-sip)# no attach
Router(config-sbc-sbe-adj-sip)# body-editor inbound beditor1
Router(config-sbc-sbe-adj-sip)# body-editor outbound beditor1
Router(config-sbc-sbe-adj-sip)# body-editor outbound beditor1
Router(config-sbc-sbe-adj-sip)# attach
```

The following examples shows how the **show sbc sbe sip body-editor** command is used to display details of the beditor1 body editor after it has been applied to an adjacency:

Router# show sbc SBC2 sbe sip body-editor beditor1
body-editor "beditor1"
Description:
Bodies:
 dtmf-relay/mixed
 action reject
 hunt-on-reject false
In use by adjacency:trans-uac (in, out)
Not in use with any method-editor

Option Editor Example

The following example shows how to configure the oeditor1 option editor on the SBC2 SBC:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip option-editor oeditor1
Router(config-sbc-sbe)# option opt
```

Example—Applying Option Editor

The **option-editor inbound oeditor1** command and the **option-editor outbound oeditor1** command applies the oeditor1 option editor on the inbound and outbound direction.

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip trans-uac
Router(config-sbc-sbe-adj-sip)# no attach
Router(config-sbc-sbe-adj-sip)# option-editor ua inbound oeditor1
Router(config-sbc-sbe-adj-sip)# option-editor ua outbound oeditor1
Router(config-sbc-sbe-adj-sip)# option-editor ua outbound oeditor1
Router(config-sbc-sbe-adj-sip)# attach
```

The following shows how the **show sbc sbe sip option-editor** command is used to display details of the oeditor1 option editor:

```
Router# show sbc SBC2 sbe sip option-editor oeditor1

option-editor "oeditor1"

Description:

Type: Whitelist

Options:

opt

In use by adjacency:ASR-15 (in-ua)
```

Router	f show	sbc S	BC2 s	sbe s	sip	optio	n-edi	tor
option	editor	rs for	SBC	serv	vice	"SBC	2 "	
Name						In	use	
=====			====:	=====		=====	===	
opt						No	•	
oedit	cor1					Ye	s	

Parameter Editor Example

The following example shows how to configure the peditor1 parameter editor on the SBC2 SBC:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip parameter-editor peditor
Router(config-sbc-sbe-mep-prm)# parameter param
Router(config-sbc-sbe-mep-prm-ele)# action strip
```

Example—Applying Parameter Editor

The following example shows how to apply the peditor parameter editor to the hel header editor on the SBC2 SBC:

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config) # sbc SBC2
Router(config-sbc)# sbe
Router(config-sbc-sbe) # sip header-editor he1
Router(config-sbc-sbe-mep-hdr) # header Subject
Router(config-sbc-sbe-mep-hdr-ele)# parameter-editor peditor
The following shows how the show sbc sbe sip header-editor command is used to display
details of the hel header editor:
Router# show sbc SBC2 sbe sip header-editor he1
 header-editor "he1"
   Description:
                 Whitelist
    Type:
    store-rules:
       No store-rule entries found.
    request-line:
       No request-line entries found.
   headers:
      subject
        entry 1
          description:
          action as-profile
            parameter-profile peditor
```

The following example shows how the **show sbc sbe sip parameter-editor** command is used to display details of the peditor parameter editor:

```
Router# show sbc SBC2 sbe sip parameter-editor peditor
parameter-editor "peditor"
Description:
Parameters:
param
action strip
In use by header-editor:he1, header:subject, entry:1
```

SDP Editing Using Script-Based Editors



This section describes script-based editors for modifying the SDP content in SIP messages. The ?\$paranum>SIP Message Editing Using Editors? section on page 23-64 describes body, header, method, option, and parameter editors that you directly configure on the SBC. You can apply any combination of script-based editors and directly configured editors to edit SIP messages.

From Release 3.4S, you can use scripts written using the Lua programming language to modify the SDP content in SIP messages. Typically, a Lua script consists of a group of one or more related functions. In the context of the SIP Message Editing feature, you write these functions with the objective of editing SIP messages. For detailed information about the Lua programming language, visit the Lua website at http://www.lua.org/.
You can configure a set of Lua scripts on the SBC. A set of scripts describes a set of editing actions to be applied to SIP messages. While configuring a script set, you specify the order in which scripts are loaded in the script set.

You can register the message-editing functions in the scripts as editors. These editors are called by the SBC at run time and applied to SIP messages. You can use these editors in conjunction with the body, header, method, option, and parameter editors configured on the SBC.

After you configure a script set, you can perform isolation testing and live testing on the script set to ensure that it works as expected.

At any point of time, only one script set can be active on the SBC. However, multiple script sets can be defined and kept ready for future use. You can switch a script set from the active state to the inactive state according to your requirements and vice versa.

The following sections provide information about creating Lua scripts and configuring script-based editing:

- Creating Lua Scripts for Script-Based Editing, page 23-85
- Configuring Script-Based Editors on the SBC, page 23-91
- Creating and Configuring Script-Based Editors: Examples, page 23-99

Creating Lua Scripts for Script-Based Editing

The following sections provide information that you can use while creating Lua scripts for script-based editing:

- Built-in Lua Classes, page 23-85
- Built-in Application Variables, page 23-89
- Built-in Logger Functions, page 23-90
- Built-in Register Function, page 23-90
- User-Defined Application Variables, page 23-91

Built-in Lua Classes

Lua scripts use an XPath-compatible method of referring to each node within the SDP body of a SIP message. The following example shows a sample SDP body in XML format. In the Lua code that you write, each XML tag can be uniquely identified by its path. A syntax-based function (such as the MeBlock:select_by_syntax function that is explained in the ?\$paranum>MeBlock Class? section on page 23-86) can accept and process data based on the path that is passed to the function. A path is a forward-slash-separated string. For example, the sdp/media[1]/line[3] path identifies the third line in the first media tag. Therefore, the sdp/media[1]/line[3] path refers to b=AS:64.

```
<sdp>
<sdp>
line>v=0</line>
line>o=user1 12345 12346 IN IP4 192.0.2.27</line>
line>s=-</line>
line>c=IN IP4 0.0.0.0</line>
line>t=0 0</line>
line>r=604800 3600 0 90000</line>
line>r=7d 1h 0 25h</line>
line>a=foo</line>
media>
<line>m=audio 32768 RTP/AVP 0 101</line>
line>c=IN IP4 0.0.0.0</line>
```

L

```
<line>b=AS:64</line>
<line>a=rtpmap:0 PCMU/8000</line>
<line>a=rtpmap:101 telephone-event/8000</line>
<line>a=ptime:20</line>
</media>
<media>
<line>m=video 32770 RTP/AVP 112</line>
<line>a=rtpmap:112 mpeg4-generic/48000</line>
</media>
</sdp>
```

You can use the following built-in Lua classes when writing scripts to modify the SDP body of SIP messages.

MeMsg Class

An object of the MeMsg class contains the top-level structure of the message, which in turn, contains the entire SIP message. Table 23-4 describes the functions of this class.

Table 23-4	Functions of the	MeMsg Class
------------	------------------	-------------

Function	Description
:get_sdp() or .sdp	Returns the MeBlock object that holds the SDP body.
:get_current_edit_point	Returns the current edit point, which is either before-receive or after-send.
:reject(error_code)	Fails a SIP request, or discards the response.
:get_app_variables() or .app_variable	Returns the table of application variables.

MeBlock Class

An object of the MeBlock class represents a node in the SDP tree. A block is a contiguous subset of the SDP that is used for accessing strings. Table 23-5 describes the functions of the MeBlock class.

 Table 23-5
 Functions of the MeBlock Class

Function	Description
.new(syntax)	Constructs a block using the given syntax.
:get_type() or .type	Returns the syntax type (line, media, or sdp) of the MeBlock object.
:get_parent() or .parent	Returns the parent of this MeBlock object, which is either another MeBlock object or NIL for the root.
:get_children() or .children	Returns a MeSelection object that contains the child elements of the block.
:select_by_prefix(text_prefix)	Returns a MeSelection object containing all the lines of the MeBlock object that have the specified prefix.

Function	Description
:select_by_syntax(syntax_path)	Returns a MeSelection object containing sub-blocks that conform to the specified syntax path. Here, syntax refers to the block type (that is, line, media, and so on).
:insert_child_last(new_block)	Inserts a MeBlock object below this MeBlock object, after all the existing child objects.
:insert_child_before(new_block, sibling)	Inserts a MeBlock object below this MeBlock object, before the specified existing child object.
:insert_child_after(new_block, sibling)	Inserts a MeBlock object below this MeBlock object, after the specified existing child object.
:delete()	Deletes this MeBlock object.
:delete_children()	Deletes all the sub-blocks of this MeBlock object, and leaves the object empty.

Table 23-5 Functions of the MeBlock Class (continued)

MeSelection Class

An object of the MeSelection class is a list of MeBlock objects. Objects of the MeSelection class are used to process a set of lines. They can also be used to process child blocks in a parent block. A MeSelection object sequences MeBlock objects in the order in which they appear in the message. Table 23-6 describes the functions of the MeSelection class.

Table 23-6Functions of the MeSelection Class

Function	Description
:empty()	Returns TRUE if this selection is empty.
:iter()	Returns a generic For iterator that performs the specified action on all the objects in the MeSelection object. Each object is either of the MeBlock class or one of its subclasses.
[] operator	Use this one-based array operator to get the <i>n</i> th block in the MeSelection object. Negative array indexes are also supported.

MeTextBlock Class

An object of the MeTextBlock class is used to assign, create, or manipulate text. Table 23-7 describes the functions of this class.

Function	Description
.new(type,text)	Constructs a new block of a specific type (line, media, and so on) using the specified text.
:get_text() or .text	Returns the text of the MeTextBlock object.

Function	Description
:set_text()	Sets the text of the MeTextBlock object. Note that existing text is replaced when this function is called.
:find(args)	Calls string.find(args) on the text of this MeTextBlock object.
:match(args)	Calls string.match(args) on the text of this MeTextBlock object.
:replace(args)	Calls string.gsub(args) on the text of this MeTextBlock object.

Table 23-7 Functions of the MeTextBlock Class (continued)

MeSdp Class

An object of the MeSdp class is used to retrieve specific parts of the SDP body. Table 23-8 describes the functions of this class.

Table 23-8Functions of the MeSdp Class

Function	Description
:get_session_lines() or .session_lines	Returns a MeSelection object containing the session lines of the SDP body.
:get_media_blocks() or .media_blocks	Returns a MeSelection object containing the media blocks.

MeSdpMedia Class

An object of the MeSdpMedia class is used to create or retrieve SDP media blocks. Table 23-9 describes the functions of this class.

 Table 23-9
 Functions of the MeSdpMedia Class

Function	Description
.new(text)	Constructs a block of the media syntax (or block type) using the specified text.
:get_media_lines() or .media_lines	Returns a MeSelection object containing the media lines of the MeSdpMedia object.

MeSdpLine Class

An object of the MeSDPLine class is used to create a line in the SDP message. Table 23-10 describes the functions of this class.

Table 23-10Functions of the MeSdpLine Class

Function	Description
.new(text)	Constructs a block of the line syntax (or block type) using the specified text.

Built-in Application Variables

This section describes the built-in application variables that you can use while writing Lua scripts.

Built-in application variables, such as configuration data for an adjacency and transport values, are initialized by the SBC and are available to the Lua scripts. They are read-only, start with the characters msg. or adj., and are reserved because you cannot create variables with these prefixes.

Table 23-11 describes the built-in application variables that can be accessed within a script.

Variable	Description
adj.account	Adjacency account.
adj.dest_addr	Adjacency signaling peer.
adj.dest_port	Adjacency signaling peer port.
adj.group	Adjacency group.
adj.home_net_id	Adjacency home network identity.
adj.ip_realm	Adjacency realm.
adj.lcl_addr	Adjacency signaling address.
adj.lcl_port	Adjacency signaling port.
adj.lcl_id	Adjacency local ID.
adj.listen_trans	Adjacency listen transport.
adj.mandatory_trans	Adjacency mandatory transport.
adj.med_loc	Adjacency media location.
adj.name	Adjacency name.
adj.preferred_trans	Adjacency preferred transport.
adj.trust_level	Adjacency trust level.
adj.target_reg_addr	Adjacency registration target address.
adj.targrt_reg_port	Adjacency registration target port.
adj.visited_net_id	Adjacency visited network identity.
adj.vpn_id	Adjacency VPN ID.

Table 23-11 Built-in Application Variables

Variable	Description
msg.status_code	Response status code. Returns the string representation of the status code for a SIP response message. Returns an empty string for a SIP request message.
msg.header("name").value	Value of the first header with the name <i>name</i> in the message. Only nonvital headers can be used with this function.
msg.header("name").uri.tel_uri.number	Directory number of the TEL URI in the first header with the name <i>name</i> . If used on a SIP URI, an empty string is returned. Only To and From headers can be used with this function.
msg.header("name").uri.sip_uri.user	User name of the SIP URI or SIPS URI in the first header with the name <i>name</i> . If used on a TEL URI, an empty string is returned. Only To and From headers can be used with this function.
msg.lcl_ip_addr	Address at which the message was received.
msg.lcl_port	Port at which the message was received.
msg.rmt_ip_addr	Previous hop IP address.
msg.rmt_port	Previous hop port.

Table 23-11	Built-in Application	Variables
-------------	----------------------	-----------

Built-in Logger Functions

The following logger functions can be called to create logs:

- MeLogger.debug(text) Log at debug level (30)
- MeLogger.detail(text) Log at detail level (50)
- MeLogger.info(text) Log at info level (60)
- MeLogger.config(text) Log at config level (63)
- MeLogger.warn(text) Log at warn level (70)
- MeLogger.error(text) Log at error level (80)

Built-in Register Function

Use the following function to register functions as editors with the SBC:

MeEditor.register(edit_point,editor_name,edit_func)

By including this line in the script, you can register a function as an editor with the SBC, assign the function a name as an editor, and specify the point at which the function must be applied as an editor on SIP messages.

The following are the arguments of the MeEditor.register function:

- edit_point—Accepts one of the following values:
 - AFTER_SEND—Specifies that the outgoing message must be edited after it is processed by the adjacency and just before it is forwarded from the adjacency.

- BEFORE_RECEIVE—Specifies that the incoming message must be edited just after it is received on the adjacency and before the adjacency begins processing it.
- editor_name—Specifies the name that you want to assign to the editor.

```
<u>Note</u>
```

Names that you assign to editors in a script set must be unique. However, editors in different script sets can have the same name.

• *edit_func* is the name of the function in the script that you want to designate as an editor.

The following example shows how to register the hello_world Lua function as an editor:

User-Defined Application Variables

User-defined application variables are used to pass user information among Lua edit functions and between script-based editors and editors that are directly configured on the SBC that is, body, header, method, option, and parameter editors.

Configuring Script-Based Editors on the SBC

This task shows how to configure a script-based editor on the SBC.

Note

Before you start performing this task, create the scripts offline and place the script files at a location from where they can be accessed from the SBC. Copy the script files to the SBC by using trivial file transfer protocol (TFTP), file transfer protocol (FTP), remote copy protocol (rcp), secure copy protocol (SCP), or any other supported application.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. script-set set-number lua
- 5. script script-name
- 6. load-order order-index-number
- 7. type {full | wrapped edit-point {before-receive | after-send | both}}
- 8. filename {bootflash: | flash: | fpd: | nvram: | obfl: | *any-other-device* }
- 9. exit
- 10. complete
- 11. end
- **12.** test sbc message sip filename *device-type:file-name* script-set *script-set-number* {after-send | before-receive} editors {*editor1-name* [*editor2-name*] [*editor3-name*] . . . [*editor8-name*]}

- 13. configure terminal
- 14. sbc sbc-name
- 15. sbe
- **16.** adjacency sip *adjacency-name*
- 17. test script-set set-number
- 18. exit
- 19. active-script-set script-set-number
- 20. adjacency sip adjacency-name
- 21. editor-list {after-send | before-receive}
- 22. editor *order-number editor-name* [condition [body contains sdp]]
- 23. end
- 24. show sbc *sbc-name* sbe script-set *set-number* [script *script-name* [line-numbers] | program [line-numbers] | statistics]
- 25. clear sbc sbc-name sbe script-set-stats set-number [editors-stats editor-name]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	• <i>sbc-name</i> —Name of the SBC.
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	
Step 4	script-set set-number lua	Configures a script set composed of scripts written using the Lua programming language.
	Example: Router(config-sbc-sbe)# script-set 20 lua	• <i>set-number</i> —Script set number. The range is from 1 to 2147483647.
		Enters the script-set configuration mode.
Step 5	script script-name	Configures a script in the script set. Note that multiple scripts can be configured in a script set.
	Example:	• <i>script-name</i> —Name of the script.
	Router(config-sbc-sbe-script-set)# script SBCScript	Enters the script configuration mode.
Step 6	<pre>load-order order-index-number Example: Router(config-sbc-sbe-scrpset-script)# load-order 2</pre>	Specifies the load order of the script. Scripts are loaded in ascending order of the order index number. For example, a script with the order index number 4 is loaded before a script with the order index number
		 order-index-number—Load order index number. The range is from 1 to 4294967295. The default order index number is 100. For scripts that are subsequently added and for which the load-order command is not run, the default order index number is set in multiples of 100 (that is, 200, 300, 400, and so on).

Step 7	<pre>type {full wrapped edit-point {after-send before-receive both}}</pre>	Specifies the script type:
		• full —Specifies a full script without
	<pre>Example: Router(config-sbc-sbe-scrpset-script)# type full</pre>	 wrapped edit-point—Specifies a script that must be autogenerated from the file and the edit point to be used in autoregistration. One of the following edit points can be specified:
		 after-send—Specifies that the outgoing message must be edited after the message is processed by the adjacency and just before it is forwarded from the adjacency.
		 before-receive—Specifies that the incoming message must be edited just after it is received on the adjacency and before the adjacency begins processing it.
		 both—Enables editing of the message both before and after it is processed by the SBC.
Step 8	<pre>filename {device-type:file-path-and-name}</pre>	Specifies the path and name of the script file.
	Example:	• <i>device-type</i> —One of the following or any other storage device installed on the router:
	Router(config-sbc-sbe-scrpset-script)# filename bootflash:lua1.lua	– bootflash:
		– flash:
		– fpd:
		– nvram:
		– obfl:
		The list of file system devices is dynamically generated and displayed. Other devices, such as a hard disk, that are available on the platform can also be used in this command.
		• <i>file-path-and-name</i> —Full path and name of the script file on the specified storage device.
Step 9	exit	Exits the script configuration mode and enters the script-set configuration mode.
	Example:	
Stop 10	Router(config-sbc-sbe-scrpset-script)# exit	Yalidatas and loo do the equipte
Step IU	complete	Validates and loads the scripts.
	Example: Router(config-sbc-sbe-script-set)# complete	If syntax errors are encountered during the validation process, error messages are displayed. If a script is syntactically correct, it is loaded into memory and the editors are registered with the Lua run-time environment. If required, you can switch to the privileged EXEC mode and then run the show sbc sbe editors command to verify that the editors are

	Command or Action	Purpose
Step 11	end	Exits the script-set configuration mode, and returns to the privileged EXEC mode.
	Example: Router(config-sbc-sbe-script-set)# end	
Step 12	test sbc message sip filename device-type:file-name script-set script-set-number {after-send	Performs isolation testing of script-based editors.
	<pre>[editor3-name] editors {editor1-name [editor2-name] [editor3-name] [editor8-name] } Example: Router# test sbc message sip filename bootflash:inv script-set 123 after-send editors sdp_add_after</pre>	testing, we recommend that you perform the procedure. See the ?\$paranum>Isolation Testing of Script-Based Editors: Example? section on page 23-100 for detailed information about the procedure.
	my-header-editor	• <i>device-type</i> —Type of storage device on which you have stored the file containing the SIP message on which you want to test the editors. In the command-line interface (CLI), when you enter a question mark after the test sbc message sip filename script-set editors command, a list of all the storage devices installed on the router is displayed. The device can be one of the following or any other storage device installed on the router:
		bootflash:
		flash:
		fpd:
		nvram:
		obfl:
		The list of file system devices is dynamically generated and displayed. Other devices, such as a hard disk, that are available on the platform can also be used in this command.
		• <i>file-name</i> —Name of the file containing the SIP message on which you want to test the editors.
		• <i>script-set-number</i> —Number of the script set containing the editors that you want to test.
		• after-send —Specifies that the outgoing message must be edited after the message is processed by the adjacency and just before it is forwarded from the adjacency.
		• before-receive —Specifies that the incoming message must be edited just after it is received on the adjacency and before the adjacency begins processing it.
		• <i>editor1-nameeditor8-name</i> —Names of the editors. You can specify up to eight editors. You must specify at least one editor.

Command or Action Purpose	
Step 13 configure terminal Enters the global configuration is	mode.
Example: Router# configure terminal	
Step 14sbc sbc-nameEnters the SBC service mode.	
• <i>sbc-name</i> —Name of the SB Router(config)# sbc mysbc	SC.
Step 15 sbe Enters the SBE configuration me	ode of the SBC.
Example: Router(config-sbc)# sbe	
Step 16 adjacency sip adjacency-name Enters the SBE SIP adjacency control	onfiguration mode.
• <i>adjacency-name</i> —Name of Example: Router(config-sbc-sbe)# adjacency sip adj1	the adjacency.
Step 17 test script-set script-set-number Performs live testing of script-be	ased editors.
Example: Note Although it is optional to we recommend that you procedure. See the ?\$par of Script-Based Editors: page 23-102 for detailed • script-set-number—Script set) perform live testing, perform the ranum>Live Testing Example? section on d information.
Step 18 exit Exits the SIP adjacency configuration	ration mode.
Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 19active-script-setscript-set-numberActivates the script set.	
• <i>script-set-number</i> —Script s	set number.
Example: Note When you run the active-script-set 20 Router(config-sbc-sbe)# active-script-set 20 Note When you run the active-for a particular script set was previously active au the inactive state. The example: script set are no longer a messages. You can switch set to the active state by active-script-set command Set to the active state by active-script-set command	-script-set command t, the script set that itomatically goes to ditors of an inactive applied to SIP ch an inactive script running the and on it.
Step 20 adjacency sip adjacency-name Enters the SBE SIP adjacency control	onfiguration mode.
• adjacency-name—Name of Example: Router (config-shc-she)# adjacency sin adj1	the adjacency.

	Command or Action	Purpose
Step 21	editor-list {after-send before-receive}	Configures a list of editors.
	Example: Router(config-sbc-sbe-adj-sip)# editor-list after-send	• after-send —Specifies that the outgoing message must be edited after the message is processed by the adjacency and just before it is forwarded from the adjacency.
		• before-receive —Specifies that the incoming message must be edited just after it is received on the adjacency and before the adjacency begins processing it.
		Enters the SIP editor configuration mode.
Step 22	<pre>editor order-number editor-name [condition [body contains sdp]] Example:</pre>	Configures an editor in the editor list. For each editor that you want to apply in a sequence, run this command to specify the order of the editor in the editor list.
	Router(config-sbc-sbe-adj-sip-ed)# editor 2 sdp_add_after condition body contains sdp	Note You can add any combination of script-based editors and body, header, method, option, and parameter editors in the editor list.
		• <i>order-number</i> —Order in which the editor must be applied. The range is from 1 to 2147483647.
		• <i>editor-name</i> —Name of the editor that you want to apply to messages that are processed by the adjacency.
		• condition —Specifies that there are one or more conditions for the editor to be applied.
		• body contains sdp —Specifies that the message body must be SDP-based content. The editor is applied only if this condition is met. Include body contains sdp in the command for script-based editors.
Step 23	end	Exits the SIP editor configuration mode, and enters the privileged EXEC mode.
	Example: Bouter(config-sbc-sbe-adi-sip)# end	

	Command or Action	Purpose
Step 24	<pre>show sbc sbc-name sbe script-set set-number [script script-name [line-numbers] program [line-numbers] statistics]</pre>	Displays a summary of the details pertaining to all the configured script sets or shows the details of the specified script set.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router# show sbc mysbc sbe script-set 20 script SBCscript line-numbers	• <i>set-number</i> —Script set number. The range is from 1 to 2147483647.
		• program —Specifies that all scripts must be displayed as a single program.
		• line-numbers —Specifies that line numbers must be included while displaying scripts.
		• script —Specifies that details of a single script from the script set must be displayed.
		• <i>script-name</i> —Name of the script that must be displayed.
		• statistics —Specifies that script set statistics must be displayed.
Step 25	clear sbc sbc-name sbe script-set-stats script-set-number [editors-stats editor-name]	Clears previously stored statistics related to a script set.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router# clear sbc mysbc sbe script-set-stats 1	• <i>script-set-number</i> —Script set number. The range is from 1 to 2147483647.
		• editor-stats—Specifies that the script set statistics must be cleared for a specific editor.
		• <i>editor-name</i> —Name of the editor.

The following example shows how the **show sbc sbe script-set** command is used to display the summary of a script set:

```
      Router# show sbc mySbc sbe script-set 1

      name
      language
      complete
      active status

      script-set 1
      lua
      yes
      no
      ok

      script
      order
      filename

      edit_invite_1
      1
      bootflash:lua_1.lua

      edit_invite_2
      2
      bootflash:lua_2.lua

      edit_invite_3
      3
      bootflash:lua_3.lua
```

Creating and Configuring Script-Based Editors: Examples

The following sections describe how to create and configure sample script-based editors:

- Creating Lua Scripts: Example, page 23-99
- Configuring Script-Based Editors: Example, page 23-100
- Isolation Testing of Script-Based Editors: Example, page 23-100
- Live Testing of Script-Based Editors: Example, page 23-102

Creating Lua Scripts: Example

The following sections provide listings of sample Lua scripts:

- Adding Text in the SDP Body: Example, page 23-99
- Deleting Lines from the SDP Body: Example, page 23-100
- Replacing Text in the SDP Body: Example, page 23-100

Adding Text in the SDP Body: Example

The following example shows a Lua script that is used to add sdp_add_after added this line at the end of the SDP body:

```
function append_text(msg)
  msg.sdp:insert_child_last(MeSdpLine.new("sdp_add_after added this line"))
end
```

The following example shows the line that registers the append_text Lua function from the preceding example as an editor. In this example, the editor is named sdp_add_after.

MeEditor.register(MeEditor.BEFORE_RECEIVE, "sdp_add_after", append_text)

```
<u>Note</u>
```

An editor is registered with the SBC when the script set containing the script with the editor code is configured on the SBC.

The complete code listing for this script is as follows:

```
function append_text(msg)
msg.sdp:insert_child_last(MeSdpLine.new("sdp_add_after added this line"))
end
MeEditor.register(MeEditor.BEFORE_RECEIVE, "sdp_add_after", append_text)
```

You can save these lines in a .lua file, copy the file to the SBC, and then perform the procedure described in the ?\$paranum>Configuring Script-Based Editors on the SBC? section on page 23-91 to configure and test the editor.

Deleting Lines from the SDP Body: Example

The following script deletes all the lines that start with a=deleteme from the SDP bodies of SIP messages:

```
function delete_lines(msg)
for line in msg.sdp:select_by_prefix("a=deleteme"):iter() do
    line:delete()
    end
end
MeEditor.register(MeEditor.BEFORE_RECEIVE, "Delete_a_Lines", delete_lines)
```

Replacing Text in the SDP Body: Example

The following script replaces rtpmap in the SDP body with srtp_remap:

```
function replace_text(msg)
  msg.sdp:replace("rtpmap","srtp_remap")
end
MeEditor.register(MeEditor.AFTER_SEND,"Switch_Protocol",replace_text)
```

Configuring Script-Based Editors: Example

The following example shows how to configure the script set created in the preceding example:

```
Router# configure terminal
Router(config) # sbc mysbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# script-set 20 lua
Router(config-sbc-sbe-script-set)# script SBCScript
Router(config-sbc-sbe-scrpset-script)# load-order 2
Router(config-sbc-sbe-scrpset-script)# type full
Router(config-sbc-sbe-scrpset-script)# filename bootflash:lua1.lua
Router(config-sbc-sbe-scrpset-script)# exit
Router(config-sbc-sbe-script-set)# complete
Router(config-sbc-sbe-script-set)# end
Router# test sbc message sip filename bootflash:inv script-set 123 after-send editors
sdp_add_after my-header-editor
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc) # sbe
Router(config-sbc-sbe) # adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# test script-set 123
Router(config-sbc-sbe-adj-sip)# exit
Router(config-sbc-sbe) # active-script-set 20
Router(config-sbc-sbe) # adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# editor-list after-send
Router(config-sbc-sbe-adj-sip-ed)# editor 2 sdp_add_after condition body contains sdp
Router(config-sbc-sbe-adj-sip)# end
Router# show sbc mysbc sbe script-set 20 script SBCscript line-numbers
```

Isolation Testing of Script-Based Editors: Example

During isolation testing of script-based editors, the SDP editing configuration is tested in isolation. No other form of SBC processing takes place. Isolation testing does not show interactions between the editing configuration and other configurations, such as, number validation configuration.

The **test sbc message** command is used to perform isolation testing on SIP messages. This command loads a file containing a valid protocol message and applies a list of user-specified editors to the message. It does not display details of interactions between editing and routing decisions. Up to eight editors can

be specified in the command. The order in which the editors are specified is the order in which they are applied. Note that profile editors that are not part of any specific script set can also be specified in the command.

In the following example, sdp_add_after is defined in script-set 123 and my_header_editor has been configured using the **sip header-editor** command. The sdp_add_after editor is the one used in the preceding sections describing examples. The lines highlighted in bold show the actions performed by the editors.

```
Router# test sbc message sip filename bootflash:inv script-set 123 after-send editors sdp_add_after my-header-editor
```

```
INVITE sip:john@example.com:55060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.195;branch=z9hG4bKff9b46fb055c0521cc24024da96cd290
Via: SIP/2.0/UDP 192.0.2.195:55061;branch=z9hG4bK291d90e31a47b225bd0ddff4353e9c
c0
From: <sip:192.0.2.195:55061;user=phone>;tag=GR52RWG346-34
To: "john@example.com" <sip:john@example.com:55060>
Call-ID: 12013223@192.0.2.195
CSeq: 1 INVITE
Contact: <sip:192.0.2.195:5060>
Content-Type: application/sdp
Content-Length:
                 229
v=0
o=Clarent 120386 120387 IN IP4 192.0.2.196
s=Clarent C5CM
c=IN IP4 192.0.2.196
t = 0 0
m=audio 40376 RTP/AVP 8 18 4 0
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:0 PCMU/8000
a=SendRecv
%Test successful, edited message:
INVITE sip:john@example.com:55060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.195; branch=z9hG4bKff9b46fb055c0521cc24024da96cd290
Via: SIP/2.0/UDP 192.0.2.195:55061;branch=z9hG4bK291d90e31a47b225bd0ddff4353e9c
с0
From: <sip:192.0.2.195:55061;user=phone>;tag=GR52RWG346-34
To: "john@example.com" <sip:john@example.com:55060>
Call-ID: 12013223@192.0.2.195
CSeq: 1 INVITE
Contact: <sip:192.0.2.195:5060>
Content-Type: application/sdp
Content-Length: 258
name: cisco
v = 0
o=Clarent 120386 120387 IN IP4 192.0.2.196
s=Clarent C5CM
c=IN IP4 192.0.2.196
t=0 0
m=audio 40376 RTP/AVP 8 18 4 0
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:0 PCMU/8000
a=SendRecv
sdp_add_after added this line
```

Live Testing of Script-Based Editors: Example

During live testing of script-based editors, an adjacency is configured as a test adjacency. Inbound editing and outbound editing of messages on that adjacency are then performed using the script set specified in the **test script-set** command instead of the script set that is currently active. The following is a sample command:

```
Router(config-sbc-sbe-adj-sip) # test script-set 123
```



The active script set is specified by the **active-script-set** command. You must ensure that the **active-script-set** command has not been run on the script set on which you run the **test script-set** command.

The **test script-set** command cannot be used to verify profile editors because the profile editors are not associated with a script set. To include a profile editor in the test, first configure the profile editor on the test adjacency by using the **editor-list** command.



Signaling Congestion Handling

Cisco Unified Border Element (SP Edition) supports signaling congestion handling to improve performance when external events can cause large bursts of user activity that exceed the capacity of the SBC. Previously, SBC discarded packets in these situations causing the sending endpoint to retransmit the packet, which increases the load on the system, increasing the latency or drop-rate further. Only a small proportion of calls succeed (much less than the rated capacity) and take a significant length of time to connect.

With congestion handling enhancements, SBC improves the successful call setup and registration rate under loads up to at least double its rated capacity. This is done by rejecting SIP calls or REGISTERs that cannot be processed to prevent retransmissions. The reject message contains a random RETRY-AFTER header that informs the sending endpoint when to send a retry.

The main advantages of rejecting incoming work are to:

- prevent retransmissions
- keep the latency of the system at an acceptable level



This feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for Signaling Congestion Handling

Release	Modification
Cisco IOS XE Release 2.5	This feature was introduced on the Cisco IOS XR.

Contents

This module contains the following sections:

- Restrictions for Signaling Congestion Handling, page 24-2
- Configuring Signaling Congestion Handling, page 24-2

Restrictions for Signaling Congestion Handling

The following restrictions apply when you configure the congestion handling enhancements on the Cisco Unified Border Element (SP Edition):

- SBC supports signaling congestion handling only at the global SBC congestion level. Congestion handling does not improve flow control from particular work-sources or sinks.
- Signaling congestion handling addresses SIP signaling workloads from out-of-dialog requests and only tests INVITE and REGISTER messages.

Configuring Signaling Congestion Handling

Signaling congestion handling is turned on by default; nevertheless it could be configured to change the reject message code.

Note

The reject message code is the code sent back to sender during congestion. Default reject message code is 503.

Cisco Unified Border Element (SP Edition) requires following configurations to enable signaling congestion handling enhancements:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. congestion sip reject-code valid-reject-code
- 5. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# config terminal	
Step 2	sbc sbc-name	Enters session border controller (SBC) configuration submode.
	Example: Router(config)# sbc test	
Step 3	sbe	Enters signaling border element (SBE) configuration submode.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	<pre>congestion sip reject-code valid-reject-code</pre>	Changes the reject message code for congestion handling. The default reject message code is 503.
	Example: Router(config-sbc-sbe)# congestion sip rejeact-code 350	
Step 5	exit	
	Example: Router(config-sbc-sbe)# exit	





SIP IP-FQDN URI Translation

Cisco Unified Border Element (SP Edition) supports translation between IP addresses and fully-qualified domain names (FQDNs) in the Request-URI, To header, and From header in Session Initiation Protocol (SIP) messages, permitting SBC to interconnect SIP devices, expecting specific SIP URI formats.

Note

This feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for IP to FQDN URI Translation Support

Release	Modification
Cisco IOS XE Release	This feature was introduced on the Cisco IOS XR.
2.5	

Contents

This module contains the following sections:

- Restrictions for SIP IP-FQDN URI Translation, page 25-1
- Information About SIP IP-FQDN URI Translation, page 25-2
- Configuring SIP IP-FQDN URI Translation, page 25-3
- Configuration Example for SIP IP-FQDN URI Translation, page 25-4

Restrictions for SIP IP-FQDN URI Translation

The following restrictions apply when you configure the SIP IP-FQDN URI translation on the Cisco Unified Border Element (SP Edition):

- Each IP address and FQDN required for translation must be explicitly configured on Cisco Unified Border Element (SP Edition).
- Cisco Unified Border Element (SP Edition) configures only one mapping for each IP address and FQDN.

 The FQDN must be unique when a bidirection mapping is configured. SBC can configure multiple IP addresses that map to a single FQDN (IP1 -> FQDN1; IP2 -> FQDN1), but it can not configure a single FQDN to map to multiple IP addresses.

Information About SIP IP-FQDN URI Translation

URI Translation

The URIs in the Request-URI, To header, and From header are translated based on the configuration of the ingress and egress adjacency.

The domain part of the SIP or SIPs URI is converted after the translation is configured and a mapping is found in the IP-to-FQDN mapping table. If no mapping is found, the SIP request is forwarded without any modification in the domain part.

For example, the following INVITE (with only the relevant parts shown):

```
INVITE sip:conf-server@12.34.56.78 SIP/2.0
From: End-User <sip:end-user@100.101.102.103>;tag=5678-EFGH
To: Conf-Server <sip:conf-server@12.34.56.78>
```

can be converted to:

```
INVITE sip:conf-server@example1.com SIP/2.0
From: End-User <sip:end-user@example2.com>;tag=5678-EFGH
To: Conf-Server <sip:conf-server@example1.com>
```

The reverse conversion is also possible.

The To and From headers are established at initial call setup and are not changed later. Hence, any change in the mapping table does not affect existing calls.

For outbound requests on an adjacency, translation can be configured in one direction only, such as from IP address to FQDN. The domain part of the URI is converted only if it is an IP address. If it is already an FQDN, no conversion is required. Similarly, for inbound requests on an adjacency, translation can be configured in one direction only; this would typically be the other direction from inbound requests.

To prevent DNS lookups for outgoing Request-URI, IP->FQDN translations configure "force-signaling-peer", which enforces strict routing.

An exact case-insensitive match is required to translate an FQDN. For example, the mapping "aaa.com" <-> 1.1.1.1" would not match the domain "web.aaa.com", but it would match the domain "AAA.com".



If SIP to Tel URI conversion is also configured for the adjacency, this takes precedence over translation between IP address and FQDN. If the egress adjacency is configured to rewrite the To header or From header, this also takes precedence over translation between IP address and FQDN.

Configuring SIP IP-FQDN URI Translation

Cisco Unified Border Element (SP Edition) provides a configurable mapping between IP addresses and FQDNs to convert an IP address to a FQDN and a FQDN to an IP address in the Request-URI, To header, and From header. This behavior is configurable on a SIP adjacency, which can be either or both of the ingress and egress adjacency. When configured on the ingress adjacency, the translated URI is used as an input to routing, screening, data modification, CAC policy, and CDRs.

This section contains the steps to configure the SIP IP-FQDN URI Translation feature.

SUMMARY STEPS

L

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. translate {request-uri | to | from} {inbound | outbound} {ip-fqdn | fqdn-ip}
- 6. exit
- 7. sip ip-fqdn-mapping index ipv4 ip-address fqdn {both-ways | ip-to-fqdn}
- 8. exit
- 9. exit
- 10. exit
- 11. show sbc *sbc-name* sbe sip ip-fqdn-mapping

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# config terminal	
Step 2	sbc sbc-name	Enters session border controller (SBC) configuration submode.
	Example: Router(config)# sbc test	
Step 3	sbe	Enters signaling border element (SBE) configuration submode.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters adjacency SIP configuration submode.
	<pre>Example: Router(config-sbc-sbe)# adjacency sip adj1</pre>	

	Command or Action	Purpose
Step 5	translate {request-uri to from} {inbound outbound} {ip-fqdn fqdn-ip}	Configures IP-to-FQDN or FQDN-to-IP translation on SBE.
	Example: Router(config-sbc-sbe-adj-sip)#translate request-uri inbound ip-fqdn	
Step 6	exit	
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 7	<pre>sip ip-fqdn-mapping index ipv4 ip-address fqdn {both-ways ip-to-fqdn}</pre>	Configures SIP IP-to-FQDN mapping on SBE.
	Example: Router(config-sbc-sbe)# sip ip-fqdn-mapping 1 ipv4 11.22.33.41 example.sbc1.com both-ways	
Step 8	exit Example:	
	Router(config-sbc-sbe)# exit	
Step 9	exit	
	Example: Router(config-sbc)# exit	
Step 10	exit	Exit from the global configuration mode and returns to privileged EXEC mode.
	Example:	
	Router(config)# exit	
Step 11	show sbc <i>sbc-name</i> sbe sip ip-fqdn-mapping	Displays the currently configured IP-FQDN mappings in the IP-FQDN mapping table.
	Example: Router# show sbc test sbe sip ip-fqdn-mapping	

Configuration Example for SIP IP-FQDN URI Translation

The following example shows how to configure the SIP IP-FQDN URI Translation for Cisco Unified Border Element (SP Edition):

```
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
RouterRouter(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# translate request-uri inbound ip-fqdn
Router(config-sbc-sbe-adj-sip)# exit
Router(config-sbc-sbe)# sip ip-fqdn-mapping 1 ipv4 11.22.33.41 example.sbc1.com both-ways
```

```
Router(config-sbc-sbe)# exit
Router(config-sbc)# exit
Router(config)# exit
Router# show sbc test sbe sip ip-fqdn-mapping
IP FQDN mappings for SBC service "test"
```

Index Up? 1 Yes IP Dir FQDN 11.22.33.41 <-> example.sbc1.com

* -> = one-way, <-> = both-ways Router#

I





SIP Tel URI Support

Cisco Unified Border Element (SP Edition) supports Tel Uniform Resource Identifier (tel URI) in Session Initiation Protocol (SIP) messages, permitting SIP users to set up calls from a SIP IP-phone or SIP User Agent Application to an endpoint in the Public Switched Telephone Network (PSTN). The addition of tel URI to the SIP URI method of connection greatly increases the functionality of Cisco Unified Border Element (SP Edition). SIP can use the tel URI anywhere a URI is allowed, for example, as a Request-URI, along with SIP and SIP URIs.

Note

For Cisco IOS XE Release 2.4 and later, this feature is supported in the unified model.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for SIP Tel URI Support

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for SIP Tel URI Support, page 26-1
- Information About SIP Tel URI Support, page 26-2

Restrictions for SIP Tel URI Support

The following is a list of restrictions for SIP tel URI support:

 Cisco Unified Border Element (SP Edition) usually rewrites the domain-name part of the SIP Request-URI header to the configured signaling peer address and port for the outbound adjacency. For example,

sip:1234567@remote.com

becomes

sip:1234567@1.2.3.4:5060

- However, in the case of tel URIs, Cisco Unified Border Element (SP Edition) does not rewrite the domain name (since this is only an optional parameter, which is rarely present), but it rewrites the Carrier Identification Code (CIC) parameter and/or the destination directory number to ensure correct onwards routing.
- Cisco Unified Border Element (SP Edition) rejects tel URIs exceeding 160 bytes in length.
- Cisco Unified Border Element (SP Edition) ignores any parameters on the tel URI, except for the CIC parameter. All other parameters are treated as an opaque string and forwarded unchanged. As a result, the "phone-context" parameter of the local-scope tel URIs is not inspected, and the URI is routed purely based on the initial number.

Information About SIP Tel URI Support

Local and Global Tel URIs

A tel URI can either be global or local. Global tel URIs are globally unique. Local tel URIs are only valid within a specific local context. For this reason, all local tel URIs must contain the phone-context parameter to specify the context in which they are valid.

The following are examples of global and local tel URIs, respectively.

tel:+358-555-1234567

Note The separator characters, such as '-' are valid in tel URIs.

tel:1234567;phone-context=+358-555

This URI locates the endpoint with the directory number 1234567 in the context 358-555.



Although the combination of local tel URI and phone-context parameter forms a globally unique identifier, attaching a local tel URI's phone-context parameter to the tel URI does not necessarily produce a global tel URI. See section 5.1.5 of RFC 3966 for more information.

Tel URI Versus SIP URI

A SIP URI consists of a username and host domain name. A SIP URI uniquely identifies a SIP subscriber but does not necessarily resolve to one particular endpoint on a network. For example,

sip:john@cisco.com

It is also possible to use a directory number as a SIP username and an IP address and port in place of the host domain name. In this case, a SIP URI can uniquely identify an endpoint on a network. For example,

sip:1234567@192.167.1.1:5060

Local tel URIs may or may not contain a domain name in the phone-context parameter. For example,

tel: 1234567;phone-context=cisco.com

The Carrier Identification Code Parameter

A Carrier Identification Code (CIC) is a three- or four-digit number used to identify the carrier network in which the destination endpoint of a call is located. It is used by network devices to determine how a call request should be routed between carrier networks. The CIC is often used to specify which carrier network is the current freephone service provider for a freephone number. The current carrier for a given freephone number can be determined by looking up a freephone database.

Tel URIs can include carrier identification codes. For example,

tel: +1-800-234-5678;cic=2345

indicates that the carrier that has been assigned the CIC 2345 is currently the service provider for the freephone number, 1-800-234-5678.

When a network device receives a call request with a tel URI containing a CIC parameter, it will try to route the request according to the value of the CIC parameter. If it cannot route the request, it must decide whether to reject it or continue, ignoring the CIC parameter. If the CIC parameter matches the CIC of the carrier network in which the network device is located, it should route the request based on its local routing policy and strip out the CIC parameter before forwarding the request.

Note

Cisco Unified Border Element (SP Edition) must be explicitly configured to map a CIC value to **0000** in order to strip it out of outbound requests.





SIP Timer

The SIP Timer feature allows the user to configure a number of Session Initiation Protocol (SIP) timers that were hard-coded in the previous releases of Cisco IOS software. The ability to configure SIP timers enables users to improve the interoperability and performance of their devices and network environment.

Note

For Cisco IOS XE Release 2.4 and later, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Timer Functions

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Information About SIP Timer, page 27-1
- How to Configure SIP Timer, page 27-3

Information About SIP Timer

The SIP timer feature allows the user to configure some of the SIP timers that were hardcoded to default values in the previous releases of Cisco IOS software. In the previous releases, Cisco Unified Border Element (SP Edition) used the default SIP timer values recommended by RFC 3261. See Table 27-1.

Timer	Value	Meaning
T1	500 ms default	round-trip time (RTT) estimate
T2	4 s	The maximum retransmit interval for non-INVITE requests and INVITE responses
T4	5 s	Maximum duration a message will remain in the network
Timer A	initially T1	INVITE request retransmit interval, for UDP only
Timer B	64* T1	INVITE transaction timeout timer
Timer C	> 3 min	Proxy INVITE transaction timeout
Timer D	 > 32 s for UDP 0 s for TCP/Stream Control Transmission Protocol (SCTP) 	Wait time for response retransmits
Timer E	initially T1	non-INVITE request retransmit interval, UDP only
Timer F	64* T1	non-INVITE transaction timeout timer
Timer G	initially T1	INVITE response retransmit interval
Timer H	64* T1	Wait time for ACK receipt
Timer I	T4 for UDP 0 s for TCP/SCTP	Wait time for ACK retransmits
Timer J	64* T1 for UDP 0 s for TCP/SCTP	Wait time for non-INVITE request retransmits
Timer K	T4 for UDP 0 s for TCP/SCTP	Wait time for response retransmits

Table 27-1	Default	Values o	f the	Timers

Cisco Unified Border Element (SP Edition) allows the user to modify T1, T2 and Timer D, using the **udp-first-retransmit-interval**, **udp-max-retransmit-interval**, and **udp-response-linger-period** commands. You can also use the **invite-timeout** command to choose how long SBC should wait for the remote SIP endpoint to respond to the SBC's outgoing INVITE or Timer B in an outgoing transaction.

In addition to the SIP protocol-level timers, Cisco Unified Border Element (SP Edition) also allows modification of transport-related timer commands: **tcp-connect-timeout** (how long TCP SYN will wait for the reply) and **tcp-idle-timeout** (how long TCP connection should stay active while idle). Although these timers are transport-level values, Cisco IOS XE Release 2.4 supports these timers in SIP only, but not in H.323, nor H.248.



The incorrect configuration of the SIP timer values may lead to unexpected behavior in certain cases.

How to Configure SIP Timer

This section contains the steps for configuring SIP timers.

Configuring SIP Timer

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. sip timer
- 5. tcp-connect-timeout interval
- 6. tcp-idle-timeout interval
- 7. invite-timeout interval
- 8. udp-first-retransmit-interval interval
- 9. udp-max-retransmit-interval interval
- 10. udp-response-linger-period interval
- 11. end
- **12**. **show sbc** *service-name* **sbe sip timers**

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	sip timer	Enters the mode of the SIP timer function of the SBC.
	Example: Router(config-sbc-sbe)# sip timer	

	Command or Action	Purpose	
Step 5	<pre>tcp-connect-timeout interval Example: Router(config-sbc-sbe-sip-tmr)# tcp-connect-timeout 3000</pre>	Configures the time (in milliseconds) that SBC waits for a SIP TCP connection to a remote peer to complete before failing that connection. The default timeout interval is 1000 milliseconds.	
Step 6	tcp-idle-timeout interval	Minimum time (in milliseconds) a TCP socket has not processed any traffic, before it is closed. The default is 2 minutes.	
	Example: Router(config-sbc-sbe-sip-tmr)# tcp-idle-timeout 30000	Note The value for this command might not be precise since the idle timers are checked every 12 seconds.	
Step 7	<pre>invite-timeout interval Example: Router(config-sbc-sbe-sip-tmr)# invite-timeout 60</pre>	Configures the time (in seconds) that SBC waits for a final response to an outbound SIP INVITE request. The default is 180 seconds. If no response is received during that time, an internal "408 Request Timeout" response is generated and returned to the caller.	
Step 8	udp-first-retransmit-interval interval	Configures the time (in milliseconds) that SBC waits fo a UDP response or ACK before sending the first retransmission of the relevant signal.	
	Router(config-sbc-sbe-sip-tmr)# udp-first-retransmit-interval 1000	If SBC keeps getting no responses, it doubles subsequent retransmission intervals each time until they reach the udp-max-retransmit-interval duration. SBC ceases retransmitting the request and time out the signal if 64 times this duration passes without the receipt of a response/ACK.	
		The default first UDP retransmission interval is 500 milliseconds.	
Step 9	<pre>udp-max-retransmit-interval interval Example: Router(config-sbc-sbe-sip-tmr)# udp-max-retransmit-interval 8000</pre>	Configures the maximum time interval (in milliseconds) at which SBC will retransmit (see Step 9, udp-first-retransmit-interval above). The default maximum UDP retransmission interval is 4 seconds.	
Step 10	udp-response-linger-period interval	Configures the time (in milliseconds) for which SBC will retain negative UDP responses to INVITE requests.	
	Example: Router(config-sbc-sbe-sip-tmr)# udp-response-linger-period 10000	All subsequent retransmitted responses received within this time will be answered with a negative ACK. Thereafter, any further retransmitted responses are ignored.	
		The default UDP response linger period is 32 seconds.	
	Command or Action	Purpose	
---------	--	--	
Step 11	end	Exits the sip timer mode and returns to Privileged EXEC mode.	
	Example: Router(config-sbc-sbe-sip-tmr)# end		
Step 12	show sbc service-name sbe sip timers	Shows the currently configured SIP-related timers.	
	Example: Router# show sbc mysbc sbe sip timers		





SIP Configuration Flexibility

Cisco Unified Border Element (SP Edition) offers flexibility in configuring the following features of a Session Initiation Protocol (SIP) adjacency:

- OPTIONS Support
- Rewriting from header on non-REGISTER requests
- Rewriting to header on non-REGISTER requests
- Auto-detecting NAT
- Routing on wildcard domains



For Cisco IOS XR Software Release, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Configuration Flexibility

Release	Modification
Cisco IOS XR Software Release	This feature was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 3.6S	The Via Header Passthrough feature was added.

Contents

This module contains the following sections:

- Restrictions for Implementing SIP Configuration Flexibility, page 28-2
- Information About SIP Configuration Flexibility, page 28-2

- How to Implement SIP Configuration Flexibility, page 28-4
- Via Header Passthrough, page 28-5

Restrictions for Implementing SIP Configuration Flexibility

The restrictions for implementing SIP configuration flexibility are listed per feature in this chapter.

Information About SIP Configuration Flexibility

This section contains the following subsections:

- OPTIONS Support, page 28-2
- Rewriting From Header on Non-Register Requests, page 28-2

OPTIONS Support

By default, Cisco Unified Border Element (SP Edition) blocks the OPTIONS method from passing through, but users can configure Cisco Unified Border Element (SP Edition) on a per-adjacency basis to pass or block the OPTIONS method by using whitelists and blacklists.

Restrictions for OPTIONS Support

- Cisco Unified Border Element (SP Edition) strips out SDP blocks from messages when it allows the OPTIONS method to pass through. This limits what the SIP endpoints can exchange.
- The SBC-SIG does not send the Accept and Allow headers on any methods, including OPTIONS.
- Cisco Unified Border Element (SP Edition) allows only the 100Rel and Replaces tags of the Supported header to pass through, while the other tags of this header are controlled by whitelists and blacklists.

Rewriting From Header on Non-Register Requests

With this feature, users can configure Cisco Unified Border Element (SP Edition) on a per-adjacency basis to control whether it rewrites the hostport section of the From header on Non-Register Requests to the outbound SIP adjacency address or port. If Cisco Unified Border Element (SP Edition) is configured to allow the From header to pass through without it being rewritten, then Cisco Unified Border Element (SP Edition) allows the entire header to pass through without changing it. The only exception occurs with the Tag parameter; Cisco Unified Border Element (SP Edition) assigns a different value to this parameter before passing it through.

Restrictions for Rewriting From Header on Non-REGISTER Requests

- This feature is not applicable for REGISTER requests.
- This feature may only work in a limited way with the Rewrite-Register feature.

- If the From header contains a Tel URI, then Cisco Unified Border Element (SP Edition) does not rewrite the header since it does not have a hostport.
- Depending on the number of headers, options and SIP whitelist profiles, Cisco Unified Border Element (SP Edition) limits the size of the From header that it allows to pass through to approximately 1000 bytes.

Rewriting To Header on Non-REGISTER Requests

The default behavior of Cisco Unified Border Element (SP Edition) is to rewrite the hostport section of the To header on Non-Register Requests to be the outbound SIP adjacency address and port. It also removes any associated parameters. With this feature, users can configure the SBC on a per-adjacency basis to pass the To header through unchanged.

Auto-detecting NAT

With the addition of a new configuration field to the SIP adjacency, it is now possible for users to specify if Cisco Unified Border Element (SP Edition) must auto-detect whether a NAT is in use on that adjacency. If Cisco Unified Border Element (SP Edition) is configured to auto-detect NAT, then for each request that it receives, Cisco Unified Border Element (SP Edition) determines whether a NAT is in use for that endpoint. If Cisco Unified Border Element (SP Edition) determines that NAT is in use, then Cisco Unified Border Element (SP Edition) stores the bindings for that request and uses them when sending a response. Additionally, Cisco Unified Border Element (SP Edition) stores and reuses bindings for REGISTER requests for subsequent Dialog-forming and Out-of-dialog requests.

Restrictions for Auto-detecting NAT

- Cisco Unified Border Element (SP Edition) can auto-detect NAT only by comparing the Sent-by stopper in the Via header with the remote address and port of the message.
- If the stopper contains a domain name, instead of an IP address, Cisco Unified Border Element (SP Edition) cannot auto-detect whether NAT is in use. In this case, Cisco Unified Border Element (SP Edition) assumes that NAT is in use.
- Auto-detecting NAT is applied only to Out-of-dialog requests or Dialog-forming requests.

Routing on Wildcard Domains

Cisco Unified Border Element (SP Edition) routing policy allows you to use the * character in a text domain name match string. This character can match any number of characters in the called address. For example, *domain.com can match both sip1.domain.com and sip2.domain.com.

Restrictions for Routing on Wildcard Domains

- You can only specify one wildcard character in a given match string.
- This feature applies only to text domain name match rules, and not to dialed digit match rules.

How to Implement SIP Configuration Flexibility

This section contains the steps for implementing SIP configuration flexibility.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. passthrough from header
- 6. header-name [contact [add [tls-param]] | from{passthrough} | to{passthrough}]
- 7. nat force-on
- 8. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipadj	• Use the <i>adjacency-name</i> argument to define the name of the SIP adjacency.
Step 5	passthrough from header	Configures the SIP adjacency to disable From rewrite.
	Example: Router(config-sbc-sbe-adj-sip)# passthrough from header	

	Command or Action	Purpose
Step 6	header-name [contact [add [tls-param]] from {passthrough} to {passthrough}]	Configures the SIP adjacency to disable To rewrite.
	Example: Router(config-sbc-sbe-adj-sip)# header-name to passthrough	
Step 7	<pre>nat force-on Example: Router(config-sbc-sbe-adj-sip)# nat force-on</pre>	Configures the SIP adjacency to assume that all endpoints are behind a NAT device. To configure the SIP adjacency to assume that no endpoints are behind a NAT device, use the nat force-off command. By default, the SBC autodetects whether the endpoints are behind a NAT device.
Step 8	exit	Exits the adj-sip mode and returns to the SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

Via Header Passthrough

The Via Header Passthrough feature enables the SBC to interoperate with certain devices that use the Via header to authenticate other devices, such as a PBX, that do not support SIP authentication. With the introduction of this feature, the SBC can be configured to function as a SIP proxy that is compliant with RFC 3261 and RFC 3581 in its handling of the received parameter and rport parameter, which are two parameters of the Via header.

This section contains the following topics:

- Restrictions for Via Header Passthrough, page 28-5
- Information About Via Header Passthrough, page 28-6
- How to Configure Via Header Passthrough, page 28-6
- Configuration Example: Via Header Passthrough, page 28-7

Restrictions for Via Header Passthrough

The following are the restrictions for the Via Header Passthrough feature:

- This feature does not support the Topology Hiding feature. After the Via Header Passthrough feature is configured, the data that is passed through the SBC includes information about the topology of the network between the sender and the SBC. If you want to protect the network between the sender and the SBC by using the Topology Hiding feature, do not configure the Via Header Passthrough feature.
- The existing restriction on editing Via headers by using the SIP Message Editing feature is still applicable.

Information About Via Header Passthrough

Certain devices use the Via header to authenticate other devices, such as a PBX, that do not support SIP authentication. The Via Header Passthrough feature enables the SBC to interoperate with the devices that use the Via header to authenticate other devices. In releases prior to Release 3.6.0, the SBC would remove the existing Via headers from an incoming SIP message and add its own Via header before forwarding the message. With the introduction of the Via Header Passthrough feature in Release 3.6.0, the SBC can be configured to allow the existing Via headers to pass through and add its own Via header.

When the Via Header Passthrough feature is configured, the SBC adds its own Via header at the top of the stack of Via headers before forwarding the message. If the remote IP address from which the SBC receives the SIP message differs from the IP address specified in the sent-by address of the header, the SBC sets the received parameter to the actual remote IP address before forwarding the message. At the same time, if the SBC receives a SIP message in which the latest entry in the Via header contains the rport parameter with no value set for it, the SBC sets the received parameter to the source port of the message. In this scenario, the SBC also adds the received parameter to the Via header, regardless of whether the sent-by address in the Via header matches the IP address from which the message was received.

The Via Header Passthrough feature is configured at the SIP adjacency level. To maintain the Via headers on a message routed through the SBC, the Via Header Passthrough feature must be configured on both the inbound adjacency and the outbound adjacency. If this feature is not configured on either one of these adjacencies, the Via headers are removed from the SIP messages that pass through these adjacencies. Note that the SBC adds its own Via header to the outbound SIP message, regardless of whether the Via Header Passthrough feature is configured.

How to Configure Via Header Passthrough

The following procedure shows how to configure the Via Header Passthrough feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. header-name via passthrough inbound
- 6. header-name via passthrough outbound
- 7. end
- 8. show sbc sbc-name sbe adjacencies adjacency-name detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the configuration mode of an SBC service.
		• <i>sbc-name</i> —Name of the SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the configuration mode of the signaling border element (SBE) function of the SBC.
	Example:	
	Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
		• <i>adjacency-name</i> —Name of the adjacency.
	Example: Router(config-sbc-sbe) # adjacency sip adj1	
Step 5	header-name via passthrough inbound	Specifies that the Via headers on inbound requests for this adjacency must be allowed to pass through.
	Example: Router(config-sbc-sbe-adj-sip)# header-name via passthrough inbound	
Step 6	header-name via passthrough outbound	Specifies that the Via headers on outbound requests for this adjacency must be allowed to pass through.
	Example: Router(config-sbc-sbe-adj-sip)# header-name via passthrough outbound	
Step 7	end	Exits the adjacency SIP configuration mode, and returns to the privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 8	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Shows the configuration details of the specified adjacency.
	Example: Router# show sbc mySBC sbe adjacencies adj1 detail	

Configuration Example: Via Header Passthrough

The following is a sample configuration of the Via Header Passthrough feature:

```
Router(config)# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
```

```
Router(config-sbc-sbe)# adjacency sip adj1
Router(config-sbc-sbe-adj-sip)# header-name via passthrough inbound
Router(config-sbc-sbe-adj-sip)# header-name via passthrough outbound
.
.
.
.
Router# show sbc mySBC sbe adjacencies adj1 detail
```

The following is a sample output of the show sbc mySBC sbe adjacencies adj1 detail command:

	Adjacency adj1 (SIP)	
	Status:	Detached
	Signaling address:	0.0.0.0:default
·		
•		
·		
	Contact header param	eters: Passthrough
	Inbound Via Passthro	ough: Allowed
	Outbound Via Passthr	ough: Allowed



SIP Renegotiation

The Cisco Unified Border Element (SP Edition) supports two Session Initiation Protocol (SIP) renegotiation related features:

• Delta Renegotiation

The Delta Renegotiation feature determines which SIP renegotiation mode will be used by the session border controller (SBC) when renegotiating media: Delta Renegotiation or Make-Before-Break Renegotiation.

• Support Renegotiated Call Over NAT

The Support Renegotiated Call Over NAT feature allows you to ensure that pinholes are preserved for deleted streams so that if the stream is re-enabled, Cisco Unified Border Element (SP Edition) will re-use the same pinhole.

These features significantly reduce the situations in which media ports change mid-call, which provides interoperability and Network Address Translation (NAT) traversal benefits.



For Cisco IOS XE Release 2.4, the Delta Renegotiation and Support Renegotiated Call Over NAT features are supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Renegotiation

Release	Modification
Cisco IOS XE Release 2.4	The Delta Renegotiation and Support Renegotiated Call Over NAT features were introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for Delta Renegotiation, page 29-2
- Information About Delta Renegotiation, page 29-2
- Restriction for Support Renegotiated Call Over NAT, page 29-3
- Information About Support Renegotiated Call Over NAT, page 29-3
- Configuring Support Renegotiated Call Over NAT, page 29-3
- Configuration Example—Support Renegotiated Call Over NAT, page 29-7

Restrictions for Delta Renegotiation

The restrictions for Delta Renegotiation are:

- When Delta Renegotiation mode is in use, stream statistics and Secure Device Provisioning (SDP) billing information will be output at call termination, not at Delta Renegotiation.
- When Delta Renegotiation mode is in use, the following precepts apply:
 - Renegotiation may cause a change in the Differentiated Services Code Point (DSCP) marking policy.
 - The port range depends on the initial incarnation of the stream.
- Under certain scenarios, if the Cisco Unified Border Element (SP Edition) fails over while a Delta Renegotiation is in progress, media resources (such a pinholes and bandwidth allowances) may be unnecessarily allocated.

Information About Delta Renegotiation

The Delta Renegotiation feature determines which SIP renegotiation mode will be used by the Cisco Unified Border Element (SP Edition) when renegotiating media:

• Delta Renegotiation mode

When the Cisco Unified Border Element (SP Edition) performs a Delta Renegotiation, it retains the existing media pinholes and modifies their variables. Delta Renegotiation mode is used for SIP/H.323 interworked calls and for IP Multimedia Subsystem (IMS) calls.

• Make-Before-Break Renegotiation mode

When the Cisco Unified Border Element (SP Edition) performs a Make-Before-Break Renegotiation, it creates new pinholes with the proposed media properties, then removes the pre-existing pinholes when the renegotiation completes. These new pinholes temporarily exist in parallel with the existing (old) media pinholes. When the renegotiation completes, Cisco Unified Border Element (SP Edition) deletes the old media pinholes, leaving just the new ones. (Or, if the renegotiation fails, it rolls back to the old state by deleting the new pinholes.)

Delta Renegotiation mode is the default SIP renegotiation mode for all SIP-to-SIP negotiations on the Cisco Unified Border Element (SP Edition) with the following exceptions:

• Change of address family

If the renegotiation changes the address family from IPv4 to IPv6, or vice versa, a new media address is required, and therefore Make-Before-Break Renegotiation mode will be used.

Mid-call media rerouting

If the renegotiation causes a call to switch between media bypass and non-media bypass mode, the endpoints will perceive a change in the media address, and therefore Make-Before-Break Renegotiation mode will be used.

Restriction for Support Renegotiated Call Over NAT

The restriction for the Support Renegotiated Call Over NAT feature is:

• Stream statistics and SDP billing information will be output at call termination, not at Delta Renegotiation.

Information About Support Renegotiated Call Over NAT

The Support Renegotiated Call Over NAT feature allows you to ensure that media pinholes are preserved for deleted streams so that if a stream is re-enabled, the Cisco Unified Border Element (SP Edition) will re-use the same pinhole.

This feature is used to avoid de-allocation of a video pinhole in a NAT scenario where Delta Renegotiation mode is in effect and a video transmission is paused. Although the standard SDP protocol when a video transmission is paused is to set the video stream to "a=inactive" (which indicates that SBC should keep the stream allocated), there are known devices that do not set the video stream to "a=inactive" to pause it. Instead, these devices delete the video stream by setting its port to 0. To ensure that the stream remains allocated and the pinhole is preserved even when the SBC receives a port value of 0 during a media stream renegotiation, you can enable the Support Renegotiated Call Over NAT feature.

Use the **media address preserve** command to enable the Support Renegotiated Call Over NAT feature on a per-call basis.

Configuring Support Renegotiated Call Over NAT

This section contains the steps to configure the Support Renegotiated Call Over NAT feature, which preserves media pinholes for deleted streams on a per-call basis.

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table table-name
- 7. table-type { policy-set | limit {list of limit tables}}

- 8. entry *entry-id*
- **9.** cac-scope {*list of scope options*}
- 10. [no] media address preserve
- 11. action cac complete
- 12. complete
- **13.** active-cac-policy set *policy-set-id*
- 14. show sbc service-name sbe cac-policy-set policy-set-id table table-name entry entry-id

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	<pre>first-cac-table table-name</pre>	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table 1	
Step 6	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table 1	SBE policy set.

	Command or Action	Purpose
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
		list of limit tables can be one of the following values:
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all—No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
Step 8	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Choose a scope at which CAC limits are applied within each entry in a Policy Set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	<i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope src-adjacency	• <i>account</i> —Events that are from the same account.
		• <i>adjacency</i> —Events that are from the same adjacency.
		• <i>adj-group</i> —Events that are from members of the same adjacency group.
		• <i>call</i> —Scope limits are per single call.
		• <i>category</i> —Events that have same category.
		• <i>dst-account</i> —Events that are sent to the same account.
		• <i>dst-adj-group</i> —Events that are sent to the same adjacency group.
		• <i>dst-adjacency</i> —Events that are sent to the same adjacency.
		• <i>dst-number</i> —Events that have the same destination.
		• global—Scope limits are global
		• <i>src-account</i> —Events that are from the same account.
		• <i>src-adj-group</i> —Events that are from the same adjacency group.
		• <i>src-adjacency</i> —Events that are from the same adjacency.
		• <i>src-number</i> —Events that have the same source number.
Step 10	media address preserve	Ensures that media pinholes are preserved (disabled) for deleted streams.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # media address preserve</pre>	• [no] —Allows media pinholes to be deleted for deleted streams.
Step 11	action cac-complete	When an event matches, this CAC policy is complete.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac complete</pre>	
Step 12	complete	Completes the CAC policy set when you have committed the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	

	Command or Action	Purpose
Step 13	active cac-policy-set policy-set-id	Sets the active CAC policy set within an SBE entity.
	Example: Router(config-sbc-sbe)# active cac-policy-set 1	
Step 14	<pre>show sbc service-name sbe cac-policy-set policy-set-id table table-name entry entry-id</pre>	Lists detailed information for a given entry in a CAC policy table, including whether the Support Renegotiated Call Over NAT feature is enabled. When this feature is enabled, the "Media Address" field shows a value of "Preserve"
	Example:	the media Address field shows a value of freserve.
	Router# show sbc mysbc sbe cac-policy-set 1 table 1 entry 1	

Configuration Example—Support Renegotiated Call Over NAT

The following example enables the Support Renegotiated Call Over NAT feature described in this chapter on a per-call basis

```
sbc mysbc
sbe
cac-policy-set 1
first-cac-table 1
cac-table 1
table-type policy-set
entry 1
media address preserve
action cac-complete
complete
active-cac-policy-set 1
```

The following example shows detailed output for the CAC policy set 1, table 1, entry 1, including the "Media Address" field that shows a value of "Preserve," indicating the Support Renegotiated Call Over NAT feature is enabled

```
Router# show sbc mysbc sbe cac-policy-set 1 table 1 entry 1
SBC Service "mysbc"
Policy set 1 table 1 entry 1
 Match value
 Action
                           Next table
 Next-table
  Max calls
                           Unlimited
  Max call rate
                           Unlimited
 Max in-call rate
                          Unlimited
 Max out-call rate
                          Unlimited
  Max registrations
                          Unlimited
                          Unlimited
  Max reg. rate
  Max bandwidth
                           Unlimited
  Max channels
                          Unlimited
                           Allowed
  Transcoder
  Caller privacy setting
                          Never hide
  Callee privacy setting
                           Never hide
  Early media
                           Allowed
  Early media direction
                           Both
  Early media timeout
                           None
  Restrict codecs to list default
  Restrict caller codecs to list default
  Restrict callee codecs to list default
  Media bypass
                           Allowed
```

SRTP TransportTrusted-Only (by default)Callee hold settingStandardCaller hold settingStandardMedia AddressPreserveRenegotiateDeltaNumber of calls rejected by this entry0



100rel Interworking Support

Cisco Unified Border Element (SP Edition) provides support for 100rel (SIP Provisional Message Reliability) interworking. This feature provides support to resolve the interoperability problem of inconsistent support for SIP reliable provisional responses encountered when SBC works with different SIP networks.

SIP defines two types of responses: provisional and final. Final responses (2xx-6xx) convey the result of the request processing and are sent reliably. SIP provisional responses (1xx) do not have an acknowledgement system so they are not reliable. There are certain scenarios in which the provisional SIP responses (1xx) must be delivered reliably. For example in a SIP/PSTN interworking scenario it is crucial that the 180 and 183 messages are not dropped. The use of the Provisional Response ACKnowledgment (PRACK) method enables reliability to be offered to SIP provisional responses.

The 100rel option is used to indicate that the reliable provisional responses are supported or required, and the PRACK message is used to acknowledge receipt of a reliable provisional response.



This feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for 100rel Interworking Support

Release	Modification
Cisco IOS XE Release 2.5	This feature was introduced on the Cisco IOS XR.

Contents

This module contains the following sections:

- Restrictions for 100rel Interworking Support, page 30-2
- Information About 100rel Interworking Support, page 30-2
- Configuring 100rel Interworking Support, page 30-5

Restrictions for 100rel Interworking Support

The following restrictions apply when you configure the 100rel interworking support on the Cisco Unified Border Element (SP Edition):

- If late to early media interworking is required, the callee must support reliable provisional responses, and the scenario shown in Figure 30-1 must not be configured.
- The 100rel interworking allows only one offer exchange on PRACK messages for each INVITE transaction.
- The 100rel interworking is configured on the adjacency facing the network that requires 100rel support:
 - The Cisco Unified Border Element (SP Edition) configuration must be set up on the inbound adjacency of the applicable call to act as a PRACK User Agent Server (UAS) during 100rel interworking.
 - The Cisco Unified Border Element (SP Edition) configuration must be set up on the outbound adjacency of the applicable call to act as a PRACK User Agent Client (UAC) during 100rel interworking.
- The SIP uses provisional responses to avoid transaction time-out while the final response is outstanding, and reduces the frequency of these progress responses when they are sent reliably. This allows a B2BUA that receives unreliable progress responses and sends reliable progress responses to send progress responses less frequently than it receives them. Cisco Unified Border Element (SP Edition) does not attempt to do this, it simply forwards provisional responses when they are received (subject to any configured header filtering rules).

Information About 100rel Interworking Support

The 100rel interoperability feature performs the following functions on individual SIP adjacencies:

- Strips the 100rel option from incoming SIP requests.
- Sends reliable provisional responses to the caller UAC even when the responses from the called UAS are unreliable.
- Receives reliable provisional responses from the called UAS even if the caller UAC does not support them.
- Adds support for the 100rel option to outgoing SIP requests.

Figure 30-1 shows SBC acting as UAS, and Figure 30-2 shows SBC acting as UAC.



Figure 30-1 SBC Acting as UAS



Figure 30-2 SBC Acting as UAC

Caller UAC Requires 100rel

Cisco Unified Border Element (SP Edition) sets up calls from a network that requires 100rel, but needs to be routed to networks that do not support 100rel. To facilitate this, SBC strips the 100rel option from Supported and Require headers in SIP requests. After stripping the 100rel option, SBC still sends reliable provisional responses with a "Require: 100rel" header if required. Cisco Unified Border Element can also be configured to send reliable provisional responses to requests that include a "Supported: 100rel" header when such requests do not include a "Require: 100rel" header and responses are received as unreliable provisional responses.

Send Reliable Responses if Required

If a SIP request includes a "Require: 100rel" header and SBC strips the 100rel option then it must send provisional responses as reliable provisional responses with a "Require: 100rel" header. In this case the called UAS sends unreliable provisional responses because SBC has stripped the 100rel option from the request.

Send Reliable Responses if Supported

If a SIP request includes a "Supported: 100rel" header then SBC must send reliable provisional responses to the caller UAC even when the SIP request does not include a "Require: 100rel" header and the called UAS sends unreliable provisional responses.

Callee UAS Requires 100rel

Cisco Unified Border Element (SP Edition) sends requests to networks that require 100rel from networks that do not. To facilitate this, the following functions are required:

Advertise Support for 100rel

SIP Requests passing through SBC should have "Supported: 100rel" header added to them.

Remove 100rel from Responses

If SBC advertises support for 100rel, then it also ensures that the non-PRACK network receives non-100rel messages.

Configuring 100rel Interworking Support

Cisco Unified Border Element (SP Edition) requires following configurations to enable 100rel interworking support:

- The configuration is applied to SIP adjacencies.
- At the incoming side, two flags are configured to indicate,
 - whether to strip 100rel option from Supported and Require headers in the incoming SIP INVITE request.
 - whether to enable 100rel interworking if incoming SIP INVITE request contains "Supported:100rel" header.
- At the outgoing side, two flags are configured to:
 - add "Supported:100rel" in the outgoing SIP INVITE request.
 - add "Require:100rel" in the outgoing SIP INVITE request.

This section contains the steps to configure the 100rel interworking support.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip *adjacency-name*
- 5. 100rel inbound {strip | support}
- 6. 100rel outbound {require-add | support-add}
- 7. end
- 8. show sbc sbc-name sbe adjacencies {adjacency-name} [detail]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# config terminal	
Step 2	sbc sbc-name	Enters session border controller (SBC) configuration submode.
	Example:	
Sten 3	Router(config) # sbc test	Enters signaling horder element (SBE) configuration submode
oreh o		Eners signating border clement (SDE) configuration submode.
	Example:	
	Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters adjacency SIP configuration submode.
	Example:	
	Router(config-sbc-sbe)# adjacency sip	
	adj1	
Step 5	100rel inbound {strip support}	Configures the 100rel interworking parameters for inbound SIP
	Example	adjacencies on signaming border elements (3DEs).
	Router(config-sbc-sbe-adj-sip)# 100rel	
	inbound strip Router(config-shc-she-adi-sin)# 100rel	
	inbound support	
Step 6	100rel outbound {require-add support-add}	Configures the 100rel interworking parameters for outbound SIP
		adjacencies on signaling border elements (SBEs).
	Example:	
	Router(config-sbc-sbe-adj-sip)# 100rel	
	Router(config-sbc-sbe-adj-sip)# 100rel	
017	outbound support-add	
Step /	end	Returns to the privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe)# end	
Sten 8	Router(config-sbc)#	Lists the adjacencies configured on signaling horder elements
orch o	{adjacency-name} [detail]	(SBEs).
	Example:	
	adj1 detail	



Customized System Error Messages

Cisco Unified Border Element (SP Edition) supports customized system error messages.

Feature History for Implementing SNMP

Release	Modification
Cisco IOS XE Release 3.1S	Customized System Error Messages feature was introduced.

Contents

- Information About Customized System Error Messages, page 31-1
- Configuring Customized System Error Messages, page 31-4
- Configuration Example of Implementing Customized System Error Message, page 31-6

Information About Customized System Error Messages

SBC provides the ability to map internal system error-codes to SIP status-codes, and gives system administrators the ability to add a customer configured Reason: header into the response.

System administrators can map and customize error messages in user-defined error profiles. The following types of existing SIP error codes can be mapped and customized in user-defined error profiles:

- Call Admission Control (CAC)
- Number Analysis (NA)
- Routing Errors (RTG)

A default error profile is automatically created and attached to SIP adjacencies during SBE configuration. The default error profile can be modified, but cannot be deleted.

User-defined error-profiles are added to the existing SIP profiles and can be attached to adjacencies.

Errors are identified by a cause and an optional a sub-cause. If no sub-cause is entered, all possible sub-causes are mapped to that cause.

Each cause/sub-cause combination can be mapped by the user to any SIP status-code in the range between 400 and 699.

When an internal error is generated, the system checks for a configured cause/sub-cause mapped to that error. The system first checks the adjacency for specific error-profile, then it checks the default profile for an equivalent error mapping. If no match is found, the existing internal error message is returned. If a configured error profile is found, it overwrites the internal error message.

A user-defined error-profile contains the following elements:

- Cause
- Sub-cause
- Status-code
- Reason

Cause

In an error profile, the cause of an internal error is specified, using the **cause** command to select one of the following available CLI causes:



To see a list of the available causes, use the question mark (?) online help function after you have selected the cause.

- cac-in-call-msg-rate—cac: The rate of mid-call messages has exceeded a maximum configured limit
- cac-max-bandwidth—cac: The bandwidth used has exceeded a maximum configured limit
- cac-max-call-rate—cac: Call setup rate exceeded a maximum configured limit
- cac-max-channels—cac: The number of media channels used has exceeded a maximum limit
- cac-max-num-calls—cac: The number of calls has exceeded a maximum limit
- **cac-max-reg**—cac: The number of registrations has exceeded a maximum configured limit
- cac-max-reg-rate—cac: The rate of registrations has exceeded a maximum configured limit
- cac-max-updates—cac: The number of call updates has exceeded the configured limit
- cac-out-call-msg-rate—cac: The rate of out of dialogue messages has exceeded a maximum configured limit
- cac-rtp-disallowed—cac: Disallowing rtp caused the call to fail
- cac-srtp-disallowed—cac: Disallowing srtp caused the call to fail
- cac-srtp-rtp-interwork—cac: call failed due to srtp to rtp interworking disallowed
- enum-failure—ENUM processing encountered an error
- max-media-streams—An offer cannot be reduced to meet the maximum number of media streams
- mg-srtp-unsupported—No MG was found which can support srtp
- na-invalid-address—na: Number validation failure
- no-acceptable-codec—No acceptable codec can be found for an offer
- rtg-max-routes-tried—rtg: The maximum number of routing attempts exceeded
- rtg-no-route-found—rtg: Routing failed to find a route
- rtg-route-unavailable—rtg: The route selected by call-policy is unavailable
- srtp-general-error—srtp general error
- **sub-media-bearer-chan-fail**—subscriber media bearer channel has failed mid-call

- **sub-media-bearer-chan-rej**—subscriber media bearer channel has rejected during setup or renegotiation
- sub-sig-bearer-chan-fail—subscriber signaling bearer channel is unavailable

Sub-cause

After the cause is selected, the sub-cause can then be selected optionally.

To see a list of the available sub-causes for each cause, use the question mark (?) online help function after you have selected the cause. The following list shows all available sub-causes:

- na-dst-number—Destination number based analysis
- na-src-adjacency—Source adjacency based analysis
- na-src-account—Source account based analysis
- **na-sub-category**—Subscriber category based analysis
- na-carrier-id—Carrier identification code based analysis
- na-src-number—Source number based analysis
- na-no-src-number—No source number present for source number based analysis
- rtg-src-address—Source address based routing
- rtg-dst-address—Destination address based routing
- rtg-src-adjacency—Source adjacency based routing
- rtg-src-account—Source account based routing
- rtg-category—Category based routing
- rtg-sub-category—Subscriber category based routing
- rtg-src-domain—Source domain based routing
- rtg-dst-domain—Destination domain based routing
- rtg-time—Time based routing
- rtg-dst-tgid—Destination trunk group Identifier based routing
- rtg-src-tgid—Source trunk group identifier based routing
- rtg-carrier-id—Carrier identification code based routing
- rtg-round-robin—Round robin based routing
- rtg-least-cost—Least cost based routing
- cac-unknown—Unknown call admission control error
- cac-per-call-scope—Call admission control call scope error
- cac-src-number-scope—Call admission control source number scope error
- cac-downstream-scope—Call admission control downstream scope attribute error
- cac-upstream-scope—Call admission control upstream scope attribute error
- **sub-rx-reg-bearer-loss**—Failed to route to a subscriber because the Rx session for the subscriber registration suffered loss of bearer
- **sub-rx-reg-bearer-rel**—Failed to route to a subscriber because the rx session for the subscriber registration suffered release of bearer
- **sub-rx-reg-bearer-term**—Failed to route to a subscriber because the rx session for the subscriber registration was terminated

- **sub-rx-media-policy-rej**—Rx session for a call was rejected for policy reasons (for example, unsupported media)
- **sub-rx-media-error**—Rx session for a call was rejected for non-policy reasons (for example, service unavailable)
- sub-rx-reg-bearer-loss—Rx session for a call suffered loss of bearer
- sub-rx-reg-bearer-rel—Rx session for a call suffered release of bearer
- sub-rx-reg-bearer-term—Rx session for a call was terminated
- enum-resource—enum encountered a resource shortage
- enum-dst-not-number—enum destination address which was not a telephone number
- enum-unknown-number—enum unable to resolve a telephone number
- enum-interface-failure—enum failed in the enum interface
- enum-regex-error—enum failed because a regex in a NAPTR record was invalid

Status-code

The SIP status-code numbers range from 400 to 699. A SIP status code can be mapped to a selected cause/sub-cause, using the **cause** command.

Reason

The reason field allows system administrators to optionally configure a SIP "Reason:" header, which is inserted into the error response and displayed when an error occurs. The configured reason header must conform to the syntax rules defined in RFC 3326.

Configuring Customized System Error Messages

Use the following procedure to configure custom error messages.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. sip error-profile error-profile-name
- 5. description description
- 6. cause cause [sub-cause sub-cause] status-code status-code [reason reason]
- 7. exit
- 8. adjacency sip adjacency-name
- 9. error-profile outbound profile-name
- 10. end
- 11. show sbc *sbc-name* sbe sip error-profile

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>sip error-profile error-profile-name</pre>	Creates an error profile and enters error profile configuration mode.
	Example: Router(config-sbc-sbe)# sip error-profile IN_Err_Profile_1	
Step 5	description description	Adds a description to an error profile.
	Example: Router(config-sbc-sbe-sip-err)# description call rate error	
Step 6	<pre>cause cause [sub-cause sub-cause] status-code status-code [reason reason]</pre>	Configures the cause, sub-cause, status-code, and reason of an internal error for an error profile.
	<pre>Example: Router(config-sbc-sbe-sip-err)# cause cac-max-reg status-code 553 reason "SIP ;cause=503 ;text=\" Exceed the max reg num\""</pre>	
Step 7	exit	Exits to the previous mode.
	Example: Router(config-sbc-sbe-err)# exit	
Step 8	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip Adj_1	
Step 9	error-profile outbound profile-name	Configures an existing error profile as the outbound SIP error profile.
	Example: Router(config-sbc-sbe-adj-sip)# error-profile outbound OUT_Err_Profile_1	

	Command or Action	Purpose
Step 10	end	Exits configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-enum-entry)# end</pre>	
Step 11	show sbc <i>sbc-name</i> sbe sip error-profile	Displays the configuration information for all error profiles.
	Example:	
	Router# show sbc SBC1 sbe sip error-profile	

Configuration Example of Implementing Customized System Error Message

The following example shows how to configure custom error messages:

Router# configure terminal
Router(config)# sbc SBC1
Router(config-sbc)# sbe
Router(config-sbc-sbe)# sip error-profile Error_Profile_1
Router(config-sbc-sbe-sip-err)# description call rate error
Router(config-sbc-sbe-sip-err)# cause cac-max-reg status-code 553 reason "SIP ;cause=503
;text=\" Exceed the max reg num\""
Router(config-sbc-sbe-err)# adjacency sip Adj_1
Router(config-sbc-sbe-err)# error-profile outbound OUT_Err_Profile_1
Router(config-sbc-sbe-enum-entry)# end
Router# show sbc SBC1 sbe sip error-profile



BFCP Support

Binary Floor Control Protocol (BFCP), defined in RFC 4582, is a protocol for controlling the access to the media resources in a conference.

Cisco Unified Border Element (SP Edition) was earlier known as Integrated Session Border Controller. It is referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all the Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for BFCP Support

Release	Modification
Cisco IOS XE Release	This feature was introduced on the Cisco ASR 1000 Series Routers.
3.3S	

Contents

This chapter contains the following sections:

- Prerequisite for BFCP Support, page 32-1
- Restrictions for BFCP Support, page 32-2
- Information About BFCP Support, page 32-2
- Configuring BFCP Support, page 32-3
- Configuration Example of BFCP Support, page 32-9

Prerequisite for BFCP Support

Following is the prerequisite pertaining to the BFCP Support feature:

• The SBC must pass through the *b*=*CT* line and the *a*=*rtcp-fb*:**nack pli* RTCP feedback information included in the Session Description Protocol (SDP).

Restrictions for BFCP Support

Following are the restrictions pertaining to the BFCP Support feature:

- The SBC treats a generic media stream the same way it treats other media streams. Therefore, a call is released only if all the media streams are reported as being inactive. The Media Packet Forwarder (MPF) media timer is processed in the same way as the other voice or video streams pertaining to the BFCP stream.
- A BFCP media stream and a generic media stream do not have a bandwidth specified. Therefore, it can be policed only by the MPF, and not the Call Admission Control (CAC) total bandwidth limits.
- The SBC does not support the generic TCP streams or BFCP over TCP. Therefore, a request to add a TCP stream to the generic media stream configuration gets rejected.
- H.323 calls or H.323-SIP interworking calls are not supported.

Information About BFCP Support

The BFCP Support feature supports BFCP over UDP in the SBC by configuring BFCP as a recognized generic media stream that can be forwarded using the best-effort traffic class.

Generic media streams are media streams in which the media (m)-line definition uses * instead of a codec list, for example, m=application *port* UDP/BFCP *. By default, the SBC cuts these m-lines out of the SDP offers and replies by setting the port to zero. These media lines carry no bandwidth information and therefore, cannot be policed against CAC limits, denial of service, or media theft attacks of the SBC.

The BFCP Support feature introduces the best-effort traffic class that allows policing of these media lines in the media forwarder.

The SBC can be configured to accept specific generic media streams. After this, the accepted generic media streams are added to the best-effort traffic class. MPF implementation supports the best-effort traffic class by policing the actual usage of the aggregate of these streams.

Best-Effort Traffic Class

Prior to Cisco IOS XE Release 3.3S, the media streams had their bandwidth specified for audio and video streams, or were not subjected to any policing, such as T120. From Cisco IOS XE Release 3.3S, the SBC is configured to accept arbitrary type and number of generic media streams. Some of the BFCP streams can now have low bandwidth protocol messages. The best-effort traffic class simplifies packet policing because it allows a media forwarder to handle such streams cumulatively. The best-effort traffic class rate limit is 1Mbps cumulatively.

Deploying BFCP Support

Figure 32-1 shows a scenario where the SBC can be deployed for the BFCP Support feature. In this scenario, the SBC is located in the Service Provider network, allows inter enterprise calls between different VPNs, and protects the core network.





Configuring BFCP Support

This section describes how to configure the BFCP Support feature on the SBC.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name

- 3. sbe
- 4. stream-list stream-list-name
- **5. description** *description*
- 6. generic-stream media-type {application | message} transport udp protocol protocol-name
- 7. exit
- 8. cac-policy-set policy-set-id
- 9. cac-table table-name
- **10.** table-type {policy-set | limit {*list of limit tables*}}
- 11. entry entry-id
- **12**. **action** [next-table *goto-table-name* | cac-complete]
- 13. generic-stream caller generic-stream-list
- 14. generic-stream callee generic-stream-list
- **15.** match-value key
- 16. exit
- 17. exit
- 18. complete
- 19. end
- 20. show sbc service-name sbe stream-list
- 21. show sbc service-name sbe cac-policy-set id table name entry entry

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the SBE entity mode within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>stream-list stream-list-name</pre>	Configures a stream list and enters the stream list configuration mode.
	Example: Router(config-sbc-sbe)# stream-list my_stream	• <i>stream-list-name</i> —The name of the stream list. The stream list name can be upto 30 characters.

	Command or Action	Purpose
Step 5	description description	Configures descriptive text for the stream list.
	Example: Router(config-sbc-sbe-stream-list)# description "This is my first stream list"	
Step 6	<pre>generic-stream media-type {application message} transport udp protocol protocol-name Example: Router(config-sbc-sbe-stream-list)# generic-stream media-type application transport udp protocol BFCP</pre>	 Configures the media type for a generic stream. application—Specifies application as media type for the generic stream. message—Specifies message as media type for the generic stream. transport—Configures the transport protocol for the generic stream. udp—Specifies the UDP protocol for the generic stream. protocol—Specifies the protocol name for the generic stream. protocol-name—The protocol name for the generic stream. The protocol name is case sensitive.
Step 7	exit Example:	Exits from the stream list configuration mode and enters the SBE configuration mode.
Step 8	Router(config-sbc-sbe-stream-list)# exit cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 2	• <i>policy-set-id</i> —The call policy set number that can range from 1 to 2147483647.
Step 9	<pre>cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# cac-table 2</pre>	Enters the admission control table configuration mode (creating one, if necessary) within the context of an SBE policy set.
Step 10	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures a limit to the table types that are to be matched by the match-value command. For the multiple SBC media bypass feature, use the following table type:
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type src-adjacency	• <i>src-adjacency</i> —Compares the name of the source adjacency.
Step 11	entry entry-id	Enters the CAC table entry mode to create or modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 12	action [next-table goto-table-name cac-complete]	Configures the action to be performed after the entry, in an admission control table. Possible actions are:
	Example:	• Identifies the next CAC table to be processed using the next-table keyword and the <i>goto-table-name</i> argument.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• Stops the processing for the scope using the cac-complete keyword.
Step 13	generic-stream caller generic-stream-list	Configures the generic media stream list settings for a caller.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # generic-stream caller my-stream</pre>	• generic-stream-list —The name of the generic stream list. This generic stream list should be defined during the configuration of the stream list.
Step 14	generic-stream callee generic-stream-list	Configures the generic media stream list settings for a callee.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # generic-stream callee my-stream</pre>	• generic-stream-list —The name of the generic stream list. This generic stream list should be defined during the configuration of the stream list.
Step 15	match-value key	Configures the match value of an entry in a CAC limit table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value SIP-adj-test</pre>	• <i>key</i> —The keyword used to match events. The format of the key is determined by the table type limit.
Step 16	exit	Exits from the CAC table entry configuration mode and enters the CAC table mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 17	exit	Exits from the CAC table configuration mode and enters the CAC policy set configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit</pre>	
Step 18	complete	Completes the CAC policy set after you have committed the complete set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 19	end	Exits from the CAC policy set configuration mode and enters the Privileged EXEC mode.
	Example: Router(config)# end	
	Command or Action	Purpose
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Step 20	show sbc sbc-name sbe stream-list	Displays the stream lists that are present on the SBE.
	Example: Router# show sbc mysbc sbe stream-list my-stream	
Step 21	<pre>show sbc service-name sbe cac-policy-set id table name entry entry</pre>	Displays detailed information about a specific entry in a CAC policy table.
	Example: Router# show sbc mysbc sbe cac-policy-set 1 table MyTable entry 1	

The following example shows the output of the show sbc sbe stream-list command:

```
Router# show sbc Mysbc sbe stream-list
SBC Service "sbc"
  Stream list: my-stream
    Description This is my first stream list
   Media-type application Transport udp protocol Streambased
   Media-type message Transport udp protocol BFCP
The following example shows the output of the show sbc sbe cac-policy-set table entry
command:
Router# show sbc Mysbc sbe cac-policy 25 table 2 entry 1
SBC Service "Mysbc"
CAC Averaging period 1: 60 sec
CAC Averaging period 2: 0 sec
CAC Policy Set 25
  Global policy set: No
  Description:
  First CAC table:
  First CAC scope: global
  Table name: 2
   Description:
   Table type: limit src-adjacency
   Total call setup failures (due to non-media limits): 0
   Entry 1
   Match value:
   Match prefix length: 0
   Action: CAC complete
   Number of call setup failures (due to non-media limits): 0
   No. of registrations rejected (due to registration limits): 0
   Max calls per scope:
                                                         Unlimited
   No. of events rejected due to Max Call Limit:
                                                         0
   Max reg. per scope:
                                                         Unlimited
   No. of events rejected due to Max Reg limit:
                                                         0
                                                         Unlimited
   Max channels per scope:
   Max updates per scope:
                                                         Unlimited
   Max bandwidth per scope:
                                                         Unlimited
```

Averaging-period 1 Averaging-period Max call rate per scope: Unlimited Unlimited No. of events rejected due to Max call rate: 0 0 Max reg. rate per scope: Unlimited Unlimited 0 No. of events rejected due to Max reg rate: 0 Max in-call message rate: Unlimited Unlimited No. of events rejected due to Max in-call rate: 0 0 Max out-call message rate: Unlimited Unlimited No. of events rejected due to Max Out call rate: 0 Timestamp when the rejection counts were last reset: 2011/01/03 22:29:40 Early media: Allowed Early media direction: Both Early media timeout: None Transcoder per scope: Allowed Callee Bandwidth-Field: None Caller Bandwidth-Field: None Media bypass: Allowed Asymmetric Payload Type: Not Set Renegotiate Strategy: Delta Trusted-Only (by default) SRTP Transport: Caller hold setting: Standard Callee hold setting: Standard Caller limited-privacy-service: Never hide identity Callee limited-privacy-service: Never hide identity Caller privacy-service: Not set Callee privacy-service: Not set Caller edit-privacy-request: Not set Callee edit-privacy-request: Not set Caller edit-privacy-request sip strip: Not set Callee edit-privacy-request sip strip: Not set Caller edit-privacy-request sip insert: Not set Callee edit-privacy-request sip insert: Not set Caller voice QoS profile: Default Callee voice OoS profile: Default Caller video QoS profile: Default Callee video QoS profile: Default Caller sig QoS profile: Default Callee sig QoS profile: Default Caller inbound SDP policy: None Callee inbound SDP policy: None Caller outbound SDP policy: None Callee outbound SDP policy: None SDP Media Profile : None Caller Generic Stream my-stream : Callee Generic Stream : mv-stream Caller media disabled: None Callee media disabled: None Caller unsignaled secure media: Not Allowed Callee unsignaled secure media: Not Allowed Caller response downgrade support: No Callee response downgrade support: No Caller retry rtp support: No Callee retry rtp support: No Resend sdp answer in 200ok: No Caller tel-event payload type: Default Callee tel-event payload type: Default Media flag: None Restrict codecs to list: Default. Restrict caller codecs to list: Default Restrict callee codecs to list: Default Codec preference list: Default Caller Codec profile: None Callee Codec profile: None Caller media caps list: None Callee media caps list: None TCS extra codec list: None

Caller media-type:	Inherit (default)
Callee media-type:	Inherit (default)
Caller Media Bypass:	Inherit (default)
Callee Media Bypass:	Inherit (default)
Media Bypass Type:	Not set
Callee local transfer support:	Inherit (default)
Maximum Call Duration:	Unlimited
Caller SRTP support:	Inherit (default)
Callee SRTP support:	Inherit (default)
SRTP Interworking:	Inherit (default)
SRTP media Interworking:	Inherit (default)
Ims rx preliminary-aar:	Disabled(default)
Ims media-service:	None(default)
media bandwidth policing:	Inherit(default)
Billing filter:	Inherit(default)
Caller ptime:	None (default)
Callee ptime:	None (default)
Caller codec variant conversion:	Disabled (default)
Callee codec variant conversion:	Disabled (default)
Caller inband DTMF mode:	Inherit(default)
Callee inband DTMF mode:	Inherit(default)
Caller Port Range Tag:	Inherit (default)
Callee Port Range Tag:	Inherit (default)
Session refresh renegotiation: In	herit(default)

Configuration Example of BFCP Support

Following is a configuration example of the BFCP Support feature on the SBC:

```
sbc sbc
     sbe
        stream-list my-stream
        description voip stream list
        generic-stream media-type application transport udp protocol BFCP
        generic-stream media-type application transport udp protocol test
        exit
    cac-policy-set 2
       cac-table 2
         table-type limit src-adjacency
          entry 1
          action cac-complete
          generic-stream caller my-stream
          generic-stream callee my-stream
          match-value SIP-adj-test
        exit
       exit
       complete
```





H.323 Support

Cisco Unified Border Element (SP Edition) supports H.323, as well as Session Initiation Protocol (SIP). This H.323 interworking capability enables multimedia products and applications from multiple vendors to interoperate and allows users to communicate without concern for compatibility.

H.323 is the international standard for multimedia communication over packet-switched networks, including local area networks (LANs), wide area networks (WANs), and the Internet. It was first defined by the International Communications Union (ITU) in 1996. The most recent version is H.323 version 6 (2006).

Note

This feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for H.323 Support

Release	Modification
Cisco IOS XE Release 2.5	H.323 feature support was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.2S	Added the restrictions for the support of SIP secure calls over an H.323 interface.
Cisco IOS XE Release 3.3S	The Limited H.323 ID Routing and Passthrough Support feature was added.

Contents

This module contains the following sections:

- Prerequisites for H.323 Support, page 33-2
- Restrictions for H.323 Support, page 33-2
- Information About H.323 Support, page 33-2
- H.323 Features, page 33-3
- Configuring H.323 Features, page 33-20
- Configuring Separate H.245 Control Channel and RAS Tech Prefix: Example
- Configuring User Protocol Timer Controls: Example

Prerequisites for H.323 Support

This feature requires basic understanding of H.323-related ITU standards, gatekeepers, and gateways. Gateways are responsible for edge routing decisions between the Public Switched Telephone Network (PSTN) and the H.323 network. Gatekeepers are used to group gateways into logical zones and perform call routing between them.

Restrictions for H.323 Support

The restrictions for H.323 support are listed per feature in this chapter and other H.323-related chapters in the guide.

Information About H.323 Support

H.323 is a suite of protocols and documents that includes the ITU-T standards H.323, H.225.0, H.245, the H.450-series, and the H.460-series. Not all components of H.323 are mandatory as part of a standard H.323 system. For example, H.460.2, which describes number portability, is generally not used in enterprise video conferencing systems. Also H.323 utilizes both ITU-defined codecs and codecs defined outside the ITU to transmit audio, video, and text.

H.323 is used in Voice over Internet Protocol (VoIP), and IP-based video conferencing and serves a similar purpose to that of the Session Initiation Protocol (SIP). It was designed from the outset to operate over IP networks primarily, although H.323 may also operate over other packet-switched networks. H.323 was designed with multipoint voice and video conferencing capabilities, although most users do not take advantage of the multipoint capabilities specified in the protocol.

H.323 is more mature than SIP, but lacks the flexibility of SIP. SIP is currently less defined, but has greater scalability which could ease the Internet application integration. Like SIP, H.323 is one of the world market leaders for transporting voice and video over networks around the world, with billions of minutes of voice traffic every month. The SBC supports both SIP and H.323, enabling multimedia products and applications from multiple vendors to interoperate, and allowing users to communicate without concern for compatibility.

The following supported H.323 features are documented in another chapter in this configuration guide or represent part of standard Q.931/H.225 protocols that are documented in this chapter:

- H.323 to SIP Interworking—Interworking of a defined subset of SIP/H.323 call and media signaling. See the H.323 to SIP Interworking chapter in this configuration guide.
- Basic conferencing passthrough (this feature is part of Q.931/H.225 passthrough)—Pass through of conferenceID and conferenceGoal. Conference is controlled by the third party equipment, such as call manager. The SBC enables the conference to pass through all the conference-related information.
- H.450 passthrough (this feature is part of Q.931/H.225 passthrough)—Pass through of H.450 elements between call legs.

Note

All H.323 calls, including established H.323-H.323 and SIP-H.323 interworking calls, are disconnected upon an SBC switchover. An SBC switchover occurs when an active RP switches over to the standby RP in a hardware redundant system (such as a Cisco ASR 1006 Router) or when the active IOS process switches over to the standby IOS process in a redundant software system (such as a Cisco ASR 1002 Router).

H.323 Features

The following supported H.323 features are documented in this chapter. The H.323 Call Routing features are documented in the ?\$paratext[CT_ChapTitle]>? chapter:

- H.323 Call Routing
- H.323 Video Codec Support
- H.323 Slow Start to H.323 Fast Start Interop
- Separate H.245 Control Channel
- H.245 Passthrough
- Slow Start Media Relay
- Codec Mappings
- DTMF Interworking
- Transcoding
- RAS Tech Prefix
- User Protocol Timer Control
- T.38 Fax Relay
- Q.931/H.225 Passthrough
- H.323 Privacy
- H.245 Address in Call Proceeding
- Multiple TCP for H.323
- Extending SIP Secure calls over H.323 Interface
- Limited H.323 ID Routing and Passthrough Support

H.323 Call Routing

Cisco Unified Border Element (SP Edition) supports the following H.323 call routing features:

- H.323 Hunting
- Picking a Next Hop in Routing Policy
- Support for H.323 Addressing
- DNS Name Resolution
- Number Validation and Editing
- Load Balancing
- Inter-VPN Calling



The H.323 call routing features are noted in this chapter for reference. However, they are described in the H.323 Call Routing Features in the ?\$paratext[CT_ChapTitle]>? chapter.

H.323 Video Codec Support

Cisco Unified Border Element (SP Edition) allows H.323 video calls to be established through it. The supported H.323 video codecs are H.261, H.263 and H.264.

No specific configuration is required on either the end points or the SBC to enable this feature.

H.323 Slow Start to H.323 Fast Start Interop

Cisco Unified Border Element (SP Edition) supports the ability of H.323 endpoints with different starting modes of operation to interoperate with one another. Note that only the H.323 slow start endpoint to H.323 fast start endpoint is supported, and not vice-versa.

H.323 has two modes of operation: slow start and fast start. The initiation of a call may proceed in a slow start or fast start in H.323. In a slow start, H.323 signaling consists of Setup, Call Proceeding, Alerting, and Connect steps. After these steps, the H.245 media negotiation is performed. When a call is initiated in H.323 fast start, the H.245 media negotiation is performed within the initial Setup message.

This H.323 Slow Start to H.323 Fast Start Interop feature is enabled on a per adjacency basis. You can use the **start fast** command to configure the H.323 fast start mode. All calls routed to an adjacency uses the fast start mode that is configured on that adjacency. H.323 endpoints start in the fast start mode for outgoing calls on the adjacency and be able to convert incoming calls to the mode configured for that endpoint on the adjacency. When the fast start mode is configured, the SBC only uses the fast start mode for outgoing calls. However, incoming slow start calls are converted to fast start mode as they cross the SBC.

If the fast start mode is not configured on the adjacency, by default, the outgoing call start is the same as the incoming call start. The mode of operation can be modified while the adjacency is active but the change will only affect new calls. See the ?\$paranum>Configuring H.323 Slow Start to H.323 Fast Start Interop? section on page 33-21 for configuration step information.



The fast start outgoing call is only a proposal, indicating the preferred mode from the SBC's perspective. The H.323 endpoint can accept it or fall back to normal slow start procedure according to the H.245 specification.

Separate H.245 Control Channel

The H.323 procedures require that the SBC sets up a separate H.245 control channel over TCP. This feature complements tunneled H.245 support, enabling the user to control whether to use tunneling or not.

This feature enables the SBC to carry out an H.323-H.323 call, where two call legs can negotiate different H.245 transport mechanisms. Each call leg decides independently whether to use a separate H.245 control channel.

The SBC sets up separate H.245 control channels only when required in one of the following cases:

- The SBC has received a startH245 Facility
- The SBC needs to send out an H.245 message and tunneling is not available

The SBC never requests separate H.245 control channel while tunneling is available unless the "disable tunneling" command line interface (CLI) command is set (see ?\$paranum>Configuring Separate H.245 Control Channel? section on page 33-22). The SBC does not connect to an H.245 address simply because the peer offered an h245Address.

The SBC does not offer an H.245 address until it needs to, performing the following:

- Where possible, the SBC connects to the peer instead.
- Where impossible, the SBC offers its own H.245 address in a startH245 Facility and waits for 10 seconds for the peer to connect. This timeout is not configurable.

Since H.323 v2 onwards has support for Facility reason startH245, support for this feature is assumed in all peer devices. If the peer requires an H.245 connection and one does not exist, the partner must use a startH245 Facility to induce the SBC to connect to it.

If there is no H.245 transport possible (tunneled or separate), and H.245 messages must be sent by the SBC, then the call is terminated.

On receipt of provisionalRespToH245Tunnelling, the SBC waits to determine the final tunneling outcome before attempting separate H.245. H.245 messages are queued at this point and sent as soon as an H.245 transport becomes available.

In the event of an H.245 connection race, the SBC only disconnects if it looses. The partner must disconnect if it loses. Races are resolved by comparing the listen address/port (not the connection address/port).

Back-pressure is exerted at call scope, or connection scope when multiple calls share a connection. So, if call leg B cannot forward H.245 messages for some reason, call leg A's connection may exert TCP back pressure on the peer. If call leg A is doing H.245 tunneling, and sharing a Q.931 TCP connection with other calls, then the peer will experience back pressure on the other calls too.

The SBC tears down separate H.245 connections at the same point as their call by closing the relevant socket.

Restrictions for Separate H.245 Control Channel

The restrictions for the H.245 control channel are:

- The SBC does not support a model, in which it induces the peers in an H.323-H.323 call to set up the H.245 TCP connection directly between themselves or to the data border element (DBE).
- No show command is provided to list the H.245 transport status on a per-adjacency or per-call basis.
- The H.245 security is not supported.

H.245 Passthrough

In media bypass, H.245 content is passed unmodified between two H.323 call legs (for more information about media bypass, see the ?\$paranum>How Adjacencies Affect Media Routing? section on page 6-5 in the Implementing Adjacencies on Cisco Unified Border Element (SP Edition) chapter). Passthrough happens irrespective of whether an H.245 message is received over tunneled H.245 transport or a separate H.245 control channel, and does not require that both H.323 call legs use the same H.245 transport mechanism. This feature permits inserting an SBC between two H.323 devices without any change to the passing H.245 content.

This is achieved by passing through H.245 messages opaquely between the endpoints. The Fast Start request and response is passed through in the same way as mainline H.245. The only messages inspected by the SBC are fast start and logical channel signaling. These are used to derive the bandwidth used for the call.

Restrictions for H.245 Passthrough

The restrictions for the H.245 passthrough are:

- Configuration to block passthrough of certain messages or message elements is not included in this feature and is covered separately.
- In a media bypass call, no Session Description Protocol (SDP) appears in the billing records.
- The SBC does not support rate limiting of passed-through H.245 traffic, other than generic rate limiting of all signaling traffic.

Slow Start Media Relay

The SBC supports media relay (which is media pass through the DBE) of unidirectional H.245 channels. H.245 codec types are converted to Session Description Protocol (SDP) for the purposes of the DBE programming, transcoder programming, and billing. This is done using a codec mapping table (see ?\$paranum>T.38 Fax Relay? section on page 33-11).

When dealing with codec types, for which no SDP mapping exists, the SBE makes a best-effort attempt, and tries to find the best possible SDP match. You can also use the **codec-restrict-to-list** command to configure a Call Admission Control (CAC) policy to restrict the codecs used in signaling a call to the set of codecs given in the named list. This configured CAC policy will have the effect of blocking setup of a particular codec or of ignoring an unknown codec.

Inserting an SBC between two H.323 devices does not impact H.245 function (see ?**\$paranum>H.245** Passthrough? section on page 33-6). For example, the SBC does not modify the logical channel numbers of H.245 channels in a media relay call. In a distributed DBE model, H.248 signals are used to establish the necessary media terminations on the DBE. The SBC supports renegotiation of media, using H.245 procedures, such as:

- Fax upspeed: Where endpoints switch over from a low-bit-rate audio codec to ITU-T G.711
- TCS=0: Where one endpoint induces the other to temporarily close all of its channels

Switchover to T.38 fax is described below in ?\$paranum>T.38 Fax Relay? section on page 33-11.

Restrictions for Slow Start Media Relay

The restrictions for slow start media relay are:

- The SBC does not support bidirectional H.245 channels in fast start or Open Logical Channel (OLCs).
- Dual tone multifrequency (DTMF) interworking is not supported between different types of UserInputIndication.
- For SIP-SIP and H.323-H.323 calls, no user configuration is needed for DTMF interworking, which is triggered solely by capability negotiation.
- The SBC does not support multipoint capabilities.

Codec Mappings

The following codec mappings (Table 20-1) are used by the SBC to represent H.245 codecs as SDP for the purpose of:

- Billing records (media relay only)
- DBE programming (media relay only)
- Bandwidth allocation (media relay and media bypass). The bandwidth here is calculated based on the SDP, not directly from the H.245.

H.245 codec	appears as:
g711Alaw64k	PCMA/8000
g711Ulaw64k	PCMU/8000
g722_64k	G722/8000
g7231	G723.1/8000
g728	G728/8000
g729	G729/8000
g729AnnexA	G729/8000
g729wAnnexB	G729/8000
g729AnnexAwAnnexB	G729/8000
gsmHalfRate	GSM-HR/8000
gsmFullRate	GSM/8000
All other audio codecs	PCMU/8000 (the default codec)
T.38	See T.38 Fax Relay, page 33-11

Note the following:

- H.245 video and data codecs, other than T.38 and also H.261, H.263, H.264 for H.323-H.323 calls, are not supported by the SBC for media relay and media bypass.
- The subset of codecs supported for H.323/SIP interworking is much smaller (for more information see the H.323 to SIP Interworking chapter)

For general information, see the Codec Handling chapter.

DTMF Interworking

Dual-tone multi-frequency (DTMF) tones are used to transfer user requests. Different systems may support different forms of DTMF. The SBC enables the DTMF interworking between these systems.

For example, some nonstandard H.323 devices do not support the lowest common denominator of alphanumeric UserInputIndication. Such devices can only signal DTMF through RFC2833 telephony events or as in-band media data. Other devices support UserInputIndication but not the RFC2833 telephony event.

If two such devices are deployed back to back, their only option is to send DTMF tones as it is done in-band media data. Deploying an SBC between them allows each side to send DTMF, using its supported signaled method, UserInputIndication on one side and RFC2833 on the other, with the SBC interworking between the two.

This function requires the SBE to program the DBE to enable interception and insertion of RFC2833 DTMF on a particular side of the call—the side facing the RFC2833-only device. The SBE and DBE then collaborate to transfer DTMF signaling between the H.245 control channel and the RTP steam.

DTMF interworking is negotiated through TerminalCapabilitySet. Therefore, the SBC must be capable of extending the TerminalCapabilitySet to advertise support for both alphanumeric and RFC2833 methods.

The feature described in this section replaces all previous H.323 DTMF interworking functions. H.323 calls must support DTMF interworking between alphanumeric UserInputIndication and RFC2833. In this case, the SBE coordinates with the DBE to carry out DTMF insertion and interception in the Real-Time Protocol (RTP).

DTMF interworking is negotiated through TerminalCapabilitySet, not manual configuration. Therefore, the SBC must always advertise support for both alphanumeric and RFC2833 methods, if necessary by extending the TerminalCapabilitySet on its way through. (The exception is a TCS=0.)

See also the Implementing Interworking DTMF chapter for general information.

Restrictions for DTMF Interworking

The restrictions for DTMF internetworking are as follows:

- The alphanumeric UserInputIndication method and DTMF RFC 2833 are supported for DTMF interworking.
- The SBE assumes that a peer, advertising any form of UserInputCapability is capable of sending and receiving alphanumeric DTMF.
- No manual configuration is provided to force DTMF interworking to occur.
- Detection or insertion of DTMF as in-band audio data is not supported.

Transcoding

The SBC supports transcoding of slow start calls, enabling communication between different endpoints with different codecs, which otherwise cannot communicate with each other. Two H.323 endpoints deployed back-to-back might fail to agree on a mutually acceptable codec.

A typical case might be where one side is insisting on a low-bandwidth codec (such as ITU-T G.729) because of bandwidth constraints or administrative policy, and the other side only supports G.711. For example, if the calling party uses g711alaw and the callee uses G.729 annex B, the SBC can convert G.711alaw codec to g729 annex B codec and enable communication between the two. When the SBC detects that codec negotiation is needed, it uses Cisco Voice Interworking Service Module (VXSM) in the Cisco MGX 8880 switch as its media gateway to perform the transcoding. Deploying the SBC between the endpoints, in conjunction with an MGX 8880 transcoder, allows such calls to succeed.

The previous releases supported a fast-start-only version of transcoding. This function is now replaced with an implementation of transcoding that is triggered off TerminalCapabilitySet.

Transcoding is supported only for SIP-SIP calls.

See also the ?\$paratext[CT_ChapTitle]>? chapter for general transcoding information.

Restrictions for Transcoding

The restrictions for H.323 transcoding are:

- No transcoding support for SIP-H.323 and H.323-H.323 calls.
- The decision whether to use a transcoder is taken once per call, and is not modified if endpoints issue updated TerminalCapabilitySets (including TCS=0).
- When transcoding is required, the SBC enforces symmetric codecs for the call.
- Transcoding is never invoked in a fast start call. If no channels are suitable, endpoints must drop to slow start at which point transcoding may be invoked.
- The only codecs supported for transcoding are G.711 (PCMU and PCMA) and G.729 (with and without annex B), and the only transcoder tested with them is the Cisco MGX 8880 switch.

RAS Tech Prefix

A technology prefix is an optional H.323 standard-based feature, supported by gateways and gatekeepers, that enables more flexibility in call routing within an H.323 VoIP network. The gatekeeper uses technology prefixes to group endpoints of the same type together. Technology prefixes can also be used to identify a type, class, or pool of gateways. This feature provides per-adjacency configuration of RAS Tech Prefix and registers this prefix with the gatekeeper.

An H.323 adjacency may now be optionally configured with a single tech prefix consisting of 1-32 dialed digits. It publishes the tech prefix to the gatekeeper in the following field of the RAS registration

request (RRQ):

terminalType.gateway.protocol.voice.supportedPrefixes.

As with existing adjacency configuration, this field may not be changed while the adjacency is attached. This feature works in conjunction with existing SBC support for adding or removing digits on dialed numbers (see the Number Manipulation section in the ?\$paratext[CT_ChapTitle]>? chapter).

Restrictions for RAS Tech Prefix

• This feature does not support zone prefixes, for example, registration of prefix AliasAddresses with the gatekeeper.

User Protocol Timer Control

H.323 standards recommends timers, timeout, and retry counts for various messages. Their values are not fixed and represent a range. The ability to define these values facilitates interworking between different devices. H.323 timers and retry counts can be now configured by the user at a global and per-adjacency level. Timers are expressed in seconds.

The following Q.931/H.225 timers are configurable.

- Q.931/H.225 Setup Timer T303
- Q.931/H.225 Establishment Timer T301
- Q.931/H.225 Incoming Call Proceeding Timer T310

The following RAS timeout and retry counts are configurable.

- GRQ
- RRQ
- URQ
- ARQ
- BRQ
- DRQ

The RAS RRQ TTL and keepalive times (governing lightweight RRQ behavior) are configurable. These two settings are interrelated. If the user configures unsafe values for a given adjacency, the SBE reverts to the defaults.

The adjacency retry timer is configurable, and can be used to automatically reattempt adjacency attachment when an adjacency fails for any reason.

The following timers are hardcoded:

- TCP shutdown timeout—when gracefully closing a TCP connection, the time allowed for remote closure before closing it ungracefully. The hardcoded value is 1 second.
- TCP connect timeout—time allowed before giving up on a TCP connection attempt to a remote peer. The hardcoded value is 1 second.

Restrictions for User Protocol Timer Control

User protocol timer control restrictions are:

- Changing timer values or retry counts while adjacencies are attached is allowed, but does not affect timers' or gatekeeper's transactions that are already in progress.
- No facility is provided to configure all RAS timeouts at once.
- H.245 timers are not included here since they only run in interworking scenarios.
- The SBC does not support the configuration of the following Q.931/H.225 timers:
 - Q.931/H.225 Overlap Sending Timer T302

- Q.931/H.225 Overlap Receiving Timer T304
- Q.931/H.225 Status Timer T322
- The SBC does not support the configuration of the following RAS timers:
 - IRQ
 - IRR
 - RAI
 - SCI

T.38 Fax Relay

This feature provides support for media relay of T.38 fax. The following features are supported:

- Both fax-only and fax-plus voice calls.
- Switchover from voice to T.38 fax.
- T.38 relay over unnumbered datagram protocol transport layer (UDPTL) only, and unidirectional H.245 channels only.

T.38 H.245 - SDP Mapping

The T.38 H.245—SDP mapping is shown below:

```
DataApplicationCapability
application
  t38fax
                             m=image 40000 {udpt1 | tcp} t38
  t38FaxProtocol
   t38FaxProfile
    fillBitRemoval
                             a=T38FaxFillBitRemoval
    transcodingJBIG
                             a=T38FaxTranscodingJBIG
    transcodingMMR
                             a=T38FaxTranscodingMMR
   version
                             a=T38FaxVersion:<digits>
    t38FaxRateManagement
                             a=T38FaxRateManagement:{localTCF | transferredTCF}
    t38FaxUdpOptions OPTIONAL
    t38FaxMaxBuffer
                             a=T38FaxMaxBuffer:<digits>
    t38FaxMaxDatagram
                           a=T38FaxMaxDatagram:<digits>
    t38FaxUdpEC
                             a=T38FaxUdpEC:{t38UDPFEC | t38UDPRedundancy}
    t38FaxTcpOptions OPTIONAL
    t38TCPBidirectionalMode [no mapping]
  maxBitRate
                             a=T38maxBitRate:<digits> (UDP only)
```

The only parameters needed for media relay function are the port and the peak-bit rate, which are highlighted in the example. Therefore, the presence of a T.38 fax function causes the following SDP to be transmitted to the DBE:

m=image <remote T.38 port> udptl t38
a=T38maxBitRate:14400

For interworking scenarios, a complete mapping needs to be carried out.

H.245 Mode Request

Switchover from a voice to fax call is handled by a RequestMode exchange. In an H.323-H.323 call this exchange is passed through transparently between call legs without DBE signaling. This allows endpoints to coordinate replacement of audio with T.38 channels.

RAS Maximum Bit Rate

In accordance with H.323v5 standards, the SBC counts UDP but not TCP towards the maximum bit rate agreed with the gatekeeper.

H.323 Annex D / T.38 Annex B Interoperability

T.38 Annex B is a fast start only (no H.245) version of H.323 Annex D. Interoperation with Annex B nodes is not supported by the SBC.

Restrictions

The restrictions are as follows:

- The SBC cannot be configured to advertise the t38FaxAnnexbOnly field of SupportedProtocols in RAS messages, and ignores this field on receipt.
- No support for TCP or Secure Real-Time Transport Protocol (SRTP) transport.
- No support for bidirectional H.245 channel signaling.

0.931/H.225 Passthrough

This feature enables message elements from Q.931/H.225 to be passed through between two H.323 call legs. This section describes the "base passthrough profile" of the SBC, listing the parts of the Q.931/H.225 message that may be passed through.

The following conventions are used in the base passthrough profile:

- ASN.1 syntax for Q.931 / H.225 messages is reproduced in this document.
- The following tags are attached to ASN.1 subtrees, specifying the passthrough behavior.
 - P = "passthrough". This subtree is passed opaquely between call legs.
 - P* = "passthrough with privacy implications". Similar to "P", but passing through this subtree may reveal information about an endpoint or a remote telephone number.
 - B = "block". This subtree is unconditionally blocked by the SBC and any information contained in it is lost.
 - SBC. This subtree is manipulated by the SBC. Typically, values are replaced by those local to the SBC.

Call Proceeding Passthrough

A Call Proceeding message is never passed through. However, fields from it are extracted and put into a Progress or Facility in the upstream call log.

• A Progress is used if the Call Proceeding contains a progress indicator.

• A Facility is used otherwise.

Unsupported Messages

The following ITU-T Q.931 messages are not supported by the SBC either because they are forbidden in H.323 or because the SBC does not currently support their corresponding features.

- Status, Status Enquiry
- SetupAck
- Information
- Notify
- userInformation.

Privacy

Subtrees marked as "P*" - "passthrough with privacy implications" are automatically blocked if the outgoing call leg has privacy enabled in CAC policy. This automatic blocking cannot be overridden by configuration, therefore, the only way to have these fields pass through is to disable privacy.

Setting of Protocol Version

When passing through messages, the SBC sets the version of outgoing messages to the lower value of its own ASN.1 version from that received in the original protocol message.

0.931 / H.225 Base Passthrough Profile

Q

931Message	
protocolDiscriminator	SBC
callReferenceValue	SBC
message	
setup	
sendingComplete	Р
bearerCapability	Р
facility	P
progressIndicator	P
progressIndicator31	P
notificationIndicator	P
display	P*
keypadFacility	P
signal	Р
callingPartyNumber	SBC
callingPartySubaddress	В
calledPartyNumber	SBC
calledPartySubaddress	В
redirectingNumber	P*
userUser	
h323-uu-pdu	
h323-message-body	
setup	
protocolIdentifier	SBC
h245Address	SBC
sourceAddress	SBC
sourceInfo	SBC
destinationAddress	SBC
destCallSignalAddress	SBC

destExtraCallInfo	В
destExtraCRV	В
activeMC	Р
conferenceID	Р
conferenceGoal	Р
callServices	Р
callType	в
sourceCallSignalAddress	SBC
remoteExtensionAddress	В
callIdentifier	Р
h245SecurityCapability	В
tokens	В
cryptoTokens	В
fastStart	SBC
mediaWaitForConnect	Р
canOverlapSend	В
endpointIdentifier	P*
multipleCalls	SBC
maintainConnection	SBC
connectionParameters	Р
language	Р
presentationIndicator	SBC
screeningIndicator	SBC
serviceControl	Р
symmetricOperationRequired	Р
capacity	в
circuitInfo	SBC
desiredProtocols	в
neededFeatures	в
desiredFeatures	В
supportedFeatures	В
parallelH245Control	В
additionalSourceAddresses	В
nonStandardData	Р
h4501SupplementaryService	Р
h245Tunneling	SBC
h245Control	SBC
nonStandardControl	Р
callLinkage	Р
tunnelledSignallingMessage	Ρ
provisionalRespToH245Tunneling	SBC
stimulusControl	Р
genericData	Ρ
user-data	Р
callProceeding	
bearerCapability	Р
facility	Р
progressIndicator	SBC
progressIndicator31	SBC
notificationIndicator	Р
display	Р
userUser	
h323-uu-pdu	
h323-message-body	
callProceeding	a= -
protocolIdentifier	SBC
destinationinto	Υ* στο
n245Adaress	SBC
callidentitier	Р
11245SecurityMode	Б
LOKENS	ы р
fagtStart	B CDC
multipleCalle	SDC
mut ctbrecattp	JDC

maintainConnection fastConnectRefused	SBC SBC
featureSet	В
nonStandardData	Р
h4501SupplementaryService	Р
h245Tunneling	SBC
h245Control	SBC
nonStandardControl	Ρ
callLinkage	P
tunnelledSignallingMessage	Р
provisionalRespToH245Tunneling	SBC
stimulusControl	P
genericData	P
user-data	Р
bearerCapability	л
facility	г D
nrogreggIndicator	r GBC
progressIndicator31	SBC
notificationIndicator	P
display	- р*
signal	P
userUser	-
h323-uu-pdu	
h323-message-body	
alerting	
protocolIdentifier	SBC
destinationInfo	P*
h245Address	SBC
callIdentifier	Р
h245SecurityMode	В
tokens	В
cryptoTokens	В
fastStart	SBC
multipleCalls	SBC
maintainConnection	SBC
alertingAddress	P*
presentationIndicator	SBC
screeningIndicator	SBC
fastConnectRefused	SBC
serviceControl	P
capacity	В
teatureSet	В
h 45 01 Gunnal aman ta maga anni an	P
h4501SupplementaryService	P
h245Control	GBC
nonStandardControl	P
callLinkage	Þ
tunnelledSignallingMessage	P
provisionalRespToH245Tunneling	SBC
stimulusControl	P
genericData	Р
user-data	Р
connect	
bearerCapability	Ρ
facility	Р
progressIndicator	SBC
progressIndicator31	SBC
notificationIndicator	Ρ
display	P*
dateTime	Ρ
connectedNumber	P*
connectedSubaddress	P*

userUser	
h323-uu-pdu	
h323-message-body	
connect	
protocolIdentifier	SBC
h245Address	SBC
destinationInfo	р*
	т П
	P
callidentifier	Р
h245SecurityMode	В
tokens	В
cryptoTokens	В
fastStart	SBC
multipleCalls	SBC
maintainConnection	SBC
language	Р
connectedAddress	P*
presentationIndicator	SBC
screeningIndicator	SBC
factConnectRefused	GBC
acrui acControl	D
servicecontrol	P
capacity	в
featureSet	В
nonStandardData	Р
h4501SupplementaryService	Р
h245Tunneling	SBC
h245Control	SBC
nonStandardControl	Ρ
callLinkage	Р
tunnelledSignallingMessage	Р
provisionalRespToH245Tunneling	SBC
stimulusControl	Ρ
genericData	Ρ
genericData user-data	P P
genericData user-data progress	P P
genericData user-data progress bearerCapability	P P P
genericData user-data progress bearerCapability cause	P P P P
genericData user-data progress bearerCapability cause facility	P P P P P
genericData user-data progress bearerCapability cause facility progressIndicator	P P P P P SBC
genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31	P P P P SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator</pre>	P P P P SBC SBC P
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu b323-messare-body</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body</pre>	P P P SBC SBC P P
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier dastinationInfo</pre>	P P P SBC SBC P P*
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo</pre>	P P P SBC SBC P P* SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address</pre>	P P P SBC SBC P P* SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier</pre>	P P P SBC SBC P P* SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode</pre>	P P P SBC SBC P P* SBC SBC SBC SBC P B
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens</pre>	P P P SBC SBC P P* SBC SBC SBC SBC P B B
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC B B B B
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart</pre>	P P P SBC SBC P P* SBC SBC SBC SBC P B B B SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls</pre>	P P P SBC SBC P P* SBC SBC SBC SBC B B B SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData h4501SupplementaryService</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData h4501SupplementaryService h245Tunneling</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData h4501SupplementaryService h245Control</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData h4501SupplementaryService h245Tunneling h245Control nonStandardControl</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC
<pre>genericData user-data progress bearerCapability cause facility progressIndicator progressIndicator31 notificationIndicator display userUser h323-uu-pdu h323-message-body progress protocolIdentifier destinationInfo h245Address callIdentifier h245SecurityMode tokens cryptoTokens fastStart multipleCalls maintainConnection fastConnectRefused nonStandardData h4501SupplementaryService h245Tunneling h245Control nonStandardControl callLinkage</pre>	P P P SBC SBC P P* SBC SBC SBC SBC SBC SBC SBC SBC SBC SBC

provisionalRespToH245Tunneling	SBC
stimulusControl	Р
genericData	P
user-data	Р
releasecomplete	CDC
facility	P
notificationIndicator	P
display	- P*
signal	Р
userUser	
h323-uu-pdu	
h323-message-body	
connect	
protocolIdentifier	SBC
reason	SBC
callIdentifier	Р
tokens	В
crypto'l'okens	B B
progentationIndicator	P^ CDC
screeningIndicator	SBC
capacity	B
serviceControl	P
featureSet	в
nonStandardData	Р
h4501SupplementaryService	Р
h245Tunneling	SBC
h245Control	SBC
nonStandardControl	Р
callLinkage	Р
tunnelledSignallingMessage	Р
provisionalRespToH245Tunneling	SBC
stimulusControl	P
genericData	Р
user-data	
facility	ъ
notificationIndicator	P
display	- Р*
callingPartvNumber	- P*
calledPartyNumber	P*
userUser	
h323-uu-pdu	
h323-message-body	
facility	
protocolIdentifier	SBC
alternativeAddress	В
alternativeAliasAddress	Р
conferenceID	P
reason	Р
callidentiller	P
destExtracallinio	P
tokens	r B
cryptoTokens	B
conferences	P
h245Address	SBC
fastStart	SBC
multipleCalls	SBC
maintainConnection	SBC
fastConnectRefused	SBC
serviceControl	Ρ

featureSet	В
destinationInfo	P*
h245SecurityMode	В
empty	
nonStandardData	Ρ
h4501SupplementaryService	Ρ
h245Tunneling	SBC
h245Control	SBC
nonStandardControl	Ρ
callLinkage	Ρ
tunnelledSignallingMessage	Ρ
provisionalRespToH245Tunneling	SBC
stimulusControl	Р
genericData	Р
iser-data	Ρ

Restrictions

- Any message elements from Q.931/H.225 that are not listed in this section cannot be passed through.
- Passthrough of security tokens is not supported.

H.323 Privacy

With the H.323 privacy feature, users can invoke identity hiding on Q.931/H.225 messages. When this feature is implemented, the SBC strips Q.931/H.225 message elements that reveal information about the remote caller or callee before passing them to the endpoints.

Note

The Q.931/H.225 message elements that impact privacy are defined in the H.323 passthrough profile.

The SBC applies the privacy service to a message if it contains a privacy request submitted by a user, or if a Call Admission Control (CAC) policy on the SBC is configured to enable privacy on a caller or callee basis. If, however, the privacy configuration fields are set to default values, then the SBC forwards the message to the next call leg without applying the privacy service to the message. You can also configure the SBC to provide the H.323 privacy service on a per-adjacency basis.

The SBC applies the following rules when providing the H.323 privacy service:

- If an H.323 adjacency is configured to allow private information, then the SBC does not apply privacy service even if an incoming message requests it or the CAC policy is configured to enable privacy.
- If an H.323 adjacency is not configured to allow private information, but a CAC policy is configured to enable privacy, then the SBC applies the privacy service to outgoing messages.
- If an incoming message requests the privacy service, but a CAC policy has not been configured to enable privacy, then the SBC applies the service if the adjacency is configured to apply the privacy service.
- If an incoming message requests the privacy service when both the CAC policy and the adjacency have not been configured to apply the privacy service, then the SBC does not apply the privacy service and allows the private information to pass through.

Restrictions and Limitations

Restrictions and limitations are as follows:

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- The SBC does not apply the H.323 privacy service to H.245 and RAS messages.
- Currently, the CAC policy for callee privacy is available for the H.323 signaling stack at "connect time", and only if a connectedNumber is present. As a result, the callee privacy service is not applied to the Q.931 protocol messages that pass through before or after a call is connected when a connectedNumber is not present. Due to this limitation, the SBC forwards the Q.931 Alerting, Q.931 Progress, and Q.931 Release Complete messages without applying the privacy service request to them.
- In an interworking call, the SBC only applies privacy requests based on the CAC policy.

H.245 Address in Call Proceeding

The SBC allows the address of the H.245 listening socket to be published in the Q.931 call proceeding message. When the caller does not support tunneling and an H.245 address published to the caller set to "wait-connect", H.323 supplies only the H.245 address on the Q.931 connect. In default behavior, H.323 supplies the H.245 address on a Q.931 call proceeding, and all subsequent messages to the caller until the H.245 connection is opened.

Multiple TCP for H.323

The SBC supports multiple TCP connections for an H.323 call, such as one H.225 signaling channel and one H.245 signaling channel if required.

Extending SIP Secure calls over H.323 Interface

The Cisco IOS XE Release 3.2S extends the security feature, by extending support of secure calls coming from an H323 adjacency or from a SIP adjacency. Before this enhancement, SBC only supported SIP secure calls and SIP secure calls were not able to interwork with H.323 networks. After this enhancement, SIP secure calls received from SIP adjacency and routed over H323 adjacency can be sent by configuring the H323 adjacency as trusted. Also, calls coming from an H323 adjacency and if it is required to be treated as a secure call then you can configure the H323 adjacency as secured.

Following are the restrictions of the SIP Secure calls over an H.323 interface:

- The SBC does not signal secure H.323 calls using the procedure described in H.235. It also does not recognize the secure nature of the incoming H.323 calls using the H.235 procedures.
- The SBC does not use a TLS or IPSec to send call signalling for secure H.323 calls.

Limited H.323 ID Routing and Passthrough Support

The Tandberg T3 device is a high-end 3-screen video-conferencing product that uses H.323 signaling. To direct the 3 video streams, it routes calls using the H.323 ID. The *Destination Number.Screen Identifier* H.323 ID format is used. For example, *12221120008.left*, indicates that the T3 device is identified using the e164 number 12221120008, and the video stream that is being negotiated is to be presented on the left side of the screen. Similarly, H.323 negotiations for other streams with *.right* and *.centre* suffixes, and the same numeric prefix can be made.

Prior to Cisco IOS XE Release 3.3S, these calls would fail when the SBC is inserted into the signaling path for the following reasons:

- The call is not routed correctly, because the SBC does not read the information in the H.225 destination address field, and this is the only field in which routing information is made available by the T3.
- The destinationAddress field is blocked by the SBC, so even if the call were routed correctly, the receiving T3 does not have the information of the screen that has to be presented with the video stream.

From Cisco IOS XE Release 3.3S, when the SBC receives a Q.931 message that contains no routing information in the standard called party number (CdPN) field, it examines the H.225 destinationAddress field. If the field contains information of type e164, the contents is copied into the CdPN field of the message. However, if the field contains information of H.323 ID type, the SBC copies the numeric prefix of this field into the CdPN field of the message.

For information on configuring the Limited H.323 ID Routing and Passthrough Support feature, see the Configuring Limited H.323 ID Routing and Passthrough Support, page 33-28.

Restrictions for Limited H.323 ID Routing and Passthrough Support

The Limited H.323 ID Routing and Passthrough Support feature has the following restrictions:

- This feature is not supported on an inbound SIP adjacency, as the configuration is done on the inbound H.323 adjacencies.
- If the inbound adjacency for a call is H.323, you can derive the caller and the callee numbers from the H.225 and H.323 ID addresses by setting *h225 address usage* to *prefix*. You cannot pass through the complete H323-ID format addresses directly.
- H.225 addresses cannot pass through during the H.323-SIP interworking because an AliasAddress field cannot be inserted in a SIP message.

Configuring H.323 Features

This section contains the following:

- Configuring H.323 Slow Start to H.323 Fast Start Interop, page 33-21
- Configuring Separate H.245 Control Channel, page 33-22
- Configuring RAS Tech Prefix, page 33-23
- Configuring User Protocol Timer Control, page 33-24
- Configuring H.323 Privacy, page 33-26
- Configuring H.245 Address in Call Proceeding, page 33-27
- Configuring Limited H.323 ID Routing and Passthrough Support, page 33-28

Configuring H.323 Slow Start to H.323 Fast Start Interop

The following example describes how to configure the SBC to use the fast start mode of operation for outgoing calls; and to effect the interop capability that enables the incoming slow start calls to be converted to fast start calls as they cross the SBC.

SUMMARY STEPS

- 1. configure terminal
- **2. sbc** *service-name*
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. start fast
- 6. exit

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the SBE function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the mode of an SBE H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	<pre>start fast Example: Router(config-sbc-sbe-adj-h323)# start fast</pre>	Specifies that the SBC uses the fast start mode of operation for call setup for outgoing calls. The interop capability enables the incoming slow start calls to be converted to fast start calls as they cross the SBC.
Step 6	exit	Exits the SBE H.323 adjacency mode to the SBE mode.
	Example:	
	Router(config-sbc-sbe-adj-h323)# exit	

Configuring Separate H.245 Control Channel

This command disables tunneling on a per-adjacency basis, facilitating interoperability with existing devices that are confused by tunneling. The command controls both incoming and outgoing calls.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. h245-tunnel disable
- 6. exit

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the SBE function of the SBC.
Step 4	Example: Router(config-sbc)# sbe adjacency h323 adjacency-name	Enters the mode of an SBE H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	h245-tunnel disable Example: Router(config-sbc-sbe-adj-h323)# h245-tunnel disable	Disables tunneling on a per-adjacency basis, facilitating interoperability with existing devices that are confused by tunneling. The command controls both incoming and outgoing calls.
		The default is tunneling enabled.
Step 6	exit	Exits the SBE H.323 adjacency mode to the SBE mode.
	Example: Router(config-sbc-sbe-adj-h323)# exit	

Configuring RAS Tech Prefix

This feature provides per-adjacency configuration of RAS Tech Prefix and registers this prefix with the gatekeeper. RAS tech prefix may consist of 1-32 dialed digits.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. tech-prefix tech-prefix
- 6. exit

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the mode of an SBE H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	<pre>tech-prefix tech-prefix Example: Router(config-sbc-sbe-adi-b323)# tech-prefix 32#</pre>	Provides per-adjacency configuration of RAS Tech Prefix and registers this prefix with the gatekeeper. RAS tech prefix may consist of 1-32 dialed digits followed by a # sign.
		The default is no tech prefix.
Step 6	exit	Exits the SBE H.323 adjacency mode to the SBE mode.
	Example:	
	Router(config-sbc-sbe-adj-h323)# exit	

Configuring User Protocol Timer Control

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. h323 | adjacency h323 adjacency-name
- 5. adjacency timeout *value*
- 6. h225 timeout
- 7. ras retry
- 8. ras rrq ttl *value*
- 9. ras rrq keepalive value
- 10. ras timeout
- 11. exit
- 12. show sbc *sbc-name* sbe h323 timers

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example:Router Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the SBE function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	h323 adjacency h323 adjacency-name	Enters the mode of either all H.323 adjacencies or a specified H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	adjacency timeout value	Defines the time in milliseconds, during which in case of failure to connect, the SBC keeps trying to
	Example: Router(config-sbc-sbe-adj-h323)# adjacency timeout	reconnect to the remote signaling peer and receive keep-alive messages from it.
	10000	The value range is 10000—30000.

	Command or Action	Purpose
Step 6	h225 timeout [establishment timeout-value proceeding timeout-value setup timeout-value	Defines the time for waiting to receive H.225 messages.
	Example: Router(config-sbc-sbe-adj-h323)# h225 timeout establishment 250000	• establishment <i>timeout-value</i> —h225 establishment state timeout value in milliseconds. The default is 180000. The value range is 30000-300000.
		• proceeding <i>timeout-value</i> —h225 proceeding state timeout value in milliseconds. 10000. The value range is 1000-30000.
		• setup <i>timeout-value</i> —h225 setup timeout value in milliseconds. The default is 4000. The value range is 1000-30000.
Step 7	ras retry [arq brq drq grq rrq urq] retry count	Defines the number of times the system trys to re-send RAS messages in case of failure to send the messages.
	Example: Router(config-sbc-sbe-adj-h323)#	• arq retry count—Number of times to retry an ARQ transaction.
	ras retry arq 2 ras retry brq 2 ras retry drq 2 ras retry rrq 2 ras retry urq 2	• brq retry count—Number of times to retry a BRQ transaction.
		• drq retry count—Number of times to retry a DRQ transaction.
		• grq retry count—Number of times to retry a GRQ transaction.
		• rrq retry count—Number of times to retry an RRQ transaction.
		• urq retry count—Number of times to retry a URQ transaction.
		The value range is 0-30.
Step 8	ras rrq ttl <i>value</i>	Defines the time to live messages (TTL) in seconds for registration request (RRQ).
	Example: Router(config-sbc-sbe-adj-h323)# ras rrq ttl 100	The default is 60. The value range is 16—300.
Step 9	ras rrq keepalive <i>value</i>	Defines the time in milliseconds for registration request (RRQ) keep-alive messages.
	Example: Router(config-sbc-sbe-adj-h323)# ras rrq keepalive 100000	The default is 45000. The value range is 15000—150000.

	Command or Action	Purpose
Step 10	ras timeout [arg brg drg grg rrg urg] timeout	Defines the common timeout in milliseconds for all RAS messages.
	Example: Router(config-sbc-sbe-adj-h323)# ras timeout arq 1000 ras timeout brq 1000 ras timeout drq 1000 ras timeout grq 1000 ras timeout rrq 1000 ras timeout urq 1000	 arq timeout—Timeout value for an ARQ transaction. brq timeout—Timeout value for an BRQ transaction. drq timeout—Timeout value for an DRQ transaction. grq timeout—Timeout value for an GRQ transaction. rrq timeout—Timeout value for an RRQ transaction. urq timeout—Timeout value for an URQ transaction. The default is 5000. The value range is 1000-45000. The default is 5000.
Step 11	exit	Exits the H.323 global or specified adjacency mode.
	Example: Router(config-sbc-sbe-adj-h323)# exit	
Step 12	show sbc service-name sbe h323 timers	Displays the values of all H.323 timers.
	Example: Router# show sbc mysbc sbe h323 timers	

Configuring H.323 Privacy

This feature allows the SBC to apply the H.323 privacy service on outbound messages.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. allow private info
- 6. privacy restrict outbound
- 7. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the SBE function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the mode of an SBE H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	<pre>allow private info Example: Router(config-sbc-sbe-adj-h323)# allow private info</pre>	Configures the H.323 adjacency to allow private information on messages sent out by the adjacency even if the CAC policy is configured to apply privacy service or the user requests privacy service. The no version of this command configures the H.323 adjacency to stop allowing private information from being sent out by the adjacency.
Step 6	<pre>privacy restrict outbound Example: Router(config-sbc-sbe-adj-h323)# privacy restrict outbound</pre>	Configures the H.323 adjacency to apply privacy restriction on outbound messages if the user requests the privacy service. The no version of this command configures the H.323 adjacency to allow private information messages sent out by the adjacency.
Step 7	exit	Exits an SBE H.323 global or specified adjacency mode.
	Example: Router(config-sbc-sbe-adj-h323)# exit	

Configuring H.245 Address in Call Proceeding

This feature allows the address of the H.323 listening socket to be published in the Q.931 call proceeding message.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name

- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. h245-address-pass wait-connect
- 6. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of the SBE function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the mode of an SBE H.323 adjacency.
	Example: Router(config-sbc-sbe)# adjacency h323 2651XM-5	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	h245-address-pass wait-connect	Configures the H.323 adjacency to allow delay passing the H.245 address to caller.
	Example: Router(config-sbc-sbe-adj-h323)# h245-address-pass	If set to wait-connect , H.323 supplies only the H.245 address on the Q.931 connect.
	walt-connect	The no form of this command shows default behavior, where H.323 supplies the H.245 address on a Q.931 call proceeding, and all subsequent messages to the caller until the H.245 connection is opened.
Step 6	exit	Exits an SBE H.323 global or specified adjacency mode.
	Example: Router(config-sbc-sbe-adj-h323)# exit	

Configuring Limited H.323 ID Routing and Passthrough Support

This section shows how to configure the Limited H.323 ID Routing and Passthrough Support feature:

- Configuring H.225 Address Passthrough, page 33-29
- Configuring H.225 Address Usage, page 33-30

Configuring H.225 Address Passthrough

This task shows how to configure the SBC to block the sourceAddress and destinationAddress fields in H.225 messages received on the H.323 adjacency. When the SBC is configured to block, these fields are not sent out of the outbound H.323 adjacency.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. h225 address block
- 6. end
- 7. show sbc sbc-name sbe adjacencies adjacency-name detail

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode. Use the <i>sbc-name</i> argument to define the name of the SBC.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the SBE H.323 adjacency mode. Use the <i>adjacency-name</i> argument to define the name of the
	Example: Router(config-sbc-sbe)# adjacency h323 h323adj	H.323 adjacency.
Step 5	h225 address block	The sourceAddress and destinationAddress fields in the H.225 message that are received on the H.323 adjacement are not received through
	<pre>Example: Router(config-sbc-sbe-adj-h323)# h225 address block</pre>	By default, the sourceAddress and destinationAddress fields are not blocked.

	Command or Action	Purpose
Step 6	end	Exits the SBE H.323 adjacency mode and enters the Privileged Exec mode.
	Example:	
	Router(config-sbc-sbe-adj-h323)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Lists the adjacencies configured in the SBE.
	Example:	
	Router# show sbc mysbc sbe adjacencies h323adj detail	

Configuring H.225 Address Usage

This task shows how to interpret H.225 sourceAddress and destinationAddress fields when the Q.931 callingPartyNumber or calledPartyNumber fields are not present. When callingPartyNumber is not provided in the Q.931 part of the message, the sourceAddress in the H.225 is checked. Similarly, when the calledPartyNumber field is not present, the destinationAddress is checked.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency h323 adjacency-name
- 5. h225 address usage {e164 | h323id}
- 6. end
- 7. show sbc sbc-name sbe adjacencies adjacency-name detail

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mysbc	Use the <i>sbc-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	adjacency h323 adjacency-name	Enters the SBE H.323 adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency h323 h323adj	Use the <i>adjacency-name</i> argument to define the name of the H.323 adjacency.
Step 5	h225 address usage {e164 h323id} Example: Router(config-sbc-sbe-adj-h323)# h225 address usage e164	Specifies either of the following interpretation formats for the H.225 sourceAddress and destinationAddress fields in the adjacency when Q.931 callingPartyNumber or calledPartyNumber is not present:
		• e164 —Specifies the e164 format for the addresses. All the other formats are ignored.
		• h323id —If the field begins with a numeric prefix, such as [0123456789*,] of 6 or greater characters, it is used as the calling or the called party number, and the rest of the ID is ignored.
		By default, the addresses in the H.323-ID format are checked.
Step 6	end	Exits the SBE H.323 adjacency mode and enters the Privileged Exec mode.
	Example: Router(config-sbc-sbe-adj-h323)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Lists the adjacencies configured in the SBE.
	Example: Router# show sho mysho she adjacencies b323adj detail	

The following example lists the adjacency that is configured in the SBE using the **show sbc sbe adjacencies detail** command. The output also displays information about the H.225 messages.

Router# show sbc mysbc sbe adjacencies h323adj detail

SBC Service "sbc"	
Adjacency h323adj (H.323)	
Status:	Detached
Signaling address:	0.0.0.0:1720 (default)
Signaling-peer:	0.0.0.0:1720 (default)
Admin Domain:	None
Account:	
Media passthrough:	Yes
Group:	
Hunting triggers:	Global Triggers
Hunting mode:	Global Mode
Techology Prefix:	
H245 Tunnelling:	Enabled
Fast-Slow Interworking:	None
Trust-level:	Untrusted
Call-security:	Insecure
Realm:	None
Warrant Match-Order:	None
Local Jitter Ratio:	0/1000
H225 address block:	Enabled

H225 address usage: h

h323id (default)

Configuring Separate H.245 Control Channel and RAS Tech Prefix: Example

```
configure terminal
sbc mysbc
sbe
adjacency h323 h323-fxs-1b
signaling-address ipv4 88.110.128.13
signaling-port 1720
remote-address ipv4 10.0.0.0/8
signaling-peer 10.124.2.2
signaling-peer-port 1720
account h323-fxs-1b
tech-prefix 2#
h245-tunnel disable
attach
exit
```

Configuring User Protocol Timer Controls: Example

```
configure terminal
sbc mysbc
sbe
adjacency h323 abcd
adjacency timeout 10000
h225 timeout establishment 40000
adjacency timeout 10000?
h225 timeout ?
  establishment h225 establishment state timeout value.
 proceeding
                h225 proceeding state timeout value.
 setup
               h225 setup timeout value.
h225 timeout proceeding 30000
h225 timeout setup 30000
ras ?
 retrv
        RAS retry configuration.
          RRQ (Registration Request) configuration.
 rrq
 timeout RAS timeout configuration.
ras retrv ?
  arg Retry count for an ARQ transaction.
  brg Retry count for an BRQ transaction.
  drq Retry count for an DRQ transaction.
  grq Retry count for an GRQ transaction.
 rrq Retry count for an RRQ transaction.
 urg Retry count for an URQ transaction.
ras retry arq 2
ras retry brq 2
ras retry drq 2
ras retry rrq 2
ras retry rrq 2
ras rrq ?
 keepalive Rate for keepalive msgs to refresh an H323 adjacency registration.
           TTL (time to live) value for an RRO request.
 ttl
ras rrg keepalive ?
 <15000-150000> Keepalive refresh time in milliseconds - default: 45000 ras rrq
keepalive 15000
```
```
ras rrq ttl ?
 <16-300> TTL value in seconds - default: 60
ras rrq ttl 30
adjacency timeout 30000
ras timeout ?
  arg Timeout value for an ARQ transaction.
  brg Timeout value for an BRQ transaction.
  drq Timeout value for an DRQ transaction.
  grq Timeout value for an GRQ transaction.
  rrq Timeout value for an RRQ transaction.
  urg Timeout value for an URQ transaction.
ras timeout arq ?
  <1000-45000> Timeout value in milliseconds - default: 5000
ras timeout arg 1000
ras timeout brg 1000
ras timeout drg 1000
ras timeout grq 1000
ras timeout rrq 1000
ras timeout urg 1000
```



H.323 to SIP Interworking

The H.323 to SIP interworking capability is very important in Voice over IP (VoIP) services since both protocols are widely used in the industry. When one VoIP service provider uses Session Initiation Protocol (SIP) and another provider uses H.323, the two protocols need to interwork to enable the customers to contact each other. H.323 is an older protocol that is gradually supplanted by SIP. The customers who have their VoIP network managed using H.323 may have to transition to SIP in the future. During this transition, both protocols need to interwork on the customers' VoIP network.

The following H.323 to SIP interworking features are supported:

- H.323 to SIP Support for Emergency Calls, page 34-4
- H.323 Slow Start Calls to SIP Calls, page 34-4
- H.323 to SIP Cause Code Mapping, page 34-5
- SIP Calls to H.323 Fast Start Calls, page 34-7
- H.323 Fast Start Calls to SIP Calls, page 34-9
- SIP to H.323 Interworking for Basic Call Hold, page 34-10
- Overview: Extending the SIP Secure Calls over the H.323 Interface, page 34-11
- Configuring the SIP Secure Calls over an H.323 Interface, page 34-12
- Overview: Extending the SIP Secure Calls over the H.323 Interface, page 34-11
- Configuring the SIP Secure Calls over an H.323 Interface, page 34-12

In addition, T.38 fax passthrough is supported for SIP, H.323-H.323, and SIP-H.323 calls. See the Fax Support chapter for more information.



This feature is supported in the unified model in Cisco IOS XE Release 2.5 and later.

Feature History for H.323 to SIP Interworking

Release	Modification
Cisco IOS XE Release	H.323 to SIP interworking capability was introduced on the Cisco
2.5	ASR1000 Series Aggregation Services Routers.
Cisco IOS XE Release	Allow the secure SIP calls to be interworked with the H.323 networks on
3.2S	the Cisco ASR1000 Series Aggregation Services Routers.

Contents

This module contains the following sections:

- Restrictions for H.323-SIP Interworking, page 34-2
- Information About H.323-SIP Interworking, page 34-3
- H.323 to SIP Support for Emergency Calls, page 34-4
- H.323 Slow Start Calls to SIP Calls, page 34-4
- H.323 to SIP Cause Code Mapping, page 34-5
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- Configuring the SIP Secure Calls over an H.323 Interface, page 34-12
- Configuration Example: Implementing Secure SIP Calls over an H.323 Adjacency, page 34-14

Restrictions for H.323-SIP Interworking

The following features are not supported:

- Transcoding of interworking calls.
- Media bypass for interworking calls.
- H.323 DTMF signaling using any method other than the alphanumeric method of UserInputIndication.
- Interworking of endpoint registrations (not supported by H.323).
- Failover of interworking calls (because H.323 call legs cannot be preserved across a failover).
- Interworking of any SIP method other than INVITE, ACK, CANCEL, BYE, INFO, or PRACK.
- End-to-end authentication on an interworking call. For example, an H.323 call branch cannot challenge a SIP call branch and vice versa. The Session Border Controller (SBC) itself can challenge a SIP call branch, but not an H.323 call branch.
- User-configurable mapping of cause codes.
- User-configurable mapping of codec types.
- Interworking of signaling support for Silence Suppression/VAD. It is assumed that the majority of endpoints interoperate correctly without explicitly signaling silence suppression.
- Interworking of video calls.
- Interworking of fax calls partially. T.38 fax is supported for SIP, H.323-H.323, and SIP-H.323 calls.
- Payload interworking is not supported.

Information About H.323-SIP Interworking

Following the usual process, after the SBC applies the call and number policy tables, a final adjacency and account are chosen. In H.323 to SIP interworking, the originating and terminating adjacencies are configured for different protocols. For example, the originating adjacency can be configured for H.323 and the terminating adjacency can be configured for SIP.

H.323 has two modes of operation: slow start and fast start. The initiation of a call may proceed in a slow start or fast start in H.323. In a slow start, H.323 signaling consists of Setup, Call Proceeding, Alerting, and Connect steps. After these steps, the H.245 media negotiation is performed.

When a call is initiated in H.323 fast start, the H.245 media negotiation is performed within the initial Setup message.

The SBC supports the following features of H.323 to SIP and SIP to H.323 interworking:

- SIP upstream, H.323 fast-start downstream, offer received on the SIP INVITE. See Cisco Unified Border Element (SP Edition) supports interworking of upstream SIP endpoints calling to downstream H.323 Fast Start endpoints. This support includes support for Early Media., page 34-7
- SIP upstream, H.323 slow-start downstream, offer received on the SIP INVITE. First, H.323 fast-start is tried downstream. The SBC drops back to slow-start procedures when it discovers the downstream endpoint does not support fast-start. See H.323 Slow Start Calls to SIP Calls, page 34-4.
- SIP upstream, H.323 downstream (either fast-start or slow-start), no offer received on the SIP INVITE. See SIP Calls to H.323 Fast Start Calls, page 34-7.
- H.323 fast-start upstream, SIP downstream. See H.323 Fast Start Calls to SIP Calls, page 34-9.
- H.323 slow-start upstream, SIP downstream. SIP downstream is tried with a default SDP offer, containing a single media channel with the following offered codecs in decreasing order of preference: G.729, G.711 U-law, G.711 A-law, and G.723. See the ?\$paranum>H.323 Slow Start Calls to SIP Calls? section on page 34-4.
- Mapping of SIP response codes to H.225 error codes used by H.323 and mapping of H.225 error codes to SIP response codes. See H.323 to SIP Cause Code Mapping, page 34-5.
- Interworking for basic call hold feature to translate, hold, and resume signaling in H.323 and SIP interworking calls. See SIP to H.323 Interworking for Basic Call Hold, page 34-10.
- Early media in SIP calls to H.323 fast start calls. See Early Media Support, page 34-8.
- DTMF interworking between SIP and H.323 in the signaling plane, using the alphanumeric method of UserInputIndication.



All H.323 calls, including established H.323-H.323 and SIP-H.323 interworking calls, are disconnected upon an SBC switchover. An SBC switchover occurs when an active RP switches over to the standby RP in a hardware redundant system (such as a Cisco ASR 1006 Router) or when the active IOS process switches over to the standby IOS process in a redundant software system (such as a Cisco ASR 1002 Router).

When you are configuring the SBC to interwork calls between H.323 and SIP networks, you can also consider the following configuration tasks:

• For networks that use RFC2833 telephone-event signaling, you may want to configure telephone-event support on the H.323 or SIP side for improved call setup efficiency.

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- For DTMF interworking with H.323-SIP calls, you may want to configure the telephone-event payload type supported by the caller and callee through Call Admission Control (CAC) policy. This allows for improved call setup efficiency.
- To allow passthrough of display name updates, for example, following a third-party call transfer—you may want to whitelist the SIP Remote-Party-ID header.

H.323 to SIP Support for Emergency Calls

Cisco Unified Border Element (SP Edition) supports H.323 to SIP call routing for emergency calls. Cisco Unified Border Element (SP Edition) routes voice and video calls according to the configured session routing policy. A call is categorized as "emergency" based on the dialed number or on the Resource-Priority header if it is originated on the SIP side. Based on the emergency categorization, special routing and Call Admission Control (CAC) logic is applied.

H.323 Slow Start Calls to SIP Calls

Cisco Unified Border Element (SP Edition) supports interworking of H.323 Slow Start upstream calls made to a SIP endpoint.

As a result of a H.323 Slow start call, the downstream SIP end point has no media information from the calling endpoint at the time it sends the initial SIP INVITE. The SBC has the ability to send a default session description protocol (SDP) offer on the INVITE that proposes a single media stream for voice traffic, listing the following candidate codecs in decreasing order of preference: G.729, G.711 U-law, G.711 A-law, and G.723.

The answer received on the 200 OK response may have either reduced the number of codecs for the stream to 1 (called a "mono-answer"), or the 200 OK response still has multiple codecs in the stream (called a "multi-answer"). The "multi-answer" is unacceptable to the H.323 protocol. The SBC has the ability to refine the codec list by making a re-INVITE to the SIP endpoint containing only the first codec. Figure 34-1 shows the flows that take place during this process.

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Figure 34-1 H.323 Slow Start Calls to SIP Calls

H.323 to SIP Cause Code Mapping

Cisco Unified Border Element (SP Edition) supports mapping of SIP response codes to H.225 error codes used by H.323 and mapping of H.225 error codes to SIP response codes.

In H.323 to SIP interworking, the SBC provides call rejection with the proper cause code in the following manner:

- If a downstream SIP endpoint rejects a call, the response is translated into the H.225 error code set for the upstream H.323 device. The SIP endpoint may also reject attempts by the SBC to refine the codec list.
- If a downstream H.323 endpoint rejects a call, there are two possible actions—the H.323 gatekeeper may reject admission for the call, or the endpoint sends a Release Complete to reject the call.

Table 34-1 shows how SIP response codes are mapped to H.225 error codes.

SIP error code in	H.225 error code out
301	UnreachableDestination
302	UnreachableDestination
400	Undefined Reason
401	No Permission
403	Security Denied
404	UnreachableDestination
405	Undefined Reason
406	Undefined Reason
407	No Permission
408	Undefined Reason
410	Unreachable Destination
413	Undefined Reason
414	Undefined Reason
415	Undefined Reason
416	Undefined Reason
420	Undefined Reason
421	Undefined Reason
423	Undefined Reason
480	Destination Rejection
481	Unreachable Destination
482	Undefined Reason
483	Undefined Reason
487	Destination Rejection
488	Undefined Reason
501	Undefined Reason
503	Undefined Reason
504	Undefined Reason
505	Undefined Reason
513	Undefined Reason
603	Undefined Reason
604	Unreachable Destination
606	Undefined Reason

 Table 34-1
 SIP Response Codes Mapped to H.225 Error Codes

Table 34-2 shows how H.225 error codes are mapped to SIP response codes.

H.225 error code in	SIP error code out
NoBandwidth	500
UnreachableDestination	604
DestinationRejection	486
No Permission	401
GatewayResource	503
BadFormatAddress	404
SecurityDenied	403
InvalidRevision	503
UnreachableGatekeeper	503
AdaptiveBusy	503
InConf	503
RouteCallToGatekeeper	503
CallForwarded	503
RouteCallToMC	503
FacilityCallDeflection	503
CalledPartyNotRegistered	503
CallerNotregistered	503
ConferenceListChoice	503
StartH245	503
NewConnectionNeeded	503
NoH245	503
NewTokens	503
FeatureSetUpdate	503
ForwardedElements	503
TransportedInformation	503

Table 34-2 H.225 Error Codes Mapped to SIP Response Codes

SIP Calls to H.323 Fast Start Calls

Cisco Unified Border Element (SP Edition) supports interworking of upstream SIP endpoints calling to downstream H.323 Fast Start endpoints. This support includes support for Early Media.

Early Media Support

The capability for SIP endpoints calling to H.323 Fast Start endpoints includes support for Early Media, where the Fast Start response may be received before the Connect. Early Media can flow when the caller (SIP endpoint) makes a media proposal on the initial call setup request and the callee (the H.323 endpoint) responds to the offer before the call is connected. In this case, the H.323 endpoint expects an SDP offer on the initial INVITE.

H.323 may send a "Progress Indicator" on any H.225 message that it sends the SBC. A progress indicator of a value of 1 or 8 indicates that the H.323 endpoint will send early media. In an interworking call, only the first Progress Indicator received from the H.323 endpoint is acted upon.

If the H.323 endpoint sends a progress indicator with a value of 1 or 8, then in an interworking call with a SIP upstream call, if sufficient media parameters have been negotiated with the H.323 endpoint, the SBC returns a 183 provisional response to the SIP caller with the SDP indicating early media.

Depending on the Call Admission Control (CAC) configuration, the SBC may allow early media to be passed through at this point. If insufficient media parameters have been received to build the SDP to send to the SIP endpoint, then the SBC waits for media negotiation with the H.323 endpoint to reach a point where the SDP can be generated and then the SBC can send the 183 provisional response. Figure 34-2 shows the flows that take place during this process.



Figure 34-2 SIP with SDP Offer Call to H.323 Fast Start

H.323 Fast Start Calls to SIP Calls

Cisco Unified Border Element (SP Edition) supports interworking of H.323 Fast Start upstream calls made to a downstream SIP endpoint.

If the Fast Start offer from the H.323 device includes alternative codec options, the SDP offer sent to the downstream SIP device lists all of these alternative codecs in the same order of preference as that supplied by H.323. The most preferred codec is listed first. If the SIP endpoint accepts more than one codec, this is not acceptable on the H.323 Fast Start Response. Therefore, the SBC is able to refine the offer. The SBC makes another offer to the SIP device with a single codec option, by taking the most preferred codec listed in the SDP answer and constructing a new offer with that codec in it. If the downstream SIP device accepts this offer, then a FastStart response is returned, selecting that codec.

If the SIP endpoint SDP does not need to be refined, the fast start response will go back on the first available message to the H.323 endpoint.

The SIP endpoint may send early media as soon as it has sent the multi-SDP answer. However, the early media will fail to get through until the mono-SDP answer is received and processed, and the Progress Indicator and Fast Start response sent to the H.323 endpoint. See the ?\$paranum>Early Media Support? section on page 34-8.

Figure 34-3 shows the flows that take place during this process.



Figure 34-3 H.323 Fast Start Calls to SIP Calls

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SIP to H.323 Interworking for Basic Call Hold

The SIP/H.323 interworking for basic call hold feature enables Cisco Unified Border Element (SP Edition) to translate, hold, and resume signaling in H.323 and SIP interworking calls.

Note

Basic call hold does not require external configuration and is enabled by default.

SIP Requirements

In RFC-3264 SDP Offer-Answer protocol, basic call hold is signaled by a re-Offer that includes an a=sendonly', 'a=inactive', or 'c=IN IP4 0.0.0.0' line.

- a=sendonly or c=IN IP4 0.0.0.0 indicates that the offerer wants to keep transmitting. The Answer may optionally force the offerer to cease transmitting by setting a=inactive or c=IN IP4 0.0.0.0.
- a=inactive indicates that the offerer will also cease transmitting. In this case, the answerer must also reply with a=inactive.

Resume is signaled by setting the direction to a=sendrecv or, because this is the default setting, omitting the direction line altogether.

For SIP, requirements are:

- The SBC must support receipt of all of the above forms of call hold signaling. On transmit, control should preferably be provided over the form that is used.
- Translation of a re-offer that opens or closes the send direction (not just the receive direction).
- Case of the offerer or answerer changing their RTP address/port on a call hold resume offer or a call hold answer.
- Sending a re-Offer on a SIP re-INVITE and processing the answer on the INVITE 200 rsp.
- Processing an incoming answer on the first re-INVITE response even if that is not the final response (In this case, a duplicate answer on the final 200 response must be ignored).
- Receipt of a re-offer on a SIP INVITE request.
- Sending an answer on a re-INVITE 200 response.

H.323 Requirements

In H.245, basic call hold is signaled by sending an empty terminal capability set (defined in H.323 section 8.4.6, and known as "TCS=0"or "ECS"). The receiver of the TCS=0 must close its send channel and avoid re-opening it. Resume is signaled by sending a non-empty terminal capability set. At this point, the send channel is re-opened. In terms of the H.245 message flows:

- Terminal capabilities are transmitted using a TerminalCapabilitySet (TCS). This message is responded to with a TerminalCapabilitySetAck (TCS Ack) or TerminalCapabilitySetReject.
- A channel is opened with an H.245 OpenLogicalChannel (OLC). This is responded to with an OpenLogicalChannelAck (OLC Ack) or OpenLogicalChannelReject.
- A channel is closed with an H.245 CloseLogicalChannel (CLC). A CloseLogicalChannelAck (CLC Ack) indicates that this message has been processed.

For H.323, requirements are:

- Sending, receiving, and acting on empty and non-empty capability sets in an interworking call, including the situation in which both sides have put the other on hold.
- Translation of channel close and re-open outside the context of call hold / resume.
- Address changing for a new incarnation of a channel that uses different RTP/RTCP addresses/ports from the previous incarnation of the channel. (In line with existing behavior, SBC may continue to assume that each side of an H.245 RTP session uses a single RTP and RTCP IP address, and that the RTCP port = RTP port + 1.)
- Receiving TCS=0 from downstream before call connection.
- Ignoring a TCS=0 received from upstream before call connection (to prevent problems on the SIP side).

Basic Call Hold Restrictions

The following restrictions apply to the Basic Call Hold feature:

- Third-party rerouting (where a device separate from the SBC reroutes the call) is not fully supported.
- The SBC does not support the following SIP mechanisms for carrying out Offer-Answer exchanges:
 - Offer on an UPDATE request, INVITE 18x response, INVITE 200 response or PRACK request.
 - Answer to a renegotiation on anything other than an INVITE 200 response.
- No call hold during early media. Translation of call hold/resume is allowed only after a call is established. This restriction follows from the previous restriction that prevents the SBC from receiving or generating Offers on INVITE 18x or UPDATE in SIP.
- The SBC cannot originate or terminate H.450.4 call hold protocol.
- The SBC supports receipt of an OLC Ack with port set to 0, but the SBC does not transmit an OLC Ack with port set to 0.
- New timers are not configurable.
- Existing general interworking restrictions are still in place:
 - Only a single media stream is allowed.
 - Only a single audio codec is allowed within that stream.
 - Transcoding and DTMF interworking is not supported.
 - Media bypass is not supported.

Overview: Extending the SIP Secure Calls over the H.323 Interface

Data security has become the prime objective enhances service providers, corporates, and government institutes. Cisco IOS XE Release 3.2S enhances the security feature, by extending support to the secure calls coming from either a H323 adjacency or a SIP adjacency. Before this enhancement, the SBC supported only the SIP secure calls, and the SIP secure calls were not able to interwork with the H.323 networks. After this enhancement, the SIP secure calls received from a SIP adjacency and routed over an H323 adjacency can be sent by configuring the corresponding H323 adjacency as trusted. Also, calls coming from H323 adjacency can be configured as secure calls.

To configure an H.323 adjacency as trusted for handling the SIP secure calls received from a SIP adjacency, use the **trunk trusted** command. Defining an adjacency as trusted, distinguishes it from untrusted adjacencies. If an incoming call is a secure call, it goes through trusted adjacency. If no trusted adjacencies are configured, the incoming secure call is rejected with the SIP response code 403 (Forbidden) or an H.225 with the reason as Security Denied. If the incoming call is not a secure call, it can go through a trusted adjacency or untrusted adjacency.

To handle the calls coming from H.323 adjacency and to treat them as secure calls, configure the H.323 adjacency as secure using the **inbound secure** command. The outgoing SIP calls become a SIP-secure calls.

Prerequisites for the SIP Secure Calls over an H.323 Interface

Following are the prerequisites for the SIP Secure calls over an H.323 interface:

- The minimum software image required for this feature to work is the Cisco IOS XE 3.2S Software image.
- An H.323 adjacency must be configured as trusted before configuring the incoming calls as secure.

Note

All the H.323 adjacencies that are defined are by default untrusted. If you want to change an adjacency from trusted to untrusted, configure the incoming calls for the adjacency as insecure by using the **no inbound secure** command.

Restrictions for the SIP Secure Calls over an H.323 Interface

Following are the restrictions of the SIP Secure calls over an H.323 interface:

- The SBC does not signal secure H.323 calls using the procedure described in H.235. It also does not recognize the secure nature of the incoming H.323 calls using the H.235 procedures.
- The SBC does not use a TLS or IPSec to send call signalling for secure H.323 calls.

Configuring the SIP Secure Calls over an H.323 Interface

To implement the SIP Secure calls over an H.323 interface, configure the following:

- An H.323 outgoing adjacency as Trusted for handling the SIP secure calls received from the SIP adjacency.
- Incoming calls from an H.323 adjacency as secure calls for calls coming from an H.323 adjacency.

The following section provides procedure for configuring an H.323 adjacency as trusted and configuring incoming calls from an H.323 adjacency as secure calls:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbcname
- 3. sbe

- 4. adjacency h323 adjacency-name
- **5.** trunk trusted
- 6. inbound secure

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency h323 adjacency-name	Enters the H.323 adjacency mode to configure the parameters for the specified adjacency name.
	Example: Router(config-sbc-sbe)# adjacency h323 trust-h323-adj	
Step 5	trunk trusted	Configures the H.323 adjacency as trusted.
	Example: Router(config-sbc-sbe-adj-h323)# trunk trusted	
Step 6	inbound secure	Configures the incoming calls from the H.323 adjacency as secure calls.
	Example: Router(config-sbc-sbe-adj-h323)# inbound secure	Note If the H.323 adjacency is configured as untrusted, incoming calls cannot be configured as secure calls.

Configuration Example: Implementing Secure SIP Calls over an H.323 Adjacency

The following example shows how to configure an H.323 adjacency as Trusted and mark the incoming calls on an H.323 adjacency as secure calls:

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency h323 trust-h323-adj
Router(config-sbc-sbe-adj-h323)# trunk trusted
Router(config-sbc-sbe-adj-h323)# inbound secure
```

The following example displays the configuration details of trust-h323-adj:

Router# show sbc mySBC sbe adjacencies trust-h323-adj detail

```
SBC Service "mysbc"
 Adjacency trust-h323-adj (H.323)
                      Detached
   Status:
   Signaling address:
                          0.0.0.0:1720 (default)
   Signaling-peer:
                          0.0.0.0:1720 (default)
   Admin Domain:
                           None
   Account:
   Media passthrough:
                            Yes
   Group:
   Hunting triggers:
                    Global Triggers
   Hunting mode:
                            Global Mode
   Techology Prefix:
   H245 Tunnelling:
                          Enabled
   Fast-Slow Interworking: None
   Trust-level:
                           Trusted
   Call-security:
                            Secure
   Realm:
                            None
   Warrant Match-Order:
                            None
```



Support for H.239

H.239 is an extension to the H.323 family of specifications to allow a second video stream in parallel with the primary live video stream to share any type of content such as slides and spreadsheets. This second stream is one-way and considered important in video-conferencing where a viewer can see the speaker and in parallel, the presentation slides. This mode of the conference is controlled by an Multipoint Control Unit (MCU).

Cisco Unified Border Element (SP Edition) was earlier known as Integrated Session Border Controller, and is referred to as SBC in this document.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all the Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or the Cisco IOS master commands list.

Feature History of Support for H.239 on the Cisco Unified Border Element (SP Edition)

Release	Modification
Cisco IOS XE Release 3.3S	The Support for H.239 feature was introduced on the Cisco ASR 1000
	Series Routers.

Contents

- Information on Support for H.239, page 35-1
- Restriction for Support for H.239, page 35-2

Information on Support for H.239

H.245-based systems provide for multiple channels of video, while H.320 systems provide for only a single video channel. However, neither of these define a one-way transmission method, methods to label a video channel's content as a presentation video stream, or methods to control presentation video in a multipoint conference. H.239 provides these extensions, along with the ability to add an additional video channel to H.320.

It defines new capabilities, such as H239ControlCapability and H239ExtendedVideoCapability, which a terminal can advertise in its Terminal Capability Set (TCS) message. These capabilities indicate that the terminal can support H.239, and the roles, presentation with slides or live video, that the terminal can support. The SBC passes these capabilities between the H.323 endpoints without requiring to understand them.

The Support for H.239 feature defines a number of H.239 messages that can be carried within H.245 generic messages. These messages are used to negotiate tokens and general flow control. The MCU provides the token on request by a terminal to send a slide-set. This process ensures that at any point only one terminal within the conference is sending a slide-set because a terminal has to wait until the MCU provides the token before initiating the new video channel.

Once a terminal has a token, it can open an additional video channel to carry the presentation stream. This can be done using the standard H.245 messages such as OpenLogicalChannel. However, the OpenLogicalChannel message for the presentation stream has an additional h239ExtendedVideoCapability block, indicating the role of this video stream. The SBC is enhanced to

understand the h239ExtendedVideoCapability block.



An endpoint can choose to originate multiple additional video channels.

Restriction for Support for H.239

The Support for H.239 feature has the following restrictions:

- Interworking SIP-H.323 Video calls using H.239 is not supported.
- Redundancy for H.323 calls is not supported.
- A fast-start request cannot include a request to open an H.239 additional video channel as it is not supported.
- H.239 systems based on H.235 is not supported.
- The SBC does not support call transfer for H.323 calls. When an H.323 endpoint is placed on hold, it closes its media as well as video channels.



Implementing Billing on Cisco Unified Border Element (SP Edition)

The Cisco Unified Border Element (SP Edition) billing component includes the following core features:

- Compatibility with existing billing systems—To be able to fit the Cisco Unified Border Element (SP Edition) billing system easily into an existing billing architecture of a provider is an important functional requirement. This requirement entails the use of mechanisms to obtain billing information in a similar fashion to those of the existing mechanisms.
- Integration with next-generation technologies and solutions—Equally important is the requirement to use next-generation billing technologies, so that service information from Cisco Unified Border Element (SP Edition), softswitches, voicemail, and unified messaging applications, and so on can be collated and billed in a distributed environment.
- High availability and fault tolerance.
- Flexible architecture.

The billing component functions as a third-party integrated, distributed Remote Authentication Dial-In User Service (RADIUS)-based call and event logging.

The function of the billing component is:

• Third-party integrated, distributed Remote Authentication Dial-In User Service (RADIUS)-based call and event logging.

Note

This feature is supported in the unified model for Cisco IOS XR Software Release 2.4 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of commands used in this chapter, see Cisco Unified Border Element (SP Edition) Command Reference: Unified Model at

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 2.5	Support for Media Information and Granular Timestamp Support were added on the Cisco ASR 1000 Series Router.
Cisco IOS XE Release 2.6.1	Support for Adjacency Information was added on the Cisco ASR 1000 Series Router.
Cisco IOS XE Release 2.6.2	Support for Endpoint Information was added on the Cisco ASR 1000 Series Router.
Cisco IOS XE Release 3.2.0S	Support for XML based billing method was introduced on the Cisco ASR 1000 Series Router.
Cisco IOS XE Release 3.3S	The Selective RADIUS Billing feature was added on the Cisco ASR 1000 Series Router.

Feature History for Implementing Cisco Unified Border Element (SP Edition) Billing

Contents

This module contains the following sections:

- Prerequisites for Implementing Billing, page 36-2
- Information About Implementing Billing, page 36-3
- Support for Local Cache, page 36-6
- Support for Media Information, page 36-6
- How to Implement Billing, page 36-7
- Configuration Examples of Implementing Billing, page 36-13
- Selective RADIUS Billing, page 36-20
- Configuration Example of Selective RADIUS Billing, page 36-22

Prerequisites for Implementing Billing

The following prerequisites are required to implement Cisco Unified Border Element (SP Edition) billing:

- Before implementing interworking billing, Cisco Unified Border Element (SP Edition) must already be configured.
- To implement billing on the signaling border element (SBE) you must obtain a unique network element ID for the SBE from your network administrator. In addition, you must perform the following task depending on what form of billing you require.
 - To implement integrated RADIUS-based call logging, you must first configure the RADIUS server and set up the RADIUS network infrastructure.

Information About Implementing Billing

It is critical to understand all Cisco Unified Border Element (SP Edition) billing features and capabilities before performing billing configurations for the Cisco Unified Border Element (SP Edition). The following sections describe Cisco Unified Border Element (SP Edition) billing topologies:

- Integrated Billing Systems, page 36-3
- Granular Timestamp Support, page 36-4

Integrated Billing Systems

Integrated billing is achieved through the PacketCable Event Messages architecture (Figure 36-1 shows the *PacketCable 1.5 Event Messages Specification*; PKT-SP-EM1.5-I01-050128) where the Cisco Unified Border Element (SP Edition) is integrated into this architecture. As shown, the billing server and softswitch both support PacketCable Event Messages.

ISP-A in Figure 36-1 shows Cisco Unified Border Element (SP Edition) operating in a unified model where the billing system is being deployed as a distributed billing system consisting of three billing servers. Cisco Unified Border Element (SP Edition) can be configured to send to these servers in a range of ways, such as to all three simultaneously, or to use one primary and two backups.

In the unified model, the system operates as follows:

- Cisco Unified Border Element (SP Edition) produces event messages (EMs). These event messages are for billable or other interesting events, such as call start, call end, and media-type changes.
- Cisco Unified Border Element (SP Edition) and other elements of the system, which produces EMs, sends them in real time (or batched up for network efficiency) using the RADIUS protocol to the billing server.
- Billing server collates EMs into call detail records (CDRs). For an example of a CDR, see ?\$paranum>Example for Event Messages from Cisco Unified Border Element (SP Edition) to RADIUS Billing Server? section on page 37-14.
- Cisco Unified Border Element (SP Edition) supports local caching of records and event messages in the Cisco ASR 1000 Series Router's local disk in the event that none of the RADIUS servers are reachable.
- Cisco Unified Border Element (SP Edition) supports multiple RADIUS servers, for example, you can define multiple servers under a single client.

Note that ISP-B in Figure 36-1 shows Cisco Unified Border Element (SP Edition) operating in a distributed model where the billing system is being deployed using a single billing server and a softswitch.

Table 36-1 shows the packet billing termination codes that are supported by Cisco Unified Border Element (SP Edition).

Code Value	Description
0003	No route to destination
0016	Normal call clearing
0017	User busy
0019	User alerting: No answer

Table 36-1 Supported Packet Billing Termination Codes

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Code Value	Description	
0020	Subscriber absent	
0027	Destination out of order	
0028	Invalid number format (incomplete address)	
0031	Unknown: Call ended during recovery processing	
0041	Temporary failure	
0042	Switching equipment congestion	
0047	Resource unavailable, unspecified	
0063	Service or option not available, unspecified	
0065	Bearer capability not implemented	
0095	Invalid message, unspecified	
0097	Message type nonexistent or not implemented	
0099	Information element nonexistent or not implemented	
0103	Parameter non-existent or not implemented, passed on	
0111	Protocol error: Unspecified	
0127	Interworking: Unspecified	

Table 36-1 Supported Packet Billing Termination Codes (continued)



The *PacketCable 1.5 Event Messages Specification* discusses sending the identifying information (the BCID and FEID) on the outgoing INVITE and responding SDP so that correlation can be done between the two sets of billing data. Cisco Unified Border Element (SP Edition) does not support this mechanism for intra-domain or inter-domain transmission. The billing server must perform the correlation using an alternative method (for example, using the telephone numbers dialed and the time of the call).

Granular Timestamp Support

Cisco Unified Border Element (SP Edition) Billing Manager maintains a granular timestamp that billing methods can use to query the current time. The granular timestamp provides a precision of 100 milliseconds. This precision is sufficient for all billing requirements without having an impact on performance.

By default, the granular timestamp is set to the maximum of 100 milliseconds.

Endpoint Information in PacketCable Billing

Beginning Cisco Unified Border Element (SP Edition) Release 2.6.2, you can configure SBC to include information—adjacency name or addressing—on endpoints in use for a given call in the PacketCable billing records.

When SBC is not configured to include the endpoint information in the messages, the Signaling_Start messages for both sides of a call contains an MTA_Endpoint_Name attribute that contains the string *MTA Endpoint*. MTA_Endpoint_Name attribute is not included in Call Answer or Signaling_Stop Event Messages.

If you configure SBC to include the adjacency name, only the names of the endpoint adjacencies are included in the billing records. For example, *SIPPB*. If you have configured SBC to include the endpoint addressing information, then IP address, port, and transport type are also included in the billing records along with the adjacency name in the following format: *IP address,port,transport type,adjacency name*. For example, *2.0.0.36,5078,UDP,SIPPB*.

If SBC is configured to include the endpoint information:

- SBC adds the source adjacency name or addressing information, as configured, to the upstream Signaling_Start Event Messages. This information is included in the MTA_Endpoint_Name attribute—replacing the hard-coded string *MTA Endpoint*. At this point, the downstream Signaling_Start message contains only the hard-coded string—*MTA Endpoint*.
- SBC adds the destination adjacency name or addressing information, as configured, to downstream Call_Answer Event Messages. This information is included in the MTA_Endpoint_Name attribute. The upstream Call_Answer message does not contain an MTA_Endpoint_Name attribute.
- SBC includes both—source and destination—endpoint details in the Signaling_Stop messages; the source adjacency name or addressing information in the upstream message, and the destination adjacency name or addressing information in the downstream message. This ensures that even if a call fails to connect, the billing server still has access to both endpoint details.

Use the **[no] cdr endpoint-info {addressing | adjacency}** command to configure SBC to include the endpoint information in PacketCable billing.

Use the **show sbc** *sbcname* **sbe billing instance** command to verify whether the SBC is configured to include the endpoint information.

Restrictions for PacketCable Billing

H.323 is supported for PacketCable billing, but with some limitations. One such limitation is that no H.323 signalling address is present in PacketCable billing.

Performing ISSU for Endpoint Information

When performing ISSU to upgrade SBC from Release 2.6.1 to Release 2.6.2, if adjacency information (cdr adj-info) is provisioned for Release 2.6.1, then the corresponding endpoint-info adjacency (cdr endpoint-info adjacency) option is provisioned and the functionality is maintained.

When performing ISSU to downgrade SBC from Release 2.6.2 to Release 2.6.1, if endpoint adjacency information (cdr endpoint-info adjacency) is provisioned for Release 2.6.2, then the corresponding adjacency (cdr adj-info) option is provisioned and the functionality is maintained. If endpoint addressing information (cdr endpoint-info addressing) is provisioned for Release 2.6.2, no provisioning happens in Release 2.6.1.

Support for Local Cache

The Cisco ASR 1000 Series Routers have a local disk where records and event messages (EMs)can be stored on a local cache. Local cache support is a significant advantage because call detail records and EMs are not lost when a billing server is unavailable. Use the **cache** command to configure parameters for storing call detail records and EMs on local disk.

In a typical integrated billing environment, as calls come up and go down, billing records are generated and sent to the RADIUS server. When for any reason the RADIUS server is not reachable or not responding to accounting packets, then the Billing Manager marks the transport as DOWN. As soon as the transport goes down and the local caching path is defined with the **cache path** command, the billing records are cached locally on the Cisco ASR 1000 Series Router disk. Your router disk may be the hard disk, bootflash or usb0, depending on router configuration. Subsequently, every 10 seconds, the Cisco ASR 1000 Series Router tries to send the cached information to the RADIUS server.

Support for Media Information

Cisco Unified Border Element (SP Edition) supports reporting media information in billing messages. The PacketCable event message (EM) billing interface reports the properties of the media streams associated with a call, including when the media stream begins and ends, the packets and octets transmitted, and lost latency and jitter statistics.

The Support for Media Information feature defines a new proprietary RADIUS Vendor-Specific Attribute that can be carried on the QoS_Commit and QoS_Release PacketCable messages. This attribute added to these billing messages makes stream creation information available to PacketCable billing.

Use the cdr media-info command to add the RADIUS Vendor-Specific Attribute to the billing messages.

The RADIUS Vendor-Specific Attribute contains the following information:

- Local IP address and port and remote media endpoint IP address and port used in the media stream.
- Direction of the media stream (send-only, receive-only, send-and-receive, or inactive).
- Codecs negotiated for that media stream.
- Bandwidth reserved for the media stream.

Restrictions for Support for Media Information

The restriction for Support for Media Information are the following:

If an endpoint is behind a NAT, then the endpoint IP address cannot be obtained from the Session Description Protocol (SDP). It is instead auto detected after the endpoint sends media packets. This means that the remote address and port may not be known at the point that the gate is committed. Therefore, this information is not available on the Media_Session_Desc attribute that is sent on the QoS_Commit PacketCable message. Instead, a zero address is specified.

In particular, in a normal call setup and teardown when an endpoint is behind a NAT, there is no remote address or port in the Media_Session_Desc sent on the QoS_Commit message. The correct remote address and port is in the Media_Session_Desc sent in the QoS_Release message.

The only case in which Cisco Unified Border Element (SP Edition) would never report a remote address and port is when the call ends before any media packets have been sent and therefore the remote address is never learned by the media forwarding component on the network processing unit.





Figure 36-1 Integrated Billing Deployment

How to Implement Billing

The SBE can perform billing. The key objects to be configured for billing are the long duration checks and the physical location of the cache. You can configure up to eight PacketCable-EM billing instances (indexed 0-7).

Follow the procedure in the ?\$paranum>Configuring Billing? section on page 36-8.

Restrictions for Billing

The restrictions for configuring billing are:

- You may not modify any billing configuration items if billing is active.
- You may only modify batch-time and batch-size when a method or the billing is active. All other commands are not allowed. However, those are blocked when more than one method exists.
- You may not modify ldr-check at billing level if any methods have been defined.
- You may not remove a RADIUS accounting client if it is currently assigned to a billing method.
- You must define a RADIUS accounting client before it is selected in a billing method.
- You can assign a RADIUS accounting client only to a single billing method.
- You cannot remove the billing when it is active or when methods are configured.
- You may not remove the **method packetcable** command while a packetcable-em configuration is in place.
- H.323 is supported for billing, but with some limitations. One such limitation is that no H.323 signalling address is present in billing instances.

Configuring Billing

This task defines how to configure billing configurations.

SUMMARY STEPS

- 1. configure
- 2. sbc service-name
- 3. sbe
- 4. control address aaa ipv4 IP_address
- 5. radius accounting client-name
- 6. concurrent-requests num
- 7. retry-interval num
- 8. retry-limit num
- 9. server *server-name*
- 10. address ipv4 A.B.C.D.
- 11. priority pri
- **12**. **key** *key*
- 13. port port-num
- 14. exit
- 15. activate
- 16. exit
- 17. billing
- 18. cdr endpoint-info addressing

- **19. ldr-check** {*HH MM*}
- **20. local-address ipv4** {*A*.*B*.*C*.*D*.}
- 21. method packetcable-em
- 22. cache [path {WORD} | alarm [critical VAL] [major VAL] [minor VAL] | max-size {0-4194303}]
- 23. packetcable-em method-index transport radius RADIUS-client-name
- 24. batch-size number
- **25. batch-time** *number*
- 26. attach
- 27. exit
- 28. activate

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
		Use the <i>service-name</i> argument to define the name of the
	Example: Router(config)# sbc mysbc	service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	control address aaa ipv4 IP_address	Configures an SBE to use a given IPv4 AAA control address when contacting an authentication or billing server. This
	Example: Router(config-sbc-sbe)# control address aaa ipv4 192.168.113.2	address is a unique address within the signaling address.
Step 5	radius accounting client-name	Enters the mode for configuring a RADIUS client for accounting purposes.
	Example: Router(config-sbc-sbe)# radius accounting set1	
Step 6	concurrent-requests 0-4000	Sets the maximum number of concurrent requests to the RADIUS server. The default value is 250 and the valid
	Example:	range is between 1 and 4000.
	Router(config-sbc-sbe-acc)# concurrent-requests 34	

	Command or Action	Purpose
Step 7	<pre>retry-interval range Example: Router(config-sbc-sbe-acc)# retry-interval 2000</pre>	Sets the interval for resending an accounting request to the RADIUS server. The default value is 1200 ms and the valid range is between 10 and 10,000 ms.
Step 8	retry-limit range	Sets the retry interval to the RADIUS server. The default value is 5 and the valid range is between 0 and 9.
	Example: Router(config-sbc-sbe-acc)# retry-limit 4	
Step 9	server server-name	Enters the mode for configuring an accounting server within this client.
	<pre>Example: Router(config-sbc-sbe-acc)# server Cisco-AR1-PC</pre>	
Step 10	address ipv4 A.B.C.D	Configures the address of an accounting server.
	Example: Router(config-sbc-sbe-acc-ser)# address ipv4 200.200.200.153	
Step 11	priority pri	Configures the priority of the accounting server. The <i>pri</i> argument must be in the range of 1 to 10 (highest to lowest).
	Example: Router(config-sbc-sbe-acc-ser)# priority 2	
Step 12	key key	Configures the RADIUS authentication key or shared secret to be used for this accounting server.
	Example: Router(config-sbc-sbe-acc-ser)# key cisco	
Step 13	port port-number	Configures the port that the RADIUS server will use to receive Access-Request or Accounting-Request packets. The default port is 1813.
	Example: Router(config-sbc-sbe-acc-ser)# port 2009	
Step 14	exit	Exits the current RADIUS server mode.
	Example: Router(config-sbc-sbe-acc-ser)# exit	Note Repeat Steps 9 to 14 to create multiple RADIUS accounting servers. Only one server is primary; the rest are backup. You would repeat the following commands:
		• server server-name
		• address ipv4 A.B.C.D.
		• priority pri
		• key <i>key</i>
		• port port-num
		• exit

	Command or Action	Purpose
Step 15	activate	Activates the RADIUS server.
	Example: Router/Admin(config-sbc-sbe-acc)# activate	
Step 16	exit	Exits the current RADIUS accounting mode.
	Example: Router/Admin(config-sbc-sbe-acc)# exit	 Note Repeat steps 5 to 16 to create multiple RADIUS accounting clients. You would repeat the following commands: radius accounting <i>client-name</i>
		concurrent-requests num
		• retry-interval num
		• retry-limit num
		• server server-name
		• address ipv4 A.B.C.D.
		• priority <i>pri</i>
		• key <i>key</i>
		• port port-num
		• exit
		• activate
		• exit
Step 17	billing	Configures billing policies.
	Example: Router(config-sbc-sbe)# billing	
Step 18	cdr endpoint-info addressing	Configures billing to include endpoint addressing information.
	Example: Router(config-sbc-sbe-billing)# cdr endpoint-info addressing	
Step 19	ldr-check {HH MM}	Configures the time of day (local time) to run the Long Duration Check (LDR).
	Example: Router(config-sbc-sbe-billing)# ldr-check 22 30	
Step 20	<pre>local-address ipv4 {A.B.C.D.}</pre>	Configures the local IPv4 address that appears in the CDR.
	Example: Router(config-sbc-sbe-billing)# local-address ipv4 10.20.1.1	

	Command or Action	Purpose
Step 21	method packetcable-em	Enables the packet-cable billing method.
Step 22	Example: Router(config-sbc-sbe-billing)# method packetcable-em cache [path {WORD} alarm [critical VAL] [major VAL] [minor VAL] max-size {0-4194303}]	Configures call detail record caching parameters, including alarm levels, maximum cache size, and cache path location.
	Example: Router(config-sbc-sbe-billing)# cache path harddisk:	Note See Tip after the table for configuring the cache path to a hard disk.
Step 23	<pre>packetcable-em method-index transport radius RADIUS-client-name</pre>	Configures a packet-cable billing instance.
	Example: Router(config-sbc-sbe-billing)# packetcable-em 4 transport radius set1	RADIUS-client-name should match the client-name configured with the radius accounting <i>client-name</i> command.
Step 24	batch-size number	Configures the maximum size of a batch when the batch must be set immediately.
	<pre>Example: Router(config-sbc-sbe-billing-packetcable-em)# batch-size 256</pre>	
Step 25	batch-time number	Configures the maximum number of milliseconds for which any record is held before the batch is sent.
	Example: Router(config-sbc-sbe-billing-packetcable-em)# batch-time 22	
Step 26	attach	Activates the billing for a RADIUS client.
	Example: Router(config-sbc-sbe-billing-packetcable-em)# attach	
Step 27	exit	Exits the current mode .
	Example: Router(config-sbc-sbe-billing-packetcable-em)# exit	Note Repeat steps 22 to 25 to create multiple billing method instances. You would repeat the following commands:
		• packetcable-em method-index transport radius RADIUS-client-name
		batch-size number
		 batch-time number attach
Step 28	activate	Activates the Billing Manager.
•		
	Example: Router(config-sbc-sbe-billing)# activate	

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If you choose to set the cache path to hard disk, the cache files are created in the root directory. To prevent cluttering up your root directory, we recommend the following steps:

- 1. Make a directory on the disk to store billing records. For example: mkdir harddisk:billcache
- **2.** Configure the cache path to point to this directory. For example, the following command configures the cache path to point to the directory billcache:

cache path harddisk:/billcache/



The trailing forward slash / is mandatory in the cache path configuration.

Configuration Examples of Implementing Billing

The following example configures billing and enables caching of call detail records and event messages on the designated hard disk:

```
Router# configure terminal
Router(config) # sbc mysbc
Router(config-sbc)# sbe
Router (config-sbc-sbe) # control address aaa ipv4 10.10.10.1 vrf default
Router(config-sbc-sbe) # radius accounting mars
Router(config-sbc-sbe-acc)# concurrent-requests 300
Router(config-sbc-sbe-acc) # retry-interval 1000
Router(config-sbc-sbe-acc)# retry-limit 6
Router(config-sbc-sbe-acc)# server moon
Router(config-sbc-sbe-acc-ser)# address ipv4 10.20.1.1
Router(config-sbc-sbe-acc-ser)# priority 4
Router(config-sbc-sbe-acc-ser)# key test
Router(config-sbc-sbe-acc-ser) # port 1820
Router(config-sbc-sbe-acc-ser) # exit
Router(config-sbc-sbe-acc)# activate
Router(config-sbc-sbe-acc)# exit
Router(config-sbc-sbe) # billing
Router(config-sbc-sbe-billing)# 1dr-check 22 30
Router(config-sbc-sbe-billing)# local-address ipv4 10.20.1.1
Router(config-sbc-sbe-billing)# method packetcable-em
Router(config-sbc-sbe-billing) # cache path harddisk:
Router(config-sbc-sbe-billing) # packetcable-em 3 transport radius test
Router(config-sbc-sbe-billing-packetcable-em)# batch-size 256
Router(config-sbc-sbe-billing-packetcable-em) # batch-time 22
Router(config-sbc-sbe-billing-packetcable-em) # attach
Router(config-sbc-sbe-billing-packetcable-em) # exit
Router(config-sbc-sbe-billing)# activate
```

The following configuration example shows that cache is enabled on the hard disk:

```
sbc asr
sbe
! - Local radius IP address
control address aaa ipv4 10.1.1.1
! - First radius accounting client group
radius accounting ACCT-CLIENT-GROUP-1
! - First radius server
server ACCT-SERVER-1
address ipv4 20.1.1.1
```

L

```
key cisco
activate
! - Billing Manager.
billing
local-address ipv4 10.1.1.1
method packetcable-em
cache path harddisk:
! - First billing method.
packetcable-em 0 transport radius ACCT-CLIENT-GROUP-1
local-address ipv4 10.1.1.1
attach
activate
```

The following configuration example shows that four RADIUS servers have been configured in pairs; the second RADIUS server is backing up server 1, the third RADIUS server is backing up server 4, and both pairs of servers are receiving copies of the same records:

```
sbc asr
sbe
   ! - Local radius IP address
  control address aaa ipv4 10.1.1.1
   ! - First radius accounting client group
  radius accounting ACCT-CLIENT-GROUP-1
   ! - First radius server
   server ACCT-SERVER-1
    address ipv4 20.1.1.1
    key cisco
    ! - Backup for First radius server
    server ACCT-SERVER-2
    address ipv4 20.1.1.2
    key cisco
    activate
   ! - Second radius accounting client group
  radius accounting ACCT-CLIENT-GROUP-2
    ! - Second radius server
   server ACCT-SERVER-3
    address ipv4 30.1.1.1
    key cisco
    ! - Backup for Second radius server
    server ACCT-SERVER-4
    address ipv4 30.1.1.2
    key cisco
   activate
   ! - Billing Manager.
  billing
   local-address ipv4 10.1.1.1
   method packetcable-em
   cache path harddisk:
    ! - First billing method.
   packetcable-em 0 transport radius ACCT-CLIENT-GROUP-1
    local-address ipv4 10.1.1.1
    attach
    ! - Second billing method for duplicate records.
   packetcable-em 1 transport radius ACCT-CLIENT-GROUP-2
    local-address ipv4 10.1.1.1
    attach
    activate
```

The following configuration example shows how to configure endpoint information to capture address information:

```
Router# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# billing
Router(config-sbc-sbe-billing)# cdr endpoint-info addressing
Router(config-sbc-sbe-billing)# end
Router#
```

The following show command output shows that the billing is configured to include the addressing information of the endpoint:

Router# show sbc mySBC sbe billing instance

Billing Manager Information:	
Local IP address:	172.18.53.179
LDR check time:	0:0
Method	packetcable-em
Method	packetcable-li
Admin Status:	DOWN
Operation Status:	DOWN
Cache path:	usb0:billing_cache/
Cache max size:	0 Kilobytes
Cache minor-alarm:	97656 Kilobytes
Cache major-alarm:	488281 Kilobytes
Cache critical-alarm:	976562 Kilobytes
Retry-interval:	20 secs
CDR Media-Info:	Not Included
CDR Endpoint-Info:	Addressing
Billing Methods:	
Radius client name:	SSSS
Instance:	0
Type:	PACKET-CABLE
Transport Mechanism Status:	DOWN
Active Calls Billed:	0
Local IP Address:	172.18.53.179
Deact-mode:	abort

DOWN

DOWN

0:0

1000 ms

0

Support Billing for IP Format

Admin Status: Operation Status:

Batch size:

Batch time:

LDR check time:

Internet is no longer used to transmit only data; it is also used to transmit voice and video. Although the transmission of voice and video through Internet has simplified communication to a large extent, it is very important to understand how voice and video services are being managed and configured.

The PacketCable billing method that is being currently used by the SBC generates call detail record (CDR) in the Bellcore AMA Format (BAF). However, the BAF format is too telephony-specific, and does not contain sufficient provision to support IP-centric logging information. For example, the BAF format does not record session description protocol (SDP) or real-time transport control protocol (RTCP) statistics. Moreover, the PacketCable billing method is not extensible, because of which it is not possible to define extensions to contain these fields.

The XML-based billing method has been selected because it can process IP-centric logging information, It is flexible, and it is commonly used in situations where data must be translated between different platforms, for example, translating billing data from the SBC and the billing server.

Overview of XML-Based Billing

The XML-based billing method is used to generate a set of XML records, each of which gives a complete description of a call. For each call, there is an XML record. In the XM billing method, the billing events are generated and stored in the Billing Manager. Only after the call is complete, the Billing Manager writes the complete CDR on the disk. The XML billing method stores the billing records using the local file daemon. The XML billing records are stored locally in the path configured using the command-line interface (CLI).

When a call begins, the SBC starts recording the billable events pertaining to that call. After the call is completed, the SBC stops recording, and collates the events into a single CDR. The format of the CDR is a proprietary XML format, which can be analyzed and post-processed with standard XML parser tools. The CDR is appended to a local file. Critical, major, and minor alarms for notifying administrator for increase in file-size upon exceeding the configured threshold value is configured using the **cdr alarm** command. This enables the administrator to free up disk space before the disk gets full and the old billing information gets overwritten by the new billing information.

For more information on XML billing schema, see Appendix 1, ?\$paratext>?.

Restrictions for XML-Based Billing

Following are restrictions for XML-based Billing:

- A maximum of only one XML billing instance can be configured.
- Each billing method configuration (under billing) consumes memory. A billing method should not be configured, unless at least one instance of the corresponding method is also configured.
- The **no method xml** command fails if an instance of the corresponding method is configured.
- Compression of the billing records is not supported.
- H.323 is supported for XML billing, but with some limitations. One such limitation is that no H.323 signalling address is present in XML billing instances.

Configuring XML-Based Billing

The following section provides configuration details for the XML billing method, the local path to store the CDR records, threshold values, and for configuring other parameters:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbcname
- 3. sbe
- 4. billing
- 5. method xml

- 6. xml xmlinstance
- 7. cdr path *path*
- 8. ldr-check hour:min
- 9. cdr alarm minor 2000 major 1000 critical 500
- 10. flipped-interval 240
- **11. flipped-size** *20480*
- 12. deact-mode quiesce
- 13. attach
- 14. exit
- 15. activate

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	billing	Configures billing policies.
	Example: Router(config-sbc-sbe)# billing	Note There can be only one instance of Billing Manager per SBC. The Billing Manager must be configured to configure billing.
Step 5	method xml	Enables the XML billing method.
	Example: Router(config-sbc-sbe-billing)# method xml	
Step 6	xml method-instance	Configures an XML billing instance. The range of valid values are 0 to 7.
	Example: Router(config-sbc-sbe-billing)# xml 1	

	Command or Action	Purpose
Step 7	<pre>cdr path path Example: Router(config-sbc-sbe-billing-xml)# cdr path usb0:cdr</pre>	Configures the path to store the CDR billing records. The path must locally point to a directory located either on the flash disk or the hard drive on the Cisco ASR 1000 Series Router.
Step 8	ldr-check hour minutes	Configures the time for checking long duration records. This is the time when all calls over 24-hours-long are reported.
	<pre>Example: Router(config-sbc-sbe-billing-xml)# ldr-check 23 30</pre>	
Step 9	cdr alarm minor 2000 major 1000 critical 500	Configures the alarms to be triggered when free disk space that is lower than the configured size is available.
	Example: Router(config-sbc-sbe-billing-xml)# cdr alarm minor 2000 major 1000 critical 500	
Step 10	flipped-interval 240	Configures the maximum interval (in seconds) to flip the billing XML file. The default value is 3 minutes.
	Example: Router(config-sbc-sbe-billing-xml)# flipped-interval <i>240</i>	
Step 11	flipped-size 20480	Configures the maximum size (in kilobytes) to flip the billing XML file. The default value is 10240 kilo bytes
	Example: Router(config-sbc-sbe-billing-xml)# flipped-size 20480	(KD).
Step 12	deact-mode quiesce	Configures the deactivate mode for the XML billing method.
	Example: Router(config-sbc-sbe-billing-xml)# deact-mode quiesce	
Step 13	attach	Activates the billing instance for XML.
	Example: Router(config-sbc-sbe-billing-xml)# attach	
Step 14	exit	Exits the current mode.
	Example: Router(config-sbc-sbe-billing-xml)# exit	
Step 15	activate	Activates the Billing Manager.
	Example: Router(config-sbc-sbe-billing)# activate	
Retrieving the XML Billing Records

Because the CDR billing records are stored locally on the Cisco ASR 1000 Series Router, it is recommended that the XML billing records are copied to another system regularly. The SBC stores the XML file under the CDR path configured using the CLI. The XML file is flipped after exceeding the fixed size or interval configured. The default file size is 10 MB, and, the default interval is 3 minutes. Copying the billing records from the local disk to remote machine everyday and removing the old billing records from the local disk is therefore recommended. For security reasons, the file should be copied using a secure transport method such as SCP or HTTPS.

Managing Disk Space Through Alarms

The XML billing CDR records are stored on the disk by the file daemon. If there are too many calls in the system can quickly fill a disk. It is therefore important to put an automated management system in place to ensure that sufficient disk space is permanently available. An automated system uses file transfer protocol (FTP) to regularly copy the CDR files to an appropriate server, and deletes the files from the local disk.

If free disk space is lower than what is configured, the SBC generates an alarm, requesting the administrator to free up the disk space by removing the CDRs. The SBC continues to accept calls until more disk space is available. To prevent unbilling of active calls due to lack of disk space, it is recommended that minor, major, and critical alarms to be configured regularly notify the administrator to free up disk space when the free disk space threshold size is exceeded.

Managing the Billing Records During RP Failover

It is important to consider various scenarios that might need attention to retain the billing records. One such scenario is managing the billing records during route processor (RP) failover from active to standby. The SBC billing architecture is designed such that billing records are not lost in case of failover to standby RP. The architecture makes certain assumptions on the infrastructure, and those assumptions should be implemented and verified.

The Billing Manager generates transient billing control block, with billing data. The primary SBC replicates these blocks to the standby SBC. In case of a failover, the call state and billing state are available on the standby, and are designed to continue the call and bill it.

In XMI-based billing, before the failover, the Billing Manager stores the XML billing records in the local disk (via the file daemon interface). When failover occurs, the file daemon flushes the billing records in cache buffer into hard disk. The file daemon writes the records in the local disk belonging to the new active RP.



Note

The CDR path for storing the XML billing records must be defined earlier on the new, active RP. If the CDR path is not defined, the billing records will not be written to the hard disk. If the CDR path is not defined, create it by executing the **cdr path** *path* command from the config-sbc-sbe-billing-xml command mode.

The old billing records that are present on the new standby RP can be copied to a remote machine using the **copy stby-harddisk:** *<destination path>* command.

MD5 Checksum Support for XML Billing Records File

The XML billing records that are stored locally are copied to a remote machine. To ensure that the billing records copied to the destination remote machine are the same as the one existing locally, MD5 checksum support has been implemented on the XML billing records file. A checksum is a form of mechanism that ensures that the file is downloaded properly. The MD5 checksum support is used to provide the XML billing record file integrity check, when the XML billing record is copied from a local storage to a remote server.

When an old XML billing record file is closed, SBC computes and generates the MD5 checksum for the old XML billing record file. The checksum value is stored in the MD5 checksum log file. If size of the log file is more than 2 MB, the MD5 value is switched to another log file to write. There are two log files, md5checksum1.log and md5checksum2.log. The log files are located under the CDR path configured under the SBC SBE billing XML instance.

Selective RADIUS Billing

The billing methods supported by the SBC are:

- XML billing—Billing records are written in a proprietary XML format to disk.
- PacketCable billing—RADIUS messages are sent to RADIUS servers.

Prior to Cisco IOS XE Release 3.3S, all calls have the billing records generated for all the active billing methods. However, the customer that has a RADIUS server of limited capacity cannot generate billing records for calls for a subset of all adjacencies. From Cisco IOS XE Release 3.3S, the Selective RADIUS Billing feature provides the function to select billing methods for calls relating to different adjacencies.

The billing method or methods used for calls can be selected at a per-adjacency scope and the user can also choose to not use the billing method for a specific adjacency.

Configuring Selective RADIUS Billing

This task configures the Selective RADIUS Billing feature.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type policy-set
- 7. entry entry-id
- 8. billing filter {enable | disable}
- 9. billing methods {xml | packetcable-em}
- 10. end
- 11. show sbc sbc-name sbe cac-policy-set id table name entry id

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example:	
	Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters the SBC configuration mode.
	Example:	
	Router(config)# sbc mySBC	
Step 3	sbe	Enters the signaling border element (SBE) function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary:
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —Integer chosen by a user to identify the policy set. The range is from 1 to 2147483647.
Step 5	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table t1	• <i>table-name</i> —Name of the admission control table.
Step 6	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC Policy table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	
Step 7	entry entry-id	Enters the CAC table entry mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	• <i>entry-id</i> —Specifies the table entry.
Step 8	billing filter {enable disable}	Specifies whether the billing filter scheme is enabled or disabled.
	Example:	• enable —Enables the billing filter.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # billing filter enable	• disable —Disables the billing filter.
Step 9	billing methods {xml packetcable-em}	Specifies the billing methods that are allowed for calls relating to different adjacencies.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• packetcable-em —Configures the PacketCable billing method for billing.
	# billing methods xml	• xml —Configures the XML billing method for billing.

	Command or Action	Purpose
Step 10	end	Enables exit from the CAC table entry configuration mode and entry into the Privileged EXEC mode.
	Example:	
	Router# end	
Step 11	show sbc sbc-name sbe cac-policy-set <i>id</i> table name entry <i>id</i>	Lists the detailed information for a given entry in a CAC policy table.
	Example:	
	Router# show sbc mySBC sbe cac-policy-set 1 table t1 entry 1	

The following example displays the partial output of the **show sbc sbe cac-policy-set table entry** command that lists the billing filter information:

Router# show sbc mySBC sbe cac-policy-set 1 table t1 entry 1

```
SBC Service "mySBC"
CAC Averaging period 1: 60 sec
CAC Averaging period 2: 0 sec
CAC Policy Set 1
 Active policy set: No
  Description:
  First CAC table:
 First CAC scope: global
  Table name: t1
   Description:
   Table type: policy-set
   Total call setup failures (due to non-media limits): 0
   Entry 1
   CAC scope:
   CAC scope prefix length: 0
   Action: Not set
   Number of call setup failures (due to non-media limits): 0
media bandwidth policing:
                               Degrade
Media policy limit:
                                mp1
IPsec maximum registers:
                                 10
IPsec maximum calls:
                                  5
Billing filter :
                                  enable
Billing filter methods:
                                  xml
```

Configuration Example of Selective RADIUS Billing

The following example configures all calls billed using XML billing, all calls on an adjacency in the IMS-adjacencies group are configured to be billed using XML and PacketCable-em billing, however, all calls on a special-adj adjacency are configured for not being billed at all.

```
cac-policy-set 1
first-cac-scope global
first-cac-table 1
table-type limit adj-group
cac-table 1
```

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```
entry 1
     action next-table 2
     billing filter enable
     billing methods xml
    !
 !
 cac-table 2
   entry 1
     match-value ims-adjacencies
     action next-table 3
     billing filter enable
     billing methods xml
     billing methods packetcable-em
    !
 !
 cac-table 3
   entry 1
     match-value special-adj
     action cac-complete
     billing filter enable
    1
 !
!
```

Support Billing for IP Format





Billing Support

The following sections describe billing and its many aspects. It is critical to understand all Cisco Unified Border Element (SP Edition) billing features and capabilities before performing billing configurations.

- Integrated Billing Systems, page 37-1
- Event Message Transmission, page 37-3
- Supported Event Message Detail, page 37-6
- Administration and Configuration, page 37-13
- Logging and Alarms, page 37-13
- Fault Tolerance, page 37-14
- Example for Event Messages from Cisco Unified Border Element (SP Edition) to RADIUS Billing Server, page 37-14
- Security, page 37-39

Integrated Billing Systems

Integrated billing is achieved through the PacketCable Event Messages architecture (see the *PacketCable 1.5 Event Messages Specification*; PKT-SP-EM1.5-I01-050128) as exemplified in Figure 37-1 where Cisco Unified Border Element (SP Edition) is integrated into this architecture. As shown, the billing server supports PacketCable Event Messages.

Cisco Unified Border Element (SP Edition) on the Cisco ASR 1000 Series Routers supports remote billing in the unified mode. Remote billing is call billing that is integrated with a third-party accounting server.

Figure 37-1 shows Cisco Unified Border Element (SP Edition) operating in a unified model where the billing system is being deployed with three billing servers. Cisco Unified Border Element (SP Edition) can be configured to send to these servers in a range of ways, such as to all three simultaneously, or to use one primary and two backups.



Figure 37-1 Integrated Billing Deployment

The system operates as follows:

- Cisco Unified Border Element (SP Edition) produces event messages (EMs). These event messages are for billable or other interesting events, such as call start, call end, and media-type changes.
- Cisco Unified Border Element (SP Edition) and other elements of the system, which produces EMs, sends them in real time (or batched up for network efficiency) using the RADIUS protocol to the billing server.



The *PacketCable 1.5 Event Messages Specification* discusses sending the identifying information (the BCID and FEID) on the outgoing INVITE and responding SDP so that correlation can be done between the two sets of billing data. Cisco Unified Border Element (SP Edition) does not support this mechanism for intra-domain or inter-domain transmission. The billing server must perform the correlation using an alternative method (for example, using the telephone numbers dialed and the time of the call).

Event Message Transmission

The generated event messages, as described in the ?**\$paranum>Event Messages Set Overview**? section are sent using the RADIUS protocol to a preconfigured set of billing servers. Before getting into the actual detail of the event messages, review the event message transmission considerations described in the following sections:

- Multiple Server Support
- Event Message Batching
- Event Messages Set Overview

Multiple Server Support

Billing servers are configured at start-up, in SETs:

- Each SET contains a list of one or more billing servers, consisting of a single primary server and an ordered list of zero or more backup servers.
- The SBE can be configured with one or more sets of billing servers.

Each event message is sent to the entire collection of sets, but to only one machine within each set.

- For each set, the SBE sends the event message to the primary server within the set.
- If the primary server is unavailable, the message is sent to the first backup server (if present). If the first backup server is also unavailable, the message is sent to the second backup and so on until either a machine accepts the message or all the servers in the set have been tried.
- If there are no machines in a set accepting messages, the entire set is marked as unavailable.

Figure 37-2 shows the multiple server support.





Event Message Batching

Because of the inefficiency of the RADIUS protocol, the SBE collates event messages into batches and sends them using a single RADIUS message to alleviate the burden on the transport mechanism.

Batching is possible only on a per-set basis. The batch size is not configurable, but is determined by the load on the billing component.

It is not possible to disable batching.

Event Messages Set Overview

This section specifies the set of event messages supported by Cisco Unified Border Element (SP Edition):

- Call-Specific Messages, page 37-5
- Out-of-Band Messages, page 37-5
- Unsupported Messages, page 37-5

Call-Specific Messages

The following table lists supported call event messages.

Event Message	Notes
Signaling_Start	Sent when signaling has begun (inbound) and when it is about to begin (outbound); for example, received INVITE on inbound and about to send INVITE on outbound for a SIP endpoint
QoS_Reserve	Sent when there is reserved QoS in the DBE. Sent for the inbound leg when the inbound QoS is reserved, and for the outbound leg when we reserve the outbound QoS is reserved.
Call_Answer	Indicates that the terminating party has answered and that media has started. This message is sent for both legs at the same time.
QoS_Commit	Sent when QoS is committed by the DBE. This message is sent for both legs at the same time.
Call_Disconnect	Sent when the call has been terminated and the media has ceased flowing. Sent for both legs at the same time.
QoS_Release	Sent when the QoS has been released by the DBE. Sent for both legs at the same time.
Signaling_Stop	Sent after all signaling is complete for each party in the call. (The event is generated once for each party, when the last signaling message has been sent.)
Media_Statistics	Media statistics for the call as reported by the DBE. This is sent for each leg when the media is released.
Media_Alive	Indicates that a long-duration call is still active. This is sent for each leg of the call, at a preconfigured time of day, every 24 hours.

Out-of-Band Messages

The following table lists event messages that are non-call-related, out-of-band event messages.

Event Message	Notes
Time_Change	Sent when changes of more than 200 ms occur in the time; also sent for daylight
	savings changes, and so on.

Unsupported Messages

The following table lists the event messages that are not supported.

Event Message	Notes	Why Not Supported?
Database_Query	Sent when querying external databases about toll-free carriers, LNP routing, and so on.	Cisco Unified Border Element (SP Edition) does not support database queries.

Event Message	Notes	Why Not Supported?	
Service_Instance	Indicates an instance of a service.	Cisco Unified Border Element (SP	
Service_Activation	Indicates service activation.	Edition) does not support services. (Services are more applicable to softswitches and application servers.)	
Service_Deactivation	Indicates service deactivation.		
Interconnect_Start	Sent when interconnecting to PSTN.	Cisco Unified Border Element (SP Edition) does not interface directly	
Interconnect_Stop	Sent when terminating a connection to PSTN.	to the PSTN.	
Conference_Party_Change	Indicates a party state change in a multi-party call.	Cisco Unified Border Element (SP Edition) does not support multi-party calls.	

Supported Event Message Detail

This section specifies the supported event messages and the attributes sent for each one.

Signaling_Start

This message is sent when signaling starts for a call; that is, when Cisco Unified Border Element (SP Edition) has ascertained that the destination is routable and the originating endpoint is allowed to make the call (that is, after the SLA has been checked).

The following	table	lists	the	attributes	sent	with	this	message.
0								0

Attribute Name	Comment
EM_Header	Common header attribute.
Direction_Indicator	Specifies if the device represents an originating or terminating part of the call.
	1= originating
	2 = terminating
MTA_Endpoint_Name	The string <i>MTA Endpoint</i> or the source endpoint information (adjacency name or addressing information).
	The value of this field is either set to <i>MTA Endpoint</i> or to the endpoint information.
	The source adjacency name is used if the SBC is configured to include the adjacency name in the billing records and if the message is from the originating device.
	The source addressing information—in the format <i>IP address,port,transport type, adjacency name</i> —is used if the SBC is configured to include the addressing information in the billing records and if the message is from the originating device.
Calling_Party_Number	The number of the calling party (if available).

Called_Party_Number	The number of the called party (always present).
Routing_Number	Indicates a routable number (always present).
Billing_Type	Included when the originating endpoint is a measured rate subscriber.

The following table lists the attributes not sent with this message.

Attribute Name	Comment
Location_Routing_Number	LNP not supported.
Carrier_Identification_Code	PSTN interfacing not supported (softswitch function).
Trunk_Group_ID	As above.
Intl_Code	Indicates the origin of an international call.
Dial_Around_Code	Carrier specification via dial-around codes not supported.
Jurisdiction_Information_Par ameter	Ported-In billing not supported [transparent to Cisco Unified Border Element (SP Edition)].
Ported_In_Calling_Number	As above.
Ported_In_Called_Number	As above.
Called_Party_NP_source	LNP not supported.
Calling_Party_NP_source	As above.

QoS_Reserve

This message is generated when the SBE has reserved bandwidth (QoS) on the network through the DBE.

If this reserved bandwidth changes, this message (along with the partner QoS_Commit message) is generated anew.

\$ Note

If the SBE is managing multiple gates, this message is generated only for the gates to and from each MTA endpoint (and not the internal gates). There are no optional attributes not sent on this message.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute.
QoS_Descriptor	Description of the QoS reserved (see below).
MTA_UDP_Portnum	The UDP port number on the network element endpoint.
Flow_Direction	1 = upstream
	2 = downstream
SF_ID	This is a required, DOCSIS-specific attribute, generated by the CMTS in a PacketCable architecture. Because Cisco Unified Border Element (SP Edition) does not support DOCSIS, this attribute is always 0.

Call_Answer

This message indicates the earliest point at which non-early two-way media is established.

The SBE sends the message to the billing servers when it is notified that the called party has gone off-hook; that is, that they have answered the call.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute
Charge_Number	The charge number in the appropriate cases such as collect call, calling-card call, call billed to a third party, or others.
	For Cisco Unified Border Element (SP Edition), this number is always the calling number.
MTA_Endpoint_Name	The destination endpoint information—adjacency name or addressing information—is added to the message if the SBC is configured to include the endpoint information in the billing records and if the SBC is the terminating device.
	If the SBC is not configured to include the endpoint information in the message, this attribute is not included.

The following table lists the attributes not sent with this message.

Attribute Name	Comment
Related_Call_Billing_Correla tion_ID	The BCID assigned to the leg from the terminating network element. Cisco Unified Border Element (SP Edition) does not share BCID and FEID information with other network elements.
FEID	Contains the FEID assigned to the network element at the other end of the leg. Cisco Unified Border Element (SP Edition) does not share BCID and FEID information with other network elements.

QoS_Commit

This message is sent by the SBE when the gate bandwidth is committed. This message is only sent if a QoS_Reserve has been previously sent.

The following table lists the attributes sent with this message.

Attribute Name	Comment	
EM_Header	Common header attribute.	
MTA_UDP_Portnum	The UDP port number on the network element endpoint.	
Flow_Direction	1 = upstream	
	2 = downstream	

SF_ID	Always 0 (Cisco Unified Border Element (SP Edition) does not support DOCSIS).
Total_Bandwidth (attribute ID 253)	The total bandwidth in use by the streams described in this QoS_Commit message. See Table B-11 for the structure of this attribute.
Media_Session_Desc (attribute ID 254)	Zero or more attributes describing the media committed in this Flow_Direction. If more than one flow is committed, multiple Media_Session_Desc attributes are differentiated by the Stream_IDs. See Table B-12 for the structure of this attribute.

The following table lists the attributes not sent with this message.

Attribute Name	Comment
QoS_Descriptor	Information is sent on the QoS_Reserve message and not duplicated on this message

The following table lists the structure of the Total_Bandwidth attribute (attribute ID 253).

Attribute Name	Length	Туре	Comment
Total_Bandwidth	8	unsigned	The total bandwidth in use by the streams described in this
(attribute ID 253)		integer	QoS_Commit message.

The following table lists the structure of the Media_Session_Desc attribute (attribute ID 254).

Attribute Name	Length	Туре	Comment
Stream_ID	4	unsigned integer	Unique stream identifier within the scope of the call. A Stream_ID identifies two flows, one upstream and one downstream - this Media_Info attribute is for the flow identified by the Flow_Direction attribute on this message.
Local_address_type	1	Byte	1 = IPv4 address—of length 4.2 = IPv6 address—of length 16.
Local_address	variable	byte array	The local address - length defined by the Local_address_type.
Local_port	2	unsigned integer	The local port.
Remote_address_type	1	Byte	1 = IPv4 address—of length 4.2 = IPv6 address—of length 16.
Remote_address	variable	byte array	The remote address—length defined by the Remote_address_type.
Remote_port	2	unsigned integer	The remote port.

Transrated	1	Byte	0—Transrating is not in use on this stream.
			1—Transrating is in use on this stream.
Truncated	1	Byte	0—The following SDP is not truncated.
			1—The following SDP was truncated after the last complete a= line to prevent the RADIUS attribute from exceeding 247 bytes.
SDP_fragment_len	1	Byte	Length of the following SDP—can be equal to 0.
SDP_fragment	variable	ASCII character string	Optionally an SDP fragment describing this flow. The port numbers present in this fragment should be ignored, and the values above used instead.

Call_Disconnect

This message is generated by the SBE when 2-way media flow terminates—when sending a 200 OK response to a BYE from either party.

Usually, this message immediately precedes QoS_Release and Signaling_Stop.

This message is only sent if a Call_Answer has previously been sent.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute.
Call_Termination_Cause	Reason for termination of the call.

There are no optional attributes not sent for this message.

QoS_Release

This message is generated by the SBE when the reserved bandwidth has been released; that is, the gate on the DBE has been closed.

The following table lists the attributes sent with this message.

Attribute Name	Comment	
EM_Header	Common header attribute.	
Flow_Direction	1 = upstream.	
	2 = downstream.	

SF_ID	A DOCSIS specific attribute, service flow ID, generated by the CMTS in a PacketCable architecture. Cisco Unified Border Element (SP Edition) does not support DOCSIS. Therefore this attribute is always set to 0.
Media_Session_Desc (attribute ID 254)	Zero or more attributes describing the media committed in this Flow_Direction. If more than one flow is committed, multiple Media_Session_Desc attributes are differentiated by the Stream_IDs. See Table B-12 for the structure of this attribute.

There are no optional attributes not sent for this message.

Signaling_Stop

This message is sent when:

- The terminating signaling request (for example, a SIP BYE) from the party terminating the call is acknowledged by the SBE
- The terminating signaling request for the party not terminating the call is sent by the SBE and acknowledged by that party.

This message is not sent if we have not sent a Signaling_Start for this call.

The following table lists the attributes sent with this message.

Attribute Name	Comment	
EM_Header	The header attribute (must be first).	
Related_Call_Billing_Correla tion_ID	The BCID of the other leg (that is, if this is the caller, then the callee, and vice-versa).	
Call_Termination_Cause	The reason the call was terminated.	
MTA_Endpoint_Name	If the SBC is configured to include the endpoint information—adjacency name or addressing information—in the message, this attribute is added. The destination endpoint information is added to the terminating device message and the source endpoint information is added to the originating device message. If the SBC is not configured to include the endpoint information in the message this attribute is not included	

The following table lists the attributes not sent with this message.

Attribute Name	Comment
FEID	The FEID of the terminating network element. Cisco Unified Border Element (SP Edition) does not transmit this between network elements.

Media_Statistics

When a call is terminated on the DBE (that is, the gate is closed), statistics are returned to the SBE. On receipt of these statistics, this message is generated.

When media QoS is renegotiated, the gate is closed and re-opened. In this case, statistics are logged for the first gate when it closes, and for the second gate when it closes (at the end of the call).

There may be multiple gates for each Media. The statistics are aggregated and result in only one Media_Statistics message per billing leg.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute.
RTCP_Data	The report data from the DBE on the gate statistics.

There are no optional attributes not sent for this message.

Media_Alive

This message is generated once a day, at a pre-configured time.

At the preconfigured time, the SBE audits the active calls, and determines which calls (if any) have been active for more than 24 hours. For each call satisfying this condition, a Media_Alive message is generated.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute.

There are no optional attributes not sent for this message.

Time_Change

This message is generated by the SBE either on its own behalf, or on the behalf of the DBE, when either the DBE or SBE experiences a time change of more than 200 ms (discounting slew adjustments via Network Time Protocol (NTP)). This includes step adjustments, manual time settings changes and daylight savings time adjustments.

The following table lists the attributes sent with this message.

Attribute Name	Comment
EM_Header	Common header attribute.
Time_Adjustment	Adjustment in milliseconds.

There are no optional attributes not sent for this message.

Administration and Configuration

Billing requires the following generic configuration:

Integrated Mode Configuration

If integrated mode is specified, then the following configuration information is required:

- The assigned element ID. This is an ID assigned by the Internet service provider (ISP). The ID must be unique across the set of SBEs, sending event messages to a particular set of billing servers.
- The minor, major, and critical threshold sizes for the event message cache file.
- The location of the event message cache file on disk.
- The time at which to generate the Media_Alive message.
- RADIUS client configuration information.

Integrated mode requires the RADIUS client component of Cisco Unified Border Element (SP Edition). This has configuration requirements (such as the sets of billing servers). Each of these sets also has a state, which depends on the existence or absence of the event message cache file for that set. The administrator may change this state. The state may be disabled, active, failed, or resending.

Administering Cisco Unified Border Element (SP Edition) Billing

The billing component is administered using the Cisco Unified Border Element (SP Edition) command-line interface. See the applicable billing commands in *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at: http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

Logging and Alarms

Alarms are tripped differently, based on how billing has been integrated, as described in the following table .

Billing System Type	Logging Conditions
Integrated Billing Alarms	Alarms are tripped under the following conditions:
	• Minor, major, and critical alarms are sent if the cache file size exceeds a preconfigured threshold.
	• Alarms are tripped when billing servers become unavailable, as follows.
	- A minor alarm is tripped if just one of the configured sets of billing servers is unavailable.
	- A major alarm is tripped if more than one of the billing server sets is unavailable.
	- A critical alarm is tripped if none of the billing servers is available.
	Note In this situation, it may be that the condition for more than one alarm is satisfied (for example, there is just one server set configured, which fails). The most severe alarm dominates.

Fault Tolerance

The Cisco Unified Border Element (SP Edition) billing system is fault tolerant on the following two levels:

- Warm Failover—Failover to a live backup (for example, a second card on the same machine).
- Cold Failover—Failover to a new machine with no software connection between the defunct machine and the new machine.

Warm Failover

During a failover to a backup system, warm failover mechanisms are supported. In the case of warm failover:

- No data is lost on the SBE.
- The value for media statistics for the call on the DBE is reset (this information is lost).

Cold Failover

During the failover to a cold, non-dedicated backup, some billing data is lost in the transition from the old, failed system to the new server. The number of billing records completely lost during this transition is less than 10,000 per failover. However, in such a situation, consider the following possibilities:

- The remaining billing records may be corrupted, and only partial billing records recovered. This is especially true with local CDR generation, because no logs are produced in a hard format until the call ends.
- If an event message cache exists on the failed machine, more billing events may be lost, because the disk record may be unrecoverable because of fire, hardware malfunction, or whatever the original cause of the total failure was. This, however, is an unlikely scenario, because it would require the billing server to be unavailable and unrecovered for a period preceding the cold failover.
- If the media to which the CDRs are written is lost, the entire store of CDRs not backed up (by extracting the records using FTP) is lost.
- It is not possible to detect long-duration calls following a cold failover. Data is only recoverable from the system only when an event occurs in the network, such as the media being terminated).

Example for Event Messages from Cisco Unified Border Element (SP Edition) to RADIUS Billing Server

This section contains the following examples:

- Example 1 shows two requests from the SBC to the RADIUS server for a single placed call.
- Example 2 shows requests from the SBC to RADIUS server where the SBC is configured to include the endpoint adjacency name in billing records.
- Example 3 shows requests from the SBC to RADIUS server where the SBC is configured to include the endpoint addressing information in billing records.

Example 1

This example shows two requests from the SBC to the RADIUS server for a single placed call. The first RADIUS event message has messages related to call setup:

• Event Message Type: Signaling_Start

- Event Message Type: QoS_Reserve
- Event Message Type: Call_Answer
- Event Message Type: QoS_Commit

The second RADIUS event message has messages related to call teardown:

- Event Message Type: QoS_Release
- Event Message Type: Call_Disconnect
- Event Message Type: Signaling_Stop

```
Radius Protocol
   Code: Accounting-Request (4)
   Packet identifier: 0x0 (0)
   Length: 1298
   Authenticator: 25CE1B487AE4AE70033D61E0EF540A4A
   [The response to this request is in frame 4]
   Attribute Value Pairs
       AVP: 1=6 t=NAS-IP-Address(4): 77.111.1.51
           NAS-IP-Address: 77.111.1.51 (77.111.1.51)
       AVP: 1=6 t=Acct-Status-Type(40): Interim-Update(3)
           Acct-Status-Type: Interim-Update (3)
       AVP: 1=26 t=Acct-Session-Id(44): HDq]
                                                  01+000000\000\000\000
                                     01+000000
           Acct-Session-Id: HDq]
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=78 t=CableLabs-Event-Message(1):
               Event Message Version ID: 4
               BCTD
                  Timestamp: 1212445021
                  Element ID:
                                    0
                  Time Zone: DST: 1, Offset: +000000
                  Event Counter: 1
               Event Message Type: Signaling_Start (1)
               Element Type: CMS (1)
               Element ID:
                                 0
               Time Zone: DST: 1, Offset: +000000
               Sequence Number: 0
               Event Time: 20080602221700.000
               Status: 0x0000008
                   (0x0000000)
                   .... .... .... .... .... .0.. = Event Origin: Trusted
Element (0x0000000)
                   ..... .... .... .... .... 1.... = Event Message Proxied:
Proxied (0x0000001)
               Priority: 128
               Attribute Count: 6
               Event Object: 0
       AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=4 t=CableLabs-Direction-indicator(37): Originating(1)
               CableLabs-Direction-indicator: Originating (1)
       AVP: 1=20 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=14 t=CableLabs-MTA-Endpoint-Name(3): MTA Endpoint
               CableLabs-MTA-Endpoint-Name: MTA Endpoint
```

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```
AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Calling-Party-Number(4):
                                                                         123
               CableLabs-Calling-Party-Number:
                                                              123
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Called-Party-Number(5):
                                                                    service
               CableLabs-Called-Party-Number:
                                                          service
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Routing-Number(25):
                                                                service
               CableLabs-Routing-Number:
                                                     service
       AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=4 t=CableLabs-Billing-Type(87): 3
               CableLabs-Billing-Type: 3
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=78 t=CableLabs-Event-Message(1):
               Event Message Version ID: 4
               BCID
                   Timestamp: 1212445021
                   Element ID:
                                     0
                   Time Zone: DST: 1, Offset: +000000
                   Event Counter: 2
               Event Message Type: Signaling_Start (1)
               Element Type: CMS (1)
               Element ID:
                                 0
               Time Zone: DST: 1, Offset: +000000
               Sequence Number: 1
               Event Time: 20080602221700.000
               Status: 0x0000008
                   (0x0000000)
                   .... 0.. = Event Origin: Trusted
Element (0x0000000)
                   ..... .... .... ..... 1.... = Event Message Proxied:
Proxied (0x0000001)
               Priority: 128
               Attribute Count: 6
               Event Object: 0
       AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=4 t=CableLabs-Direction-indicator(37): Terminating(2)
               CableLabs-Direction-indicator: Terminating (2)
       AVP: 1=20 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=14 t=CableLabs-MTA-Endpoint-Name(3): MTA Endpoint
               CableLabs-MTA-Endpoint-Name: MTA Endpoint
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Calling-Party-Number(4):
                                                                         123
               CableLabs-Calling-Party-Number:
                                                              123
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Called-Party-Number(5):
                                                                    service
               CableLabs-Called-Party-Number:
                                                          service
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Routing-Number(25):
                                                                service
               CableLabs-Routing-Number:
                                                     service
       AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=4 t=CableLabs-Billing-Type(87): 3
               CableLabs-Billing-Type: 3
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=78 t=CableLabs-Event-Message(1):
               Event Message Version ID: 4
               BCID
                   Timestamp: 1212445021
                   Element ID:
                                     0
                   Time Zone: DST: 1, Offset: +000000
                   Event Counter: 1
               Event Message Type: QoS_Reserve (7)
               Element Type: CMS (1)
```

Element ID: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 2 Event Time: 20080602221700.000 Status: 0x0000008 $(0 \times 0 0 0 0 0 0 0 0)$0.. = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 4 Event Object: 0 AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=26 t=CableLabs-QoS-Descriptor(32): OoS Status: 0x00000005 Reserved but not Activated (1)1.. = Service Flow Scheduling Type: 1 Grant Interval: 0 Tolerated Grant Jitter: 00... = Unsolicited Grant Size: 0 = Traffic Priority: 0 Sustained Rate: 0 = Maximum Traffic Burst: 00... = Minimum Reserved Traffic Rate: 0 = Minium Packet Size: 0 = Maximum Concatenated Burst: 0 = Status Request/Transmission Policy: 00.. = Nominal Polling Interval: 0 0... 0... = Tolerated Poll Jitter: 0 = Type of Service Override: 00. = Maximum Downstream Latency: 0 Service Class Name: Service Flow Scheduling Type: 1 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-MTA-UDP-Portnum(26): 0 CableLabs-MTA-UDP-Portnum: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-SF-ID(30): 0 CableLabs-SF-ID: 0 AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=4 t=CableLabs-Flow-Direction(50): Upstream(1) CableLabs-Flow-Direction: Upstream (1) AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=78 t=CableLabs-Event-Message(1): Event Message Version ID: 4 BCID Timestamp: 1212445021 Element ID: Time Zone: DST: 1, Offset: +000000 Event Counter: 2 Event Message Type: QoS_Reserve (7) Element Type: CMS (1) Element ID: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 3

Event Time: 20080602221700.000 Status: 0x0000008 (0x0000000)....0.. = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 4 Event Object: 0 AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=26 t=CableLabs-QoS-Descriptor(32): QoS Status: 0x0000005 Reserved but not Activated (1)1.. = Service Flow Scheduling Type: 1 0... = Grant Interval: 0 = Grants Per Interval: 00.. = Unsolicited Grant Size: 0 = Traffic Priority: 0 Sustained Rate: 0 Traffic Burst: 00.. = Minimum Reserved Traffic Rate: 0 = Minium Packet Size: 0 = Maximum Concatenated Burst: 00. = Status Request/Transmission Policy: 00.. = Nominal Polling Interval: 0 0... 0... = Tolerated Poll Jitter: 0 = Type of Service Override: 0 0 Service Class Name: Service Flow Scheduling Type: 1 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-MTA-UDP-Portnum(26): 0 CableLabs-MTA-UDP-Portnum: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-SF-ID(30): 0 CableLabs-SF-ID: 0 AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=4 t=CableLabs-Flow-Direction(50): Downstream(2) CableLabs-Flow-Direction: Downstream (2) AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=78 t=CableLabs-Event-Message(1): Event Message Version ID: 4 BCID Timestamp: 1212445021 Element ID: 0 Time Zone: DST: 1, Offset: +000000 Event Counter: 1 Event Message Type: Call_Answer (15) Element Type: CMS (1) Element ID: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 4 Event Time: 20080602221701.000 Status: 0x0000008

```
(0x0000000)
                      .... .... .... .... .0.. = Event Origin: Trusted
Element (0x0000000)
                      .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 2
              Event Object: 0
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=22 t=CableLabs-Charge-Number(16):
                                                               123
              CableLabs-Charge-Number:
                                                    123
       AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=26 t=CableLabs-Related-Call-Billing-Correlation-ID(13):
              Timestamp: 1212445021
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Event Counter: 2
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCID
                  Timestamp: 1212445021
                  Element ID:
                  Time Zone: DST: 1, Offset: +000000
                  Event Counter: 2
              Event Message Type: Call_Answer (15)
              Element Type: CMS (1)
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 5
              Event Time: 20080602221701.000
              Status: 0x0000008
                  (0x0000000)
                  ..... .... .... .... .... .... .0... = Event Origin: Trusted
Element (0x0000000)
                  .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 2
              Event Object: 0
       AVP: 1=28 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=22 t=CableLabs-Charge-Number(16):
                                                            service
              CableLabs-Charge-Number:
                                                 service
       AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=26 t=CableLabs-Related-Call-Billing-Correlation-ID(13):
              Timestamp: 1212445021
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Event Counter: 1
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCID
                  Timestamp: 1212445021
                  Element ID:
                  Time Zone: DST: 1, Offset: +000000
                  Event Counter: 1
              Event Message Type: QoS_Commit (19)
              Element Type: CMS (1)
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 6
```

Event Time: 20080602221701.000 Status: 0x0000008 (0x0000000).... 0... = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 3 Event Object: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-MTA-UDP-Portnum(26): 0 CableLabs-MTA-UDP-Portnum: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-SF-ID(30): 0 CableLabs-SF-ID: 0 AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=4 t=CableLabs-Flow-Direction(50): Upstream(1) CableLabs-Flow-Direction: Upstream (1) AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=78 t=CableLabs-Event-Message(1): Event Message Version ID: 4 BCID Timestamp: 1212445021 Element ID: 0 Time Zone: DST: 1, Offset: +000000 Event Counter: 2 Event Message Type: QoS_Commit (19) Element Type: CMS (1) Element ID: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 7 Event Time: 20080602221701.000 Status: 0x0000008 (0x0000000).... 0... = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 3 Event Object: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-MTA-UDP-Portnum(26): 0 CableLabs-MTA-UDP-Portnum: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-SF-ID(30): 0 CableLabs-SF-ID: 0 AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=4 t=CableLabs-Flow-Direction(50): Downstream(2) CableLabs-Flow-Direction: Downstream (2) _____

Radius Protocol Code: Accounting-Response (5) Packet identifier: 0x0 (0)

Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model

```
Length: 20
   Authenticator: EB0BD7E187D3301CB7D73349761F9DE0
   [This is a response to a request in frame 1]
   [Time from request: 0.041131000 seconds]
      Time
                 Source
                                    Destination
                                                       Protocol Info
No.
     5 29.324537 77.111.1.51
                                    200.200.1.2
                                                       RADTUS
Accounting-Request(4) (id=0, 1=1162)
_____
_____
Radius Protocol
   Code: Accounting-Request (4)
   Packet identifier: 0x0 (0)
   Length: 1162
   Authenticator: 78D7DE7EA0162046A7936593F80048D5
   [The response to this request is in frame 6]
   Attribute Value Pairs
       AVP: 1=6 t=NAS-IP-Address(4): 77.111.1.51
          NAS-IP-Address: 77.111.1.51 (77.111.1.51)
       AVP: 1=6 t=Acct-Status-Type(40): Interim-Update(3)
          Acct-Status-Type: Interim-Update (3)
       AVP: 1=26 t=Acct-Session-Id(44): HDq]
                                              01+000000\000\000\000\001
          Acct-Session-Id: HDq]
                                 01+000000
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCID
                 Timestamp: 1212445021
                 Element TD:
                               0
                 Time Zone: DST: 1, Offset: +000000
                 Event Counter: 1
              Event Message Type: Unknown (22)
              Element Type: CMS (1)
              Element ID:
                              0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 8
              Event Time: 20080602221731.000
              Status: 0x0000008
                 (0x0000000)
                 .... .... .... .... .... .0... = Event Origin: Trusted
Element (0x0000000)
                 .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 1
              Event Object: 0
       AVP: 1=134 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=128 t=CableLabs-RTCP-Data(93): PS=0, OS=0, PR=0, OR=0, PD=0, OD=0,
PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0,
00\000\000\000
             CableLabs-RTCP-Data: PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0,
PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0, PC/RJI=0
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCID
                 Timestamp: 1212445021
                 Element ID:
                                  0
                 Time Zone: DST: 1, Offset: +000000
                 Event Counter: 2
```

Event Message Type: Unknown (22) Element Type: CMS (1) Element TD: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 9 Event Time: 20080602221731.000 Status: 0x0000008 (0x0000000)....0.. = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 1 Event Object: 0 AVP: 1=134 t=Vendor-Specific(26) v=CableLabs(4491) VSA: l=128 t=CableLabs-RTCP-Data(93): PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0, 00\000\000\000 CableLabs-RTCP-Data: PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0, PC/RJI=0 AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=78 t=CableLabs-Event-Message(1): Event Message Version ID: 4 BCID Timestamp: 1212445021 Element ID: 0 Time Zone: DST: 1, Offset: +000000 Event Counter: 1 Event Message Type: QoS_Release (8) Element Type: CMS (1) Element ID: 0 Time Zone: DST: 1, Offset: +000000 Sequence Number: 10 Event Time: 20080602221731.000 Status: 0x0000008 $(0 \times 0 0 0 0 0 0 0 0)$0.. = Event Origin: Trusted Element (0x0000000) 1... = Event Message Proxied: Proxied (0x0000001) Priority: 128 Attribute Count: 2 Event Object: 0 AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=6 t=CableLabs-SF-ID(30): 0 CableLabs-SF-ID: 0 AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=4 t=CableLabs-Flow-Direction(50): Upstream(1) CableLabs-Flow-Direction: Upstream (1) AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491) VSA: 1=78 t=CableLabs-Event-Message(1): Event Message Version ID: 4 BCID Timestamp: 1212445021 Element ID: 0 Time Zone: DST: 1, Offset: +000000 Event Counter: 2 Event Message Type: QoS_Release (8) Element Type: CMS (1) Element TD: 0

```
Time Zone: DST: 1, Offset: +000000
              Sequence Number: 11
              Event Time: 20080602221731.000
              Status: 0x0000008
                 (0x0000000)
                 .... .... .... .... .... .0.. = Event Origin: Trusted
Element (0x0000000)
                     .... Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 2
              Event Object: 0
       AVP: 1=12 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=6 t=CableLabs-SF-ID(30): 0
             CableLabs-SF-ID: 0
       AVP: 1=10 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=4 t=CableLabs-Flow-Direction(50): Downstream(2)
             CableLabs-Flow-Direction: Downstream (2)
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCID
                 Timestamp: 1212445021
                 Element ID: 0
                 Time Zone: DST: 1, Offset: +000000
                 Event Counter: 1
              Event Message Type: Call_Disconnect (16)
              Element Type: CMS (1)
              Element ID:
                              0
             Time Zone: DST: 1, Offset: +000000
              Sequence Number: 12
              Event Time: 20080602221731.000
              Status: 0x0000008
                 (0x0000000)
                 .... .... .... .... .... .0... = Event Origin: Trusted
Element (0x0000000)
                 .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
             Attribute Count: 1
              Event Object: 0
       AVP: 1=14 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=8 t=CableLabs-Call-Termination-Cause(11):
              Source Document: BAF (0x0001)
              Event Object: 16
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCTD
                 Timestamp: 1212445021
                 Element ID:
                                  0
                 Time Zone: DST: 1, Offset: +000000
                 Event Counter: 1
              Event Message Type: Signaling_Stop (2)
              Element Type: CMS (1)
              Element ID:
                              0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 13
              Event Time: 20080602221731.000
              Status: 0x0000008
                 (0 \times 00000000)
```

```
.... 0... = Event Origin: Trusted
Element (0x0000000)
                      .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 2
              Event Object: 0
       AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=26 t=CableLabs-Related-Call-Billing-Correlation-ID(13):
              Timestamp: 1212445021
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Event Counter: 2
       AVP: 1=14 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=8 t=CableLabs-Call-Termination-Cause(11):
              Source Document: BAF (0x0001)
              Event Object: 16
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCTD
                  Timestamp: 1212445021
                  Element ID:
                                  0
                  Time Zone: DST: 1, Offset: +000000
                  Event Counter: 2
              Event Message Type: Call_Disconnect (16)
              Element Type: CMS (1)
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 14
              Event Time: 20080602221731.000
              Status: 0x0000008
                  .... .... .... .... .... .... ...00 = Status: No Error
(0x0000000)
                  .... 0.. = Event Origin: Trusted
Element (0x0000000)
                  .... .... .... .... .... 1... = Event Message Proxied:
Proxied (0x0000001)
              Priority: 128
              Attribute Count: 1
              Event Object: 0
       AVP: 1=14 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=8 t=CableLabs-Call-Termination-Cause(11):
              Source Document: BAF (0x0001)
              Event Object: 16
       AVP: 1=84 t=Vendor-Specific(26) v=CableLabs(4491)
           VSA: 1=78 t=CableLabs-Event-Message(1):
              Event Message Version ID: 4
              BCTD
                  Timestamp: 1212445021
                  Element ID:
                                  0
                  Time Zone: DST: 1, Offset: +000000
                  Event Counter: 2
              Event Message Type: Signaling_Stop (2)
              Element Type: CMS (1)
              Element ID:
                               0
              Time Zone: DST: 1, Offset: +000000
              Sequence Number: 15
              Event Time: 20080602221731.000
              Status: 0x0000008
                  (0x0000000)
                  .... 0.. = Event Origin: Trusted
Element (0x0000000)
```

Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model

```
..... Event Message Proxied:
                 . . . .
Proxied (0x0000001)
             Priority: 128
             Attribute Count: 2
             Event Object: 0
      AVP: 1=32 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=26 t=CableLabs-Related-Call-Billing-Correlation-ID(13):
             Timestamp: 1212445021
             Element ID:
                             0
             Time Zone: DST: 1, Offset: +000000
             Event Counter: 1
      AVP: 1=14 t=Vendor-Specific(26) v=CableLabs(4491)
          VSA: 1=8 t=CableLabs-Call-Termination-Cause(11):
             Source Document: BAF (0x0001)
             Event Object: 16
_____
Radius Protocol
   Code: Accounting-Response (5)
   Packet identifier: 0x0 (0)
   Length: 20
   Authenticator: 663449DAB02BF4CC5480672195DFFFE0
   [This is a response to a request in frame 5]
   [Time from request: 0.063580000 seconds]
```

Example 2

The following example shows the requests from the SBC to RADIUS server where SBC is configured to include the endpoint adjacency name in billing records:

```
Mon May 24 10:43:02 2010
    NAS-IP-Address = 172.18.53.179
    Acct-Status-Type = Interim-Update
    Acct-Session-Id = "K \setminus 372Ds
                                 326311+030000\000\000\000\t"
    CableLabs-Event-Message =
0x00044bfa44732020203332363331312b303330303030000000000000100012020203332363331312b30333030
30300000004032303130303532343132313834332e3430370000000880000600
       Version_{ID} = 4
       Timestamp = 1274692723
       Element_ID =
                        32631
       Time_Zone = 1+030000
       Event_Counter = 9
       Event_Message_Type = Signaling-Start
       Element_Type = 1
       Element_ID = 32631
       Time_Zone = 1+030000
       Sequence_Number = 64
       Event_Time = 20100524121843.407
       Status = 8
       Priority = 128
       Attribute_Count = 6
       Event_Object = 0
    CableLabs-Direction-indicator = 0x0001
       Originating
    CableLabs-MTA-Endpoint-Name = "SIPPB"
    CableLabs-Calling-Party-Number = "
                                                       sipp"
    CableLabs-Called-Party-Number = "
                                                   service"
    CableLabs-Routing-Number = "
                                              service"
    CableLabs-Attr-87 = 0 \times 0003
```

```
Billing type -
       flat rate
   CableLabs-Event-Message =
3030000004132303130303532343132313834332e3430370000000880000600
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event_Counter = 10
      Event_Message_Type = Signaling-Start
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence_Number = 65
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 6
      Event_Object = 0
   CableLabs-Direction-indicator = 0x0002
      Terminating
   CableLabs-MTA-Endpoint-Name = "MTA Endpoint"
   CableLabs-Calling-Party-Number = "
                                              sipp"
   CableLabs-Called-Party-Number = "
                                           service"
   CableLabs-Routing-Number = "
                                      service"
   CableLabs-Attr-87 = 0x0003
      Billing type -
       flat rate
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = K\372Ds
                           326311+030000\000\000\000\t"
   CableLabs-Event-Message =
3030000004232303130303532343132313834332e3430370000000880000400
      Version TD = 4
      Timestamp = 1274692723
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event\_Counter = 9
      Event_Message_Type = QoS-Reserve
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence Number = 66
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute Count = 4
      Event_Object = 0
   Status_Bitmask = 5
      Service_Class_Name =
      QoS_Parameter_Array = 1
         resource reserved but not committed
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0001
      Upstream
```

```
CableLabs-Event-Message =
3030000004332303130303532343132313834332e3430370000000880000400
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                   32631
      Time_Zone = 1+030000
      Event_Counter = 10
      Event_Message_Type = QoS-Reserve
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence_Number = 67
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 4
      Event Object = 0
   Status_Bitmask = 5
      Service_Class_Name =
      QoS_Parameter_Array = 1
         resource reserved but not committed
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0002
      Downstream
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 372Ds
                          326311+030000\000\000\t"
   CableLabs-Event-Message =
3030000004432303130303532343132313834332e3430370000000880000200
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                  32631
      Time_Zone = 1+030000
      Event_Counter = 9
      Event_Message_Type = Call-Answer
      Element_Type = 1
      Element_ID =
                 32631
      Time_Zone = 1+030000
      Sequence_Number = 68
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 2
      Event_Object = 0
   CableLabs-Charge-Number = "
                                      sipp"
   CableLabs-Related-Call-Billing-Crl-ID =
0x4bfa44732020203332363331312b303330303030000000a
      Timestamp = 1274692723
      Element_ID =
                  32631
      Time_Zone = 1+030000
      Event Counter = 10
   CableLabs-Event-Message =
3030000004532303130303532343132313834332e343037000000880000300
      Version_{ID} = 4
```

```
Timestamp = 1274692723
       Element_ID = 32631
       Time_Zone = 1+030000
       Event_Counter = 10
       Event_Message_Type = Call-Answer
       Element_Type = 1
       Element_ID = 32631
       Time_Zone = 1+030000
       Sequence_Number = 69
       Event_Time = 20100524121843.407
       Status = 8
       Priority = 128
       Attribute_Count = 3
       Event_Object = 0
   CableLabs-Charge-Number = "
                                            service"
   CableLabs-Related-Call-Billing-Crl-ID =
0x4bfa44732020203332363331312b3033303030300000009
       Timestamp = 1274692723
       Element_ID =
                       32631
       Time_Zone = 1+030000
       Event Counter = 9
   CableLabs-MTA-Endpoint-Name = "SIPPA"
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 372Ds
                               326311+030000\000\000\t"
   CableLabs-Event-Message =
0x00044bfa44732020203332363331312b303330303000000009001300012020203332363331312b30333030
3030000004632303130303532343132313834332e3430370000000880000300
       Version_{ID} = 4
       Timestamp = 1274692723
       Element_ID =
                       32631
       Time_Zone = 1+030000
       Event_Counter = 9
       Event_Message_Type = QoS-Commit
       Element_Type = 1
       Element_ID =
                     32631
       Time_Zone = 1+030000
       Sequence_Number = 70
       Event_Time = 20100524121843.407
       Status = 8
       Priority = 128
       Attribute_Count = 3
       Event_Object = 0
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-TD = 0
   CableLabs-Flow-Direction = 0x0001
       Upstream
   CableLabs-Event-Message =
0x00044bfa44732020203332363331312b30333030300000000a001300012020203332363331312b30333030
30300000004732303130303532343132313834332e3430370000000880000300
       Version_{ID} = 4
       Timestamp = 1274692723
       Element_ID = 32631
       Time_Zone = 1+030000
       Event Counter = 10
       Event_Message_Type = QoS-Commit
       Element_Type = 1
       Element_ID =
                       32631
       Time_Zone = 1+030000
```

```
Sequence_Number = 71
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 3
      Event_Object = 0
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0002
      Downstream
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 372Ds
                           326311+030000\000\000\000\t"
   CableLabs-Event-Message =
3030000004832303130303532343132313834332e3430370000000880000100
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event\_Counter = 9
      Event_Message_Type = Media-Statistics
      Element_Type = 1
      Element_ID =
                  32631
      Time_Zone = 1+030000
      Sequence Number = 72
      Event Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 1
      Event_Object = 0
   CableLabs-Attr-93 =
3d302c204c413d302c2050432f5250533d302c2050432f524f533d302c2050432f5250523d302c2050432f5250
4c3d302c2050432f524a493d30
      RTCP Data:
      PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0,
PC/RPR=0, PC/RPL=0, PC/RJI=0
   CableLabs-Event-Message =
30300000004932303130303532343132313834332e3430370000000880000100
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID = 32631
      Time_Zone = 1+030000
      Event_Counter = 10
      Event_Message_Type = Media-Statistics
      Element_Type = 1
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Sequence_Number = 73
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute Count = 1
      Event_Object = 0
```

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```
CableLabs-Attr-93 =
3d302c204c413d302c2050432f5250533d302c2050432f524f533d302c2050432f5250523d302c2050432f5250
4c3d302c2050432f524a493d30
      RTCP Data:
      PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0,
PC/RPR=0, PC/RPL=0, PC/RJI=0
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 372Ds
                             326311+030000\000\000\000\t"
   CableLabs-Event-Message =
0x00044bfa44732020203332363331312b3033303030000000009000800012020203332363331312b30333030
30300000004a32303130303532343132313834332e3430370000000880000200
       Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event_Counter = 9
      Event_Message_Type = QoS-Release
      Element_Type = 1
                   32631
      Element_ID =
      Time_Zone = 1+030000
      Sequence_Number = 74
       Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute Count = 2
      Event_Object = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0001
      Upstream
   CableLabs-Event-Message =
3030000004b32303130303532343132313834332e3430370000000880000200
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event_Counter = 10
      Event_Message_Type = QoS-Release
      Element_Type = 1
       Element_ID =
                   32631
      Time_Zone = 1+030000
      Sequence Number = 75
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 2
      Event Object = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0 \times 0002
      Downstream
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
Mon May 24 10:43:02 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
```
```
Acct-Session-Id = "K \setminus 372Ds
                            326311+030000\000\000\t"
   CableLabs-Event-Message =
0x00044bfa44732020203332363331312b30333030303000000000000012020203332363331312b30333030
3030000004c32303130303532343132313834332e3430370000000880000100
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event_Counter = 9
      Event_Message_Type = Call-Disconnect
      Element_Type = 1
      Element_ID =
                   32631
      Time_Zone = 1+030000
      Sequence_Number = 76
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 1
      Event_Object = 0
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-Event-Message =
3030000004d32303130303532343132313834332e343037000000880000300
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID = 32631
      Time_Zone = 1+030000
      Event\_Counter = 9
      Event_Message_Type = Signaling-Stop
      Element_Type = 1
      Element ID =
                   32631
      Time_Zone = 1+030000
      Sequence_Number = 77
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute_Count = 3
      Event_Object = 0
   CableLabs-Related-Call-Billing-Crl-ID =
0x4bfa44732020203332363331312b303330303030000000a
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event_Counter = 10
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-MTA-Endpoint-Name = "SIPPB"
   CableLabs-Event-Message =
30300000004e32303130303532343132313834332e3430370000000880000100
      Version_{ID} = 4
      Timestamp = 1274692723
      Element ID =
                   32631
      Time_Zone = 1+030000
      Event_Counter = 10
      Event_Message_Type = Call-Disconnect
      Element_Type = 1
                   32631
      Element ID =
      Time Zone = 1+030000
      Sequence_Number = 78
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
```

```
Attribute_Count = 1
      Event_Object = 0
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-Event-Message =
3030000004f32303130303532343132313834332e343037000000880000300
      Version_{ID} = 4
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event Counter = 10
      Event_Message_Type = Signaling-Stop
      Element_Type = 1
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Sequence_Number = 79
      Event_Time = 20100524121843.407
      Status = 8
      Priority = 128
      Attribute Count = 3
      Event Object = 0
   CableLabs-Related-Call-Billing-Crl-ID =
0x4bfa44732020203332363331312b3033303030300000009
      Timestamp = 1274692723
      Element_ID =
                     32631
      Time_Zone = 1+030000
      Event_Counter = 9
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-MTA-Endpoint-Name = "SIPPA"
   Acct-Unique-Session-Id = "3479bc93d50898b5"
   Timestamp = 1274712182
   Request-Authenticator = Verified
```

Example 3

The following example shows requests from the SBC to RADIUS server where SBC is configured to include endpoint addressing information in billing records:

```
Tue May 11 13:26:00 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 351GA
                           326311+030000\000\000\000\000
   CableLabs-Event-Message =
3030000000032303130303531313135303230382e3936340000000880000600
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event_Counter = 1
      Event_Message_Type = Signaling-Start
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence Number = 0
      Event_Time = 20100511150208.964
      Status = 8
      Priority = 128
      Attribute_Count = 6
```

```
Event_Object = 0
   CableLabs-Direction-indicator = 0x0001
       Originating
   CableLabs-MTA-Endpoint-Name = "2.0.0.36, 5078, UDP, SIPPB"
   CableLabs-Calling-Party-Number = "
                                                 sipp"
   CableLabs-Called-Party-Number = "
                                              service"
   CableLabs-Routing-Number = "
                                         service"
   CableLabs-Attr-87 = 0 \times 0003
      Billing type -
       flat rate
   CableLabs-Event-Message =
3030000000132303130303531313135303230382e3936340000000880000600
       Version_{ID} = 4
       Timestamp = 1273579329
       Element_ID =
                   32631
      Time_Zone = 1+030000
       Event_Counter = 2
       Event_Message_Type = Signaling-Start
       Element_Type = 1
       Element_ID =
                   32631
       Time_Zone = 1+030000
       Sequence_Number = 1
       Event_Time = 20100511150208.964
       Status = 8
       Priority = 128
      Attribute_Count = 6
       Event_Object = 0
   CableLabs-Direction-indicator = 0x0002
       Terminating
   CableLabs-MTA-Endpoint-Name = "MTA Endpoint"
   CableLabs-Calling-Party-Number = "
                                                 sipp'
   CableLabs-Called-Party-Number = "
                                              service"
   CableLabs-Routing-Number = "
                                         service"
   CableLabs-Attr-87 = 0 \times 0003
      Billing type -
       flat rate
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598760
   Request-Authenticator = Verified
Tue May 11 13:25:59 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 351GA
                             326311+030000\000\000\000\001"
   CableLabs-Event-Message =
3030000000232303130303531313135303230392e303733000000880000400
       Version_{ID} = 4
       Timestamp = 1273579329
       Element_ID = 32631
       Time_Zone = 1+030000
       Event_Counter = 1
       Event_Message_Type = QoS-Reserve
       Element_Type = 1
       Element_ID =
                     32631
       Time_Zone = 1+030000
       Sequence_Number = 2
      Event Time = 20100511150209.073
       Status = 8
       Priority = 128
       Attribute_Count = 4
       Event_Object = 0
```

```
Status_Bitmask = 5
      Service_Class_Name =
      QoS_Parameter_Array = 1
         resource reserved but not committed
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0001
      Upstream
   CableLabs-Event-Message =
3030000000332303130303531313135303230392e3037330000000880000400
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID =
                   32631
      Time_Zone = 1+030000
      Event_Counter = 2
      Event_Message_Type = QoS-Reserve
      Element_Type = 1
      Element_ID =
                  32631
      Time_Zone = 1+030000
      Sequence_Number = 3
      Event_Time = 20100511150209.073
      Status = 8
      Priority = 128
      Attribute_Count = 4
      Event_Object = 0
   Status_Bitmask = 5
      Service_Class_Name =
      QoS\_Parameter\_Array = 1
         resource reserved but not committed
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0002
      Downstream
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598759
   Request-Authenticator = Verified
Tue May 11 13:26:00 2010
  NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K\351GA 326311+030000\000\000\000\001"
   CableLabs-Event-Message =
3030000000432303130303531313135303230392e3537330000000880000200
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID = 32631
      Time_Zone = 1+030000
      Event_Counter = 1
      Event_Message_Type = Call-Answer
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence_Number = 4
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 2
      Event_Object = 0
   CableLabs-Charge-Number = "
                                      sipp"
```

```
CableLabs-Related-Call-Billing-Crl-ID =
0x4be947412020203332363331312b3033303030300000002
      Timestamp = 1273579329
      Element_ID =
                   32631
      Time_Zone = 1+030000
      Event_Counter = 2
   CableLabs-Event-Message =
3030000000532303130303531313135303230392e3537330000000880000300
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID = 32631
      Time_Zone = 1+030000
      Event_Counter = 2
      Event_Message_Type = Call-Answer
      Element_Type = 1
      Element_ID =
                   32631
      Time_Zone = 1+030000
      Sequence_Number = 5
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 3
      Event_Object = 0
   CableLabs-Charge-Number = "
                                       service"
   CableLabs-Related-Call-Billing-Crl-ID =
0x4be947412020203332363331312b3033303030300000001
      Timestamp = 1273579329
      Element_ID =
                   32631
      Time_{Zone} = 1 + 030000
      Event Counter = 1
   CableLabs-MTA-Endpoint-Name = "1.0.0.36,5068,UDP,SIPPA"
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598760
   Request-Authenticator = Verified
Tue May 11 13:26:00 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K\351GA 326311+030000\000\000\000\001"
   CableLabs-Event-Message =
3030000000632303130303531313135303230392e3537330000000880000300
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID = 32631
      Time_Zone = 1+030000
      Event_Counter = 1
      Event_Message_Type = QoS-Commit
      Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence_Number = 6
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 3
      Event_Object = 0
   CableLabs-MTA-UDP-Portnum = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0001
      Upstream
```

CableLabs-Event-Message =

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```
Element_Type = 1
      Element_ID = 32631
      Time_Zone = 1+030000
      Sequence_Number = 9
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 1
      Event_Object = 0
   CableLabs-Attr-93 =
3d302c204c413d302c2050432f5250533d302c2050432f524f533d302c2050432f5250523d302c2050432f5250
4c3d302c2050432f524a493d30
      RTCP Data:
      PS=0, OS=0, PR=0, OR=0, PD=0, OD=0, PL=0, JI=0, LA=0, PC/RPS=0, PC/ROS=0,
PC/RPR=0, PC/RPL=0, PC/RJI=0
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598760
   Request-Authenticator = Verified
Tue May 11 13:26:00 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 351GA
                           326311+030000\000\000\000\001"
   CableLabs-Event-Message =
3030000000a32303130303531313135303230392e3537330000000880000200
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID = 32631
      Time_Zone = 1+030000
      Event Counter = 1
      Event_Message_Type = QoS-Release
      Element_Type = 1
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Sequence_Number = 10
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute Count = 2
      Event_Object = 0
   CableLabs-SF-ID = 0
   CableLabs-Flow-Direction = 0x0001
      Upstream
   CableLabs-Event-Message =
3030000000b32303130303531313135303230392e3537330000000880000200
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID = 32631
      Time_Zone = 1+030000
      Event_Counter = 2
      Event_Message_Type = QoS-Release
      Element_Type = 1
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Sequence_Number = 11
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 2
      Event_Object = 0
   CableLabs-SF-ID = 0
```

```
CableLabs-Flow-Direction = 0x0002
      Downstream
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598760
   Request-Authenticator = Verified
Tue May 11 13:26:00 2010
   NAS-IP-Address = 172.18.53.179
   Acct-Status-Type = Interim-Update
   Acct-Session-Id = "K \setminus 351GA
                           326311+030000\000\000\000\001"
   CableLabs-Event-Message =
3030000000c32303130303531313135303230392e3537330000000880000100
      Version_{ID} = 4
      Timestamp = 1273579329
                    32631
      Element_ID =
      Time_Zone = 1+030000
      Event_Counter = 1
      Event_Message_Type = Call-Disconnect
      Element_Type = 1
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Sequence_Number = 12
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 1
      Event_Object = 0
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-Event-Message =
3030000000d32303130303531313135303230392e3537330000000880000300
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event_Counter = 1
      Event_Message_Type = Signaling-Stop
      Element_Type = 1
      Element TD =
                   32631
      Time_Zone = 1+030000
      Sequence_Number = 13
      Event_Time = 20100511150209.573
      Status = 8
      Priority = 128
      Attribute_Count = 3
      Event_Object = 0
   CableLabs-Related-Call-Billing-Crl-ID =
0x4be947412020203332363331312b3033303030300000002
      Timestamp = 1273579329
      Element_ID =
                    32631
      Time_Zone = 1+030000
      Event Counter = 2
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-MTA-Endpoint-Name = "2.0.0.36,5078,UDP,SIPPB"
   CableLabs-Event-Message =
3030000000e32303130303531313135303230392e3537330000000880000100
      Version_{ID} = 4
      Timestamp = 1273579329
      Element_ID =
                    32631
      Time_Zone = 1+030000
```

```
Event_Counter = 2
       Event_Message_Type = Call-Disconnect
       Element_Type = 1
       Element_ID =
                    32631
      Time_Zone = 1+030000
       Sequence_Number = 14
       Event_Time = 20100511150209.573
       Status = 8
       Priority = 128
       Attribute_Count = 1
       Event_Object = 0
   CableLabs-Call-Termination-Cause = 0x000100000010
       Cause: Normal call clearing
   CableLabs-Event-Message =
3030000000f32303130303531313135303230392e3537330000000880000300
      Version ID = 4
       Timestamp = 1273579329
       Element_ID =
                     32631
      Time_Zone = 1+030000
       Event_Counter = 2
      Event_Message_Type = Signaling-Stop
      Element_Type = 1
       Element_ID =
                     32631
      Time_Zone = 1+030000
       Sequence_Number = 15
       Event_Time = 20100511150209.573
       Status = 8
       Priority = 128
       Attribute_Count = 3
       Event_Object = 0
   CableLabs-Related-Call-Billing-Crl-ID =
0x4be947412020203332363331312b3033303030300000001
      Timestamp = 1273579329
       Element_ID =
                   32631
       Time_Zone = 1+030000
       Event_Counter = 1
   CableLabs-Call-Termination-Cause = 0x000100000010
      Cause: Normal call clearing
   CableLabs-MTA-Endpoint-Name = "1.0.0.36,5068,UDP,SIPPA"
   Acct-Unique-Session-Id = "95a26a97e3e08c3c"
   Timestamp = 1273598760
   Request-Authenticator = Verified
```

Security

The *PacketCable 1.5 Event Messages Specification* mandates that the billing messages are sent using the RADIUS protocol and IPSec for security.

Note

In ACE SBC Release 3.0.00, only the RADIUS security mechanism, based on its own Request Authenticator, is supported.



Secure Media and SRTP Passthrough

Cisco Unified Border Element (SP Edition) supports two methods of encrypted data streams—Secure Real-Time Protocol (SRTP) Passthrough and Secure Media. The preferred method is to use SRTP Passthrough because it allows the end points themselves to signal their encryption capabilities.

The Secure Media feature is enabled on the global level for all calls and is disabled by default. When Secure Media is turned on globally, the SBC assumes that all end points are going to use encrypted data streams regardless of the actual end point capabilities.

Starting with Cisco IOS XE Release 2.6, using the Unsignaled Secure Media feature you are able to configure secure media on a granular level for specific calls and adjacencies using Call Admission Control (CAC) table entry commands.

You can configure SRTP Passthrough on a granular basis using CAC policy.

Regardless of the method used to configure the Cisco Unified Border Element (SP Edition) to accept encrypted media packets, Cisco Unified Border Element (SP Edition) reserves additional bandwidth to ensure these packets pass through. Typically, the bandwidth of a media stream is determined by the codecs that the endpoints use. However, the use of the encryption in the media streams increases the packet size. As a rule of thumb, the bandwidth requirements are 10% more than the unencrypted codec. However, this increase is not reflected in the media flow statistics.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 2.4	These features were introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 2.6	The Unsignaled Secure Media feature was introduced to allow configuration at a granular level using CAC table entry commands. With the introduction of this feature, the Configuring Secure Media-Global Level feature has been deprecated.

Feature History for Secure Media and SRTP Passthrough

Cisco IOS XE Release 3.1S	The SRTP to RTP Interworking and SRTP Passthrough features were added.
Cisco IOS XE Release 3.4S	The SRTP Support for RTCP Multiplexed with RTP and for SSRC-Based Multiplexing feature was added.

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Prerequisites for Secure Media and SRTP Passthrough

The following prerequisites are required to implement both features:

Before implementing the Secure Media and SRTP Passthrough features, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions for Secure Media

The following is a restriction for Global and Unsignaled Secure Media:

• With this feature enabled, RTCP related statistics displayed in the **show sbc dbe media-flow-stats** command are displayed as unknown.

The following is a restriction for Unsignaled (granular-level) Secure Media:

• Both caller and callee sides of the call need to be configured with the **caller secure-media** and **callee secure-media** commands. If only one leg of the call is configured, then the call will fail.

Note

In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** commands. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

Information About Secure Media

Typically, an endpoint will indicate that the media traffic is encrypted through the SIP signaling. The encryption keys are either exchanged through Session Description Protocol (SDP) or using the Datagram Transport Layer Security (DTLS) mechanism.

In Cisco IOS XE Release 2.4 and Release 2.5, Cisco Unified Border Element (SP Edition) interworked with end points or SIP device that use encrypted media (DTLS or Secure-RTP [SRTP]), but the endpoints did not indicate this in the SIP signaling. In those earlier releases, the SBC supported a globally enabled Secure Media configuration where all calls on the SBC were treated as consisting of SRTP media. Even though the endpoint may not have signaled for SRTP media, media pinholes were created as if the traffic was SRTP. A global configuration under the SBE submode indicates that the endpoints are using encrypted SRTP media, but they will not be using SIP signaling to communicate and negotiate as such. The consequence of this configuration being applied at a global level is that even for flows that are not encrypted, additional bandwidth is reserved and RTP and RTCP checking and validations are disabled.

When interworking with a SIP device that does not have full support for signaling SRTP media streams, the SBC cannot know in advance that the media will be SRTP because it is not signaled as SRTP. Starting with Cisco IOS XE Release 2.6, the Unsignaled Secure Media feature allows the SBC to successfully interoperate with SIP devices that generate SRTP media but signal this as a regular RTP media stream.

You are able to configure the SBC to know which SIP devices it communicates with require support for unsignaled SRTP. Such SIP devices are assumed to always send SRTP media. Minimally you must granularly configure all devices on a given adjacency to require support for SRTP. In configuring secure media on a granular level, you use Call Admission Control (CAC) table entry commands. We highly recommend you use the granular level configuration because, instead of turning on secure media globally, you can specify the calls and adjacencies where you want to use secure media. Using the granular option of Unsignaled Secure Media, additional bandwidth is allocated and RTCP no check is performed only for those calls that match the CAC match criteria. Unsignaled Secure Media, like the global option, is disabled by default.

In Cisco IOS XE Release 2.6, when you configure the SBC to allow unsignaled SRTP media on a granular level for adjacencies, observe these recommended guidelines:

- If the adjacencies are trusted to allow secure calls—use either the **security trusted-encrypted** or **security trusted-unencrypted** command to configure both adjacencies where caller and callee side are located for SRTP passthrough first. Both sides need to be configured because it is a passthrough. This is the default where SRTP calls are allowed between trusted adjacencies.
- If an adjacency is not trusted, you can still configure granular-level Unsignaled Secure Media on that adjacency by configuring SRTP Passthrough in a CAC configuration on the untrusted adjacency. Use the **srtp support** command to allow an SRTP call on the adjacency where the CAC policy is applied.
- Configure both legs of the call to enable the granular-level Unsignaled Secure Media—use the **caller** secure-media command on the caller side, and the **callee** secure-media command on the callee side.



In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** commands. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

For information on the configuration steps, see the ?\$paranum>Configuring Unsignaled Secure Media at a Granular Level? section on page 38-13.

Information About SRTP Passthrough

Cisco Unified Border Element (SP Edition) supports SIP calls between endpoints using Transport Layer Security (TLS) for SIP signaling encryption and Secure Real-Time Protocol (SRTP) to provide RTP media encryption. However, these two encryption mechanisms may not be deployed simultaneously, depending on the required call flow invoked on the associated configuration.

Before delving further into SRTP passthrough configuration, it would be useful to understand the two concepts—the *trusted* vs. *untrusted* and *encrypted* vs. *unencrypted*.

The "trusted" implies that an associated adjacency is trusted to allow secure calls. Calls to a standard SIP: URI will be accepted. Calls to a secure SIPS: URI will be accepted and routed over a trusted adjacency (encrypted or unencrypted). The "untrusted" indicates that an associated adjacency is not trusted to carry secure calls. The calls to standard SIP: URI will be accepted. Calls to a secure SIPS: URI will be rejected immediately.

The "encrypted" implies that an associated adjacency uses TLS for SIP signaling and the "unencrypted" implies that an associated adjacency does not use TLS for SIP signaling.

The trusted/untrusted are configured in conjunction with encrypted/unencrypted as outlined in the following four (4) combinations. This is invoked using the **security** command:

- **untrusted-unencrypted**: The adjacency is untrusted and unencrypted. The adjacency is not trusted to carry secure SIP calls (calls with SIPS URI) and it does not use TLS encryption for SIP signaling.
- **untrusted-encrypted**: The adjacency is untrusted and encrypted. The adjacency is not trusted to carry secure SIP calls (calls with SIPS URI) and it does use TLS encryption for SIP signaling.
- **trusted-unencrypted**: The adjacency is trusted and unencrypted. The adjacency is trusted to carry secure SIP calls (calls with SIPS URI) and it does not use TLS encryption for SIP signaling.
- **trusted-encrypted**: The adjacency is trusted and encrypted. The adjacency is trusted to carry secure SIP calls (calls with SIPS URI) and it does use TLS encryption for SIP signaling.

When Cisco Unified Border Element (SP Edition) comes up, the default is to allow SRTP calls to pass through on the trusted interfaces.

The following are conditions of the SRTP Passthrough feature:

- SRTP Passthrough must be configured on both legs of the call. If the target adjacency does not support SRTP Passthrough, then the call is rejected by error message 415 (Unsupported Media Type).
- "m= .. RTP/SAVP .." and a="crypto:..." fields coming in on an Invite from one adjacency are passed on in an Invite to the target adjacency.
- "m= ...RTP/SAVP..." is a required field in the Invite to trigger SRTP Passthrough behavior in the SBC.

The following shows a sample SRTP Invite and Response call flow from endpoints, as described in RFC-4568.

Offerer sends:

```
v=0
o=sam 2890844526 2890842807 IN IP4 10.47.16.5
s=SRTP Discussion
i=A discussion of Secure RTP
u=http://www.example.com/seminars/srtp.pdf
e=marge@example.com (Marge Simpson)
c=IN IP4 168.2.17.12
t=2873397496 2873404696
m=audio 49170 RTP/SAVP 0
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:WVNfX19zZW1jdGwgKCkgewkyMjA7fQp9CnVubGVz|2^20|1:4
FEC_ORDER=FEC_SRTP
a=crypto:2 F8_128_HMAC_SHA1_80
 inline:MTIzNDU2Nzg5QUJDREUwMTIzNDU2Nzg5QUJjZGVm|2^20|1:4;
 inline:QUJjZGVmMTIzNDU2Nzg5QUJDREUwMTIzNDU2Nzg5|2^20|2:4
 FEC_ORDER=FEC_SRTP
```

Answerer replies:

```
v=0
o=jill 25690844 8070842634 IN IP4 10.47.16.5
s=SRTP Discussion
i=A discussion of Secure RTP
u=http://www.example.com/seminars/srtp.pdf
e=homer@example.com (Homer Simpson)
c=IN IP4 168.2.17.11
t=2873397526 2873405696
m=audio 32640 RTP/SAVP 0
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:PS1uQCVeeCFCanVmcjkpPywjNWhcYD0mXXtxaVBR|2^20|1:4
```



Figure 38-1 diagram illustrates an SRTP Passthrough Call Flow.

Figure 38-1 SRTP Passthrough Call Flow

The SRTP Passthrough feature defines a new Call Admission Control (CAC) entry variable, called "srtp transport," in the admission control table. If you configure the "srtp transport" variable, then CAC policy has the option to set the policy for the adjacency to either "allowed," "disallowed," or "trust only."

Calls using SRTP Passthrough are allowed on the adjacencies specified by the policy. Where there are conflicting policies, "disallowed" overrides "allowed" which overrides "trusted-only." If you configure the CAC policy, but you do not define the "srtp transport" variable, then the CAC policy takes the default value of "trusted-only" and restricts the SRTP calls between trusted endpoints.

See the **srtp support** command which sets the adjacency CAC policy for more information. The **no** form of the command sets the "srtp support" variable to "trusted-only." The **show sbc sbe cac-policy-set table entry** command is modified to display a "SRTP Transport" field and whether the policy for the adjacency is to allow, disallow, or trust only for SRTP Transport.

You can set the CAC policy to allow SRTP passthrough and allow configuration of certain security policing, such as the following:

- Preventing secure calls on a given adjacency
- Ensuring that all media sent over a given adjacency is secure
- Ensuring that secure streams are signaled over secure SIP adjacencies.

Information About SRTP to RTP Interworking and SRTP Passthrough

Secure Real-time Transport Protocol (SRTP) to Real-time Transport Protocol (RTP) interworking is supported on Session Border Controller (SBC) services on Cisco ASR 1000 Series Aggregation Services Routers.

System Administrators may configure SRTP to RTP interworking to enable their networks to communicate with other networks and add additional security to a network. SRTP to RTP interworking allows networks that use SRTP to accept calls from networks that use RTP.

The SRTP to RTP interworking feature provides SBC with the ability to encrypt and decrypt data streams to and from both types of networks, SRTP networks and RTP networks.

SRTP to RTP interworking can be deployed on both User to Network Interfaces (UNI) and Network to Network Interfaces (NNI).

Features Supported

The following SRTP to RTP interworking features are supported by SBC:

- SBC-generated SRTP encryption and decryption keys.
- Configurable policies, for SRTP pass-through, termination, and re-origination when both caller and callee CAC policies support SRTP.
- SRTP to RTP interworking in distributed DBE mode via H.248.
- PD logs with information for verifying SBC call handling for different SRTP preference and policy settings. (Encryption keys are not displayed in PD logs.)
- Stateful Switchover (SSO) for SRTP streams.

CAC policies can support the following types of SRTP to RTP interworking:

- RTP-only
- SRTP-only
- SRTP-optional
- SRTP-prefer

When a CAC policy uses SRTP-only:

- All media streams associated to that CAC policy use SRTP. The SRTP stream is end-to-end if the peer adjacency supports SRTP. If the peer adjacency does not support SRTP, or if the policy configuration is set to terminate and re-originate, SBC performs the necessary SRTP encryption and decryption.
- SBC rejects incoming RTP calls and sends the appropriate response code.

When a CAC policy uses RTP-only:

- All media streams associated to that CAC policy use RTP. The RTP stream is end-to-end if the peer adjacency does not require SRTP. If the peer adjacency requires SRTP, SBC perform RTP to SRTP interworking.
- SBC rejects incoming SRTP calls and sends the appropriate response code.

When a CAC policy uses SRTP-optional:

- SRTP-optional is by negotiation on inbound calls.
- SBC accepts both incoming RTP and incoming SRTP calls.
- No RTP to SRTP interworking is needed for incoming RTP calls unless the callee CAC policy uses SRTP-only.
- No SRTP encryption is needed for incoming SRTP calls unless the callee CAC policy uses RTP-only, or the policy configuration prohibits pass-through mode

When a CAC policy uses SRTP-prefer:

- SBC accepts either RTP or SRTP offers from endpoints.
- SBC offers SRTP to endpoints whether the inbound offer is RTP or SRTP.

The following SRTP and RTP statistics are collected and available in show commands at the global level and the adjacency level:

- Number of calls rejected due to RTP requested
- Number of calls rejected due to SRTP requested
- Number of calls using SRTP pass-through
- Number of calls performing RTP to SRTP interworking
- Number of calls using RTP
- Number of calls using SRTP

SIP SRTP Offer Retry Feature

When the SIP SRTP Offer Retry feature is configured, using the **srtp {branch | callee | caller} retry rtp** command, and a 415 or 488 reject error code is generated in response to a prior SRTP (RTP/SAVP) offer, SBC reissues the offer, using RTP (RTP/AVP). This allows SBC to attempt to configure SRTP on a call leg and downgrade it to RTP if SRTP is not supported.

٩, Note

415 and 488 error codes are general purpose errors. After the SRTP Offer Retry feature is configured, the SBC interprets that the 415 and 488 error codes are caused by an initial RTP/SAVP offer.

Downgraded Response to an SRTP Offer

The **srtp {branch | callee | caller} response downgrade** command allows SBC to send an RTP/AVP answer in response to an RTP/SAVP offer and downgrade media security. For instance, if SRTP interworking is not configured in the CAC policy, and the caller offers RTP/SAVP, but the callee answers with RTP/AVP, this command allows the SBC to downgrade the answer to RTP/AVP instead of rejecting the call.

If downgrade is not set, SBC provides strict adherence to the offer/answer protocol and rejects RTP/SAVP offers that are not supported.

This is a non-standard procedure, and is not widely supported. SBC always supports receiving an SRTP downgrade answer, but only sends a downgrade answer when this downgrade flag is set.

Both of the following cases, for SRTP fallback to RTP, are subject to the overall per-side SRTP policy and RTP-SRTP interworking policy:

- If the policy does not allow RTP at all, SBC does not attempt fallback.
- If the policy does not allow RTP-SRTP interworking, SBC allows a fallback on the answer side, but only if SBC can downgrade the offer side as well.

How SBC Processes SRTP

STRP policies behave differently depending on how the following commands are set:

- srtp branch forbid | mandate | allow | prefer
- srtp caller forbid | mandate | allow | prefer
- srtp callee forbid | mandate | allow | prefer
- srtp media interworking forbid | allow
- srtp interworking forbid | allow

The settings for these commands are defined as follows:

- forbid—SRTP is not supported on the caller side or the callee side of the call.
- mandate—SRTP is mandatory on the caller side or the callee side of the call.
- **allow**—SRTP is optional on the caller side or the callee side of the call.
- **prefer**—SRTP is preferred on this adjacency. Both RTP and SRTP are accepted inbound, but only SRTP is offered outbound. When the prefer option is set on the offer side of a call, it functions the same as allow. When the prefer option is set on the answer side of the call, and there is a choice between offering RTP or SRTP, SRTP is offered.

SRTP Policy Passthrough Tables

The following tables show the behavior of SBC based on the configuration of SRTP policies for each side of a call.

Table 38-1 shows how SBC selects the SRTP passthrough type for a stream offered as RTP when an SRTP policy is present.

SRTP Policy		SRTP Passthrough Type	SRTP Passthrough Type	
Offer Side	Answer Side	Interworking Not Possible	Interworking Possible	
Mandate	*	Reject	Reject	
Forbid	Mandate	Reject	RTP-SRTP	
Forbid	Forbid	RTP-RTP	RTP-RTP	
Forbid	Allow	RTP-RTP	RTP-RTP	
Forbid	Prefer	RTP-RTP	RTP-SRTP	
Allow/Prefer	Mandate	Reject	RTP-SRTP	
Allow/Prefer	Forbid	RTP-RTP	RTP-RTP	
Allow/Prefer	Allow	RTP-RTP	RTP-RTP	
Allow/Prefer	Prefer	RTP-RTP	RTP-SRTP	

 Table 38-1
 SBC Processing of RTP Offers with Presence of SRTP Policies

Table 38-2 shows how SBC selects the SRTP passthrough type for a stream offered as SRTP.

SRTP Policy		SRTP Passthrough Type		
Offer Side	Answer Side	No Interworking No Downgrade	Interworking Possible	Downgrade Possible
Forbid	Mandate	Reject	Reject	RTP-SRTP
Forbid	Forbid	Reject	Reject	RTP-RTP
Forbid	Allow	Reject	Reject	RTP-RTP
Forbid	Prefer	Reject	Reject	RTP-SRTP
Mandate	Mandate	SRTP-SRTP	SRTP-SRTP	SRTP-SRTP
Mandate	Forbid	Reject	SRTP-RTP	Reject
Mandate	Allow/Prefer	SRTP-SRTP	SRTP-SRTP	SRTP-SRTP
Allow/Prefer	Mandate	SRTP-SRTP	SRTP-SRTP	SRTP-SRTP
Allow/Prefer	Forbid	Reject	SRTP-RTP	RTP-RTP (3)
Allow/Prefer	Allow/Prefer	SRTP-SRTP	SRTP-SRTP	SRTP-SRTP

 Table 38-2
 SBC Processing of SRTP Policies for SRTP Offers

Table 38-3 shows how SBC selects the SRTP passthrough type when it receives a SIP 415 or SIP 488 rejection code in response to its SRTP offer, and Retry SRTP as RTP is set.

Table 38-3	SBC Processing of SRTP Policies for SRTP Rejection with Retry SRTP as RTP
------------	---

SRTP Policy		SRTP Passthrough Type	
Offer Side Answer Side		Interworking Not Possible	Interworking Possible
*	Mandate	Reject	Reject
Mandate	Allow/Prefer	SRTP-RTP	Reject
Allow/Prefer	Allow/Prefer	SRTP-RTP	RTP-RTP

Table 38-4 shows how SBC selects the SRTP passthrough type when it receives an RTP downgrade answer to an SRTP offer.

Table 38-4 SBC Processing of SRTP Policies for SRTP to RTP Downgrad	ide Answer
---	------------

SRTP Policy		SRTP Passthrough Type	
Offer Side	Answer Side	Interworking Not Possible	Interworking Possible
*	Mandate	Fail Call	Fail Call
Mandate	Allow/Prefer	SRTP-RTP	Fail Call
Allow/Prefer	Allow/Prefer	SRTP-RTP	RTP-RTP

Restrictions

The following restrictions apply to SRTP to RTP interworking and SRTP passthrough:

- Packet cable event messages continue to bill SRTP/RTP interworking calls and SRTP passthrough calls, and the billing does not indicate whether SRTP was used on one or both call legs.
- In late to early interworking and SRTP to RTP interworking, SBC does not support SRTP in a generated SDP offer. The call is forced to be an RTP-RTP call. If this violates the configured call policy, the event is logged and the call fails at setup.
- If a call has multiple streams (multiple m= lines in the SDP), each stream may have a different passthrough type. If any specific stream cannot be satisfied, the call is rejected. Calls with multiple streams and different passthrough types can occur when:
 - An offer is received containing a mix of RTP and SRTP streams.
 - An answer is downgraded from SRTP streams to a subset of RTP streams.
 - Some streams require interworking, others do not.
- SRTP capability is not signaled in H.248 and hence cannot be discovered automatically by SBC. This capability must be manually configured on SBC.
- SBC MG selection does not select an MG on the basis of which crypto-suites it supports.
- SBC does not allow the user to configure distinct SRTP session parameters on a per-call basis.
- SBC SRTP features do not work in conjunction with unsignaled SRTP.
- SBC will fail an SRTP call if it receives a SIP forking answer.
- SIP late-early interworking does not support SRTP.
- H.323-SIP interworking does not support SRTP.
- SBC cannot terminate RFC5027 security preconditions signaling in RTP-SRTP calls.
- SBC does not support local call transfer of SRTP calls.
- SBC currently only supports the AES_CM_128_HMAC_SHA1_32 crypto suite.
- SBC does not refresh any master keys that it generates.
- SBC does not renegotiate master key rotation when the packet usage count is reached (as specified in RFC3711).
- If the transcoder does not support SRTP (such as MGX), SBC does not allow an SRTP-SRTP call. SBC cannot perform SRTP-RTP interworking on the two media gates on either side of the transcoder.
- RTP-SRTP and SRTP-RTP calls can be transcoded by a third-party transcoder. In such cases, the media through the transcoder RTP, and the interworking is performed by SBC on the side closest to the SRTP endpoint.

To configure SRTP to RTP interworking, see the ?\$paranum>Configuring CAC Policies for SRTP to RTP Interworking? section on page 38-23 and the ?\$paranum>CAC Policies for SRTP to RTP Interworking Configuration: Example? section on page 38-33

You can display policy failure statistics for a specified source adjacency, using this existing command that has been updated for SRTP:

show sbc sbe call-stats src-adjacency

You can display all the calls on the SBEs, using this existing command that has been updated for SRTP: show sbc sbe calls srtp-iw

Configuring Secure Media—Global Level



The Unsignaled Secure Media feature was introduced in Cisco IOS XE Release 2.6 to allow configuration of secure media at a granular level using CAC table entry commands. With the introduction of this feature, the Configuring Secure Media-Global Level feature has been deprecated. If you are upgrading from a release earlier than Release 2.6, see the procedure described in the ?\$paranum>Configuring Unsignaled Secure Media at a Granular Level? section on page 38-13.

Perform the following steps to configure secure media globally.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. secure-media
- 5. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config) # sbc mysbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	secure-media	Configures the SBC to treat every media flow as an encrypted media flow. This allows media packets, such as
	Example: Router(config-sbc-sbe)# secure-media	DTLS and SRTP packets, to pass through the SBC.
Step 5	end	Exits SBE mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	

Configuring Unsignaled Secure Media at a Granular Level

Use the following steps to configure both adjacencies and both call legs using CAC policy set to enable Unsignaled Secure Media at a granular level.

Note The call

The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. security [untrusted | trusted-encrypted | untrusted-encrypted | trusted-unencrypted]
- 6. exit
- 7. adjacency {sip | h323} adjacency-name
- 8. security [untrusted | trusted-encrypted | untrusted-encrypted | trusted-unencrypted]
- 9. exit
- 10. cac-policy-set policy-set-id
- 11. first-cac-table table-name
- **12.** cac-table table-name
- 13. table-type limit list of limit tables
- 14. entry entry-id
- **15.** match-value key
- 16. srtp support [allow | disallow | trusted-only]
- 17. caller secure-media
- 18. callee secure-media
- **19.** action {cac-complete | next-table goto-table-name}
- 20. exit
- 21. complete
- 22. exit
- 23. active-cac-policy-set policy-set-id
- 24. end
- **25.** show sbc *sbc-name* sbe cac-policy-set [id [table name [entry id]] | active [table name [entry id]]] [detail]



DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the caller side SIP adjacency, that is named 'client' in the example. And enters the mode of an SBE SIP
	<pre>Example: Router(config-sbc-sbe)# adjacency sip client</pre>	aujacency, often caned aujacency sip mode.
Step 5	security [untrusted trusted-encrypted untrusted-encrypted trusted-unencrypted]	Configures transport-level security (TLS) on a SIP adjacency.
	Example:	For granular-level Secure Media, configure the trusted adjacency as trusted-encrypted or trusted-unencrypted.
	Router(config-sbc-sbe-adj-sip)# security trusted-encrypted	Trusted means the adjacency is trusted to carry secure SIP calls (calls with SIPS URI). Encrypted means the adjacency uses TLS encryption for SIP signaling. Unencrypted means it does not use TLS encryption for SIP signaling.
		Note If this adjacency is <i>untrusted</i> , skip steps Step 4 through Step 6. You need to configure for an untrusted adjacency in a CAC policy table.
Step 6	exit	Exits the SBE SIP adjacency mode to the SBE mode.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# exit</pre>	
Step 7	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the callee side SIP adjacency, that is named 'server' in the example. And enters the mode of an SBE SIP
	Example: Router(config-sbc-sbe)# adjacency sip server	adjacency, often called adjacency sip mode.

	Command or Action	Purpose
Step 8	security [untrusted trusted-encrypted untrusted-encrypted trusted-unencrypted]	Configures transport-level security (TLS) on a SIP adjacency.
	Example:	For granular-level Secure Media, configure the trusted adjacency as trusted-encrypted or trusted-unencrypted.
	Router(config-sbc-sbe-adj-sip)# security trusted-unencrypted	Trusted means the adjacency is trusted to carry secure SIP calls (calls with SIPS URI). Encrypted means the adjacency uses TLS encryption for SIP signaling. Unencrypted means it does not use TLS encryption for SIP signaling.
		Note If this adjacency is <i>untrusted</i> , skip steps Step 7 through Step 9. You need to configure for an untrusted adjacency in a CAC policy table.
Step 9	exit	Exits the SBE SIP adjacency mode to the SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 10	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 - 2147483647.
Step 11	<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)#</pre>	Configures the name of the first policy table to process. A CAC policy may have many tables configured. To start the application of the CAC policy, the first table that is used needs to be defined.
	first-cac-table testSecure	<i>table-name</i> —The admission control table that should be processed first.
Step 12	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose	
Step 13	table-type limit list of limit tables	Configures a new CAC Limit table type where you enter the criteria that is used to match the entries.	
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit all	list of limit tables can be one of the following values:	
		• account —Compare the name of the account.	
		• adj-group —Compare the name of the adjacency group.	
		• adjacency —Compare the name of the adjacency.	
		• all —No comparison type. All events match this type.	
		• call-priority —Compare with call priority.	
		• category —Compare the number analysis assigned category.	
		• dst-account —Compare the name of the destination account.	
		• dst-adj-group —Compare the name of the destination adjacency group.	
		• dst-adjacency —Compare the name of the destination adjacency.	
		• dst-prefix —Compare the beginning of the dialed digit string.	
		• event-type —Compare with CAC policy event types.	
		• src-account —Compare the name of the source account.	
		• src-adj-group —Compare the name of the source adjacency group.	
		• src-adjacency —Compare the name of the source adjacency.	
		• src-prefix —Compare the beginning of the calling number string.	
Step 14	entry entry-id	Enters the mode to modify an entry in an admission control table.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	<i>entry-id</i> —Specifies the table entry.	
Step 15	match-value key	Configures the match-value of an entry in a CAC Limit table type.	
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value call-update		

	Command or Action	Purpose
Step 16	<pre>srtp support [allow disallow trusted-only] Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp support allow</pre>	If an adjacency is <i>untrusted</i> and you want granular-level Secure Media, you need to configure this step—configuring with srtp support allow will allow an SRTP call on the untrusted adjacency where the CAC policy is applied. Continue on to Step 17.
		Configures the srtp support variable in the CAC table to allow or disallow SRTP Passthrough of secure media on the adjacency where the policy is applied.
		• allow —allows SRTP Transport when an event matches this CAC policy.
		• disallow —do not allow SRTP Transport when an event matches this CAC policy.
		• trusted-only —allows SRTP Transport on a trusted adjacency (default) when an event matches this CAC policy.
		Calls using SRTP Passthrough are allowed on the adjacencies specified by the policy.
Step 17	caller secure-media	Configures a Secure Media call on the caller side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller secure-media</pre>	
Step 18	callee secure-media	Configures a Secure Media call on the callee side.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee secure-media</pre>	
Step 19	action {cac-complete next-table goto-table-name}	Configures the action to perform after this entry in an admission control table. Each entry requires a match criteria and an action. The action is to accept the transport.
	Example:	action is one of the following:
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• cac-complete —When an event matches, this CAC policy is complete.
		• next-table —Specifies the name of the next cac table.
		• <i>goto-table-name</i> —Specifies the table name identifying the next CAC table to process (or cac-complete, if processing should stop).
Step 20	exit	Exits CAC Table Entry mode and enters CAC Policy-set configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	

	Command or Action	Purpose
Step 21	complete	Completes the CAC-policy set after committing the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 22	exit	Exits CAC Policy-set configuration mode and enters SBE mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# exit</pre>	
Step 23	active-cac-policy-set policy-set-id	Sets the newly created CAC policy to be active. When the policy is active, it can no longer be modified.
	<pre>Example: Router(config-sbc-sbe)# active-cac-policy-set 1</pre>	<i>policy-set-id</i> —Identifies the policy set that is made active. Range is 1 to 2147483647.
Step 24	end	Exits the SBE mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 25	<pre>show sbc name sbe cac-policy-set [id [table name [entry id]] active [table name [entry id]]] [detail]</pre>	Displays detailed information for a given entry in a CAC policy table. In this example, that includes the caller/callee unsignaled secure media: Allowed fields and the security trusted-unencrypted for both adjacencies of the Secure
	Example: Router# show sbc mysbc sbe cac-policy-set 1 detail	Media call.

Configuring SRTP Passthrough

These steps show how to configure the CAC policy set to allow SRTP Passthrough.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name
- 6. first-cac-table *table-name*
- 7. cac-table *table-name*
- 8. table-type limit list of limit tables
- 9. entry entry-id
- 10. match-value key
- 11. srtp support [allow | disallow | trusted-only]
- 12. action [cac-complete | next-table | goto-table-name]

- 13. exit
- 14. exit
- 15. complete
- 16. exit
- 17. active-cac-policy-set policy-set-id
- 18. end
- 19. show sbc sbc-name sbe cac-policy-set id table name entry entry

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 - 2147483647.

	Command or Action	Purpose
Step 5	<pre>first-cac-scope scope-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope call</pre>	Configures scope at which limits should be initially defined when performing the admission control stage of the policy. Each CAC policy has a scope that is applied to it. This CAC policy applies on a per call basis.
		scope-name has one of the following values:
		• adj-group —Limits for events from members of the same adjacency group.
		• call —Limits are per single call.
		• category—Limits per category.
		• dst-account —Limits for events sent to the same account.
		 dst-adj-group—Limits for events sent to the same adjacency group. dst-adjacency—Limits for events sent to the same adjacency.
		• dst-number —Limits for events that have the same adjacency number.
		• global —Limits are global (May not be combined with any other option).
		• src-account —Limits for events from the same account.
		• src-adj-group —Limits for events from the same adjacency group.
		• arc-adjacency —Limits for events from the same adjacency.
		• src-number —Limits for events that have the same source number.
Step 6	<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table testSecure</pre>	Configures the name of the first policy table to process. A CAC policy may have many tables configured. To start the application of the CAC policy, the first table that is used needs to be defined
		<i>table-name</i> —The admission control table that should be processed first.
Step 7	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure</pre>	<i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose
Step 8	table-type limit list of limit tables	Configures a new CAC Limit table type where you enter the criteria that is used to match the entries.
	Example:	<i>list of limit tables</i> can be one of the following values:
	Router(config-sbc-sbe-cacpolicy-cactable)#	• account —Compare the name of the account.
	table-type limit all	• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all—No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
Step 9	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 10	match-value key	Configures the match-value of an entry in a CAC Limit table type.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value call-update</pre>	

	Command or Action	Purpose
Step 11	<pre>srtp support [allow disallow trusted-only] Example:</pre>	Configures the srtp support variable in the CAC table to allow or disallow SRTP Passthrough of secure media on the adjacency where the policy is applied.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp support allow	• allow —allows SRTP Transport when an event matches this CAC policy.
		• disallow —do not allow SRTP Transport when an event matches this CAC policy.
		• trusted-only —allows SRTP Transport on a trusted adjacency (default) when an event matches this CAC policy.
		Calls using SRTP Passthrough are allowed on the adjacencies specified by the policy. Where there are conflicting policies, "disallowed" overrides "allowed" which overrides "trusted-only."
Step 12	<pre>action [cac-complete next-table goto-table-name]</pre>	Configures the action to perform after this entry in an admission control table. Each entry requires a match criteria and an action. The action is to accept the transport.
	Example:	action is one of the following:
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete	• cac-complete —When an event matches, this CAC policy is complete.
		• next-table —Specifies the name of the next cac table.
		• <i>goto-table-name</i> —Specifies the table name identifying the next CAC table to process (or cac-complete, if processing should stop).
Step 13	exit	Exits CAC table entry submode and enters into cacpolicy cactable mode
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit	
Step 14	exit	Exits cacpolicy cactable submode and enters into cacpolicy mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit</pre>	
Step 15	complete	Completes the CAC policy after all the entries within the CAC tables have been configured.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 16	exit	Exits the cacpolicy submode and enters into SBE mode.
	Example: Router(config-sbc-sbe-cacpolicy)# exit	

	Command or Action	Purpose
Step 17	active-cac-policy-set policy-set-id	Sets the newly created CAC policy to be active. When the policy is active, it can no longer be modified.
	<pre>Example: Router(config-sbc-sbe)# active-cac-policy-set 1</pre>	<i>policy-set-id</i> —Identifies the policy set that is made active. Range is 1 to 2147483647.
Step 18	end	Exits the SBE mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 19	show sbc <i>sbc-name</i> sbe cac-policy-set <i>id</i> table <i>name</i> entry <i>entry</i>	Displays detailed output, including a "SRTP Transport" field and whether the policy for the adjacency is to allow, disallow, or trust only for SRTP Transport.
	Example:	
	Router# show sbc mysbc sbe cac-policy-set 1 table testSecure entry 1	

Configuring CAC Policies for SRTP to RTP Interworking

Use the following procedure to configure the CAC policies for the caller side and the callee side of a call for SRTP to RTP interworking.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name

CAC Table for Caller Side of the Call

- 6. cac-table table-name
- 7. table-type limit list of limit tables

(repeat steps 8 through 14 as many times as needed)

- 8. entry entry-id
- 9. match-value key
- 10. srtp support allow
- **11.** action next-table goto-table-name
- 12. srtp caller forbid | mandate | allow | prefer
- 13. srtp interworking forbid | allow
- 14. srtp media interworking forbid | allow

CAC Table for Callee Side of the Call

15. cac-table table-name

16. table-type limit list of limit tables

(repeat steps 17 through 23 as many times as needed)

- 17. entry entry-id
- 18. match-value key
- 19. srtp support allow
- **20**. action cac-complete
- 21. srtp callee forbid | mandate | allow
- 22. srtp interworking forbid | allow
- 23. srtp media interworking forbid | allow

(issue complete command after all entries are configured)

- 24. complete
- 25. end
- 26. show sbc name sbe cac-policy-set id detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set.
	Example: Router(config-sbc-sbe)# cac-policy-set 44	• <i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	first-cac-table table-name	Specifies which CAC table is processed first.
		• <i>table-name</i> —The name table to be processed first.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table 44	r

CAC Table for Caller Side of the Call

	Command or Action	Purpose
Step 6	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example:	SBE policy set.
	Router(config-sbc-sbe-cacpolicy)# cac-table 44	• <i>table-name</i> —Name of the admission control table.
Step 7	table-type limit list of limit tables	Configures the limit of the table types to be matched by the match-value command. For this example, use the following
	Example:	table type:
	Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit src-adjacency	• <i>src-adjacency</i> —Compare the name of the source adjacency.

Repeat steps 8 through 14 as many times as necessary to configure as many entries as needed.

Step 8	entry entry-id	Enters the mode to modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	• <i>entry-id</i> —Specifies the table entry.
Step 9	match-value key	Configures the match-value of an entry in a Call Admission Control (CAC) Limit Table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value A</pre>	• <i>key</i> —Specifies the keyword used to match events. The format of the key is determined by the table-type limit.
Step 10	srtp support allow	Configures SRTP support.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp support allow</pre>	
Step 11	action next-table goto-table-name	Configures the action to take when this routing entry is chosen.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action next-table 45</pre>	• <i>goto-table-name</i> —Specifies the next routing table to process when an event matches the entry.
Step 12	<pre>srtp caller forbid mandate allow prefer</pre>	Configures SRTP for the caller side of the call with one of the following SRTP settings.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	• forbid —SRTP is not supported on the caller side of the call.
	# srtp caller forbid	• mandate —SRTP is mandatory on the caller side of the call.
		• allow —SRTP is optional on the caller side of the call.
		• prefer —SRTP is preferred on this adjacency. Both RTP and SRTP are accepted inbound, but only SRTP is offered outbound.

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	Command or Action	Purpose
Step 13	<pre>srtp interworking forbid allow</pre>	Configures SRTP to RTP interworking.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	 forbid—Prohibits SRTP to RTP interworking on the call. allow—Allows SRTP to RTP interworking on the call
Stop 1/	<pre># srtp interworking allow srtp media interworking forbid allow</pre>	Configures SPTD to PTD media interworking
Step 14	Example:	 forbid—Prohibits SRTP to RTP media interworking on the call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp media interworking allow	• allow —Allows SRTP to RTP media interworking on the call.

CAC Table for Callee Side of the Call

Step 15	<pre>cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# cac-table 45</pre>	 Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set. <i>table-name</i>—Name of the admission control table.
Step 16	<pre>table-type limit list of limit tables Example:</pre>	Configures the limit of the table types to be matched by the match-value command. For this example, use the following table type:
	Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit src-adjacency	• <i>src-adjacency</i> —Compare the name of the source adjacency.

Repeat steps 17 through 23 as many times as necessary to configure as many entries as needed.

Step 17	entry entry-id	Enters the mode to modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	• <i>entry-id</i> —Specifies the table entry.
Step 18	match-value key	Configures the match-value of an entry in a Call Admission Control (CAC) Limit Table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value A</pre>	• <i>key</i> —Specifies the keyword used to match events. The format of the key is determined by the table-type limit.
Step 19	srtp support allow	Configures SRTP support.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp support allow</pre>	
Step 20	action next-table goto-table-name	Configures the action to take when this routing entry is selected.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action next-table 45	• <i>goto-table-name</i> —Specifies the next routing table to process if the event matches the entry.

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	Command or Action	Purpose
Step 21	<pre>srtp callee forbid mandate allow prefer</pre>	Configures SRTP for the callee side of the call.
	Example:	• forbid —SRTP is not supported on the callee side of the call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp callee forbid	• mandate —SRTP is mandatory on the callee side of the call.
		• allow —SRTP is optional on the callee side of the call.
		• prefer —SRTP is preferred on this adjacency. Both RTP and SRTP are accepted inbound, but only SRTP is offered outbound.
Step 22	srtp interworking forbid allow	Configures SRTP to RTP interworking.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	 forbid—Prohibits SRTP to RTP interworking on the call. allow—Allows SRTP to RTP interworking on the call.
Sten 23	<pre># srtp interworking allow</pre>	Configures SRTP to RTP media interworking
516p 25	Example:	 forbid—Prohibits SRTP to RTP media interworking on the call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp media interworking allow	• allow —Allows SRTP to RTP media interworking on the call.

Issue the complete command only after all entries are configured.

Step 24	complete	Completes the CAC-policy after all entries are entered.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # complete</pre>		
Step 25	end	Exits configuration mode and returns to privileged EXEC mode.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>		
Step 26	show sbc name sbe cac-policy-set id detail Example:	Displays detailed information for the given entry ID in a CAC policy table. In this case, it shows the default values for SRTP-RTP interworking. For example:	
	Router# show sbc SBC1 sbe cac-policy-set 1 detail	Caller SRTP support: Inherit (default) Callee SRTP support: Inherit (default) SRTP Interworking: Inherit (default) SRTP media Interworking: Inherit (default)	

SRTP Support for RTCP Multiplexed with RTP

In earlier releases, the SBC could process incoming RTP and RTCP streams that were sent over separate UDP channels. From Release 3.4S, the SBC can also process RTCP streams multiplexed with RTP streams and sent over a single UDP channel. The SBC distinguishes between RTCP and RTP streams by examining the payload format of each stream. This also applies to SRTCP streams multiplexed with SRTP streams.

Note

RFC 5761 describes the multiplexing of RTCP streams with RTP streams. The same principle applies to SRTCP and SRTP.

This feature is an enhancement to the support for interworking of RTP-based and SRTP-based endpoints that are linked through the SBC. The Cisco TelePresence System is an example of an RTP-based endpoint, and Cisco Umi TelePresence is an example of an SRTP-based endpoint. With the introduction of this feature, the SBC processes RTCP streams multiplexed with RTP streams coming from the Cisco TelePresence System. In a similar manner, the SBC identifies and correctly processes SRTCP streams multiplexed with SRTP streams coming from Cisco Umi TelePresence.

By default, the detection of RTCP streams multiplexed with RTP streams is disabled in the SBC. You can enable this feature by performing the procedure described in the following section.

Configuring the Detection of RTCP Multiplexed with RTP

This task explains how to configure the detection of RTCP streams multiplexed with RTP streams.



The same procedure can be used to configure the detection of SRTCP streams multiplexed with SRTP streams.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. rtcp-mux

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# sbc MySbc	
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	
Step 4	rtcp-mux	Enables the detection of RTCP streams multiplexed with RTP streams.
	Example: Router(config-sbc-sbe)# rtcp-mux	By default, this feature is disabled.

SRTP Support for SSRC-Based Multiplexing

An SBC endpoint such as the Cisco TelePresence System multiplexes RTP streams of the same type (audio or video) on a single UDP channel. It uses the 32-bit synchronization source (SSRC) field of RTP streams to differentiate between discrete RTP streams originating from a single source.

When an SRTP-based or RTP-based endpoint sends multiplexed streams over a single UDP channel, the channel contains multiple streams and each stream has its own SSRC field. In earlier releases, the SBC could support only a single SSRC field in a UDP channel. Therefore, the SBC could not support interworking of endpoints that sent multiplexed SRTP and RTP. From Release 3.4S, the SBC can process multiple SSRC fields in multiplexed SRTP or RTP streams. In combination with SRTP support for RTCP multiplexed with RTP, this feature enhances interworking of RTP-based and SRTP-based endpoints.

Configuring Global Secure Media Example

This section provides a sample configuration for the Secure Media Passthrough feature.

Router# configure Router(config)# sbc mysbc Router(config-sbc)# sbe Router(config-sbc-sbe)# secure-media Router(config-sbc-sbe)# end

Configuring Unsignaled, Granular-Level Secure Media: Examples

The following configuration example shows how the client and server SIP adjacencies are configured as "security trusted-unencrypted" and how the CAC table entry 1 is configured for secure media on both the caller and callee sides.

```
<u>Note</u>
```

The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

```
. . .
cac-policy-set 2
first-cac-table 1
cac-table 1
 table-type limit all
  entry 1
  match-value call-update
  caller secure-media
   callee secure-media
   action cac-complete
   exit
 complete
 exit
active-cac-policy-set 2
adjacency sip client
nat force-off
 security trusted-unencrypted
 signaling-address ipv4 10.10.100.110
 signaling-port 9060
 remote-address ipv4 10.10.100.10 255.255.255.255
 signaling-peer 10.10.100.10
signaling-peer-port 9060
attach
adjacency sip server
nat force-off
 security trusted-unencrypted
 signaling-address ipv4 10.10.100.110
 signaling-port 9070
 remote-address ipv4 10.10.100.10 255.255.255.255
 signaling-peer 10.10.100.10
 signaling-peer-port 9070
attach
```

The following example shows how to configure granular-level unsignaled secure media where an adjacency is *untrusted* by using the **srtp support allow** command on the untrusted adjacency in a CAC policy table:

```
cac-policy-set 2
first-cac-table 1
cac-table 1
table-type limit all
entry 1
match-value call-update
```

```
srtp support allow
   caller secure-media
   callee secure-media
   action cac-complete
   exit
 complete
 exit
active-cac-policy-set 2
```

Router# show sbc asr sbe cac-policy-set 2 detail

The following example lists detailed information pertaining to CAC policy set 2, and shows how secure media is configured on the caller and callee sides:

SBC Service "asr"	
CAC Policy Set 2 Active policy set: Yes Description: Averaging period: 60 sec First CAC table: 1 First CAC scope: global First CAC prefix length: 4294967250	6
Description.	
Table type: policy-set	Total call failures. 0
Table type: policy-set	Total Call Tallures: 0
Entry 1 CAC scope: CAC scope prefix length: 0	
Action: CAC complete	Number of calls rejected: 0
Max calls per scope: Unlimited	d Max call rate per scope: Unlimited
Max in-call rate: Unlimited	d Max out-call rate: Unlimited
Max req. per scope: Unlimited	d Max reg. rate per scope: Unlimited
Max channels per scope: Unlimited	d Max updates per scope: Unlimited
Early media: Allowed	Early media direction: Both
Early media timeout: None	Transcoder per scope: Allowed
Callee Bandwidth-Field: None	Caller Bandwidth-Field: None
Media bypass:	Allowed
Renegotiate Strategy:	Delta
Max bandwidth per scope:	Unlimited
SRTP Transport:	Trusted-Only (by default)
Caller hold setting:	Standard
Callee hold setting:	Standard
Caller privacy setting:	Never hide
Callee privacy setting:	Never hide
Caller voice QoS profile:	Default
Callee voice QoS profile:	Default
Caller video QoS profile:	Default
Callee video QoS profile:	Default
Caller sig QoS profile:	Default
Callee sig QoS profile:	Default
Caller inbound SDP policy:	None
Callee inbound SDP policy:	None
Caller outbound SDP policy:	None
Callee outbound SDP policy:	None
Caller media disabled:	
Strip All Answer	
Callee media disabled:	
Strip All Offer	
Caller unsignaled secure media:	Allowed
Callee unsignaled secure media:	Allowed
Caller tel-event payload type:	Detault

```
Callee tel-event payload type: Default
Media flag:
Ignore bandwidth-fields (b=), Telephone Event Interworking
Restrict codecs to list: Default
Restrict caller codecs to list: Default
Restrict callee codecs to list: Default
Maximum Call Duration: Unlimited
```

The following example shows an excerpt of detailed information for the callee side SIP adjacency 'server' showing that security trusted-unencrypted is configured:

```
Router# show sbc asr sbe adjacencies server detail
```

```
SBC Service "asr"

Adjacency server (SIP)

Status: Attached

[snip]

Security: Trusted-Unencrypted

[snip]
```

Configuring SRTP Passthrough Example

The following shows a configuration where the "srtp transport" variable is set in the CAC policy set 1 table for an adjacency to allow SRTP Passthrough:

```
sbc SBE-NODE2-SBE1
   sbe
      cac-policy-set 1
         first-cac-scope global
        first-cac-table STANDARD-LIST-BY-ACCOUNT
         cac-table STANDARD-LIST-BY-ACCOUNT
            table-type limit dst-account
            entry 1
               media-bypass-forbid
                match-value STP-CUSTOMER-1
               max-num-calls 100
               max-call-rate 20
               max-bandwidth 1000000 bps
               callee-privacy never
               srtp support allow
               action cac-complete
               exit
            entry 2
               match-value SIP-CUSTOMER-2
               max-num-calls 100
               max-call-rate 20
               max-bandwidth 1000000 bps
               transcode-deny
               max-regs 500
               action cac-complete
               exit
               exit
               complete
      active-call-policy-set 1
```

The following example displays entries in table CAC1 for CAC policy set 100 and shows that the SRTP Transport variable has been set to allow SRTP Passthrough on whichever adjacency the policy is applied:

Router# show sbc SBC1 sbe cac-policy-set 100 table CAC1 entry 1000

SBC Service "SBC1"

```
Policy set 100 table CAC1 entry 1000
 Match value
                        src-adjacency
                         CAC policy complete
 Action
 Max calls
                         Unlimited
 Max call rate
                         100
 Max registrations
Max reg. rate
Max bandwidth
                        Unlimited
                         Unlimited
                         Unlimited
 Max bandwidth
 Max channels
                         Unlimited
 Transcoder
                          Allowed
 Caller privacy setting Never hide
 Callee privacy setting Never hide
 Early media
                         Allowed
 Early media direction Both
 Early media timeout
                        0
 Restrict codecs to list default
 Media bypass
                         Allowed
 Number of calls rejected by this entry
                                         0
 SRTP Transport
                          Allowed
```

CAC Policies for SRTP to RTP Interworking Configuration: Example

The following example shows specific details of how to configure the CAC policies for the caller side and the callee side of a call for SRTP to RTP interworking. Multiple entries with specific settings are given. Figure 38-2 shows the adjacencies that are used by the match-value command in this example.



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entry 2 match-value B srtp support allow action next-table 45 srtp caller forbid srtp interworking allow srtp media interworking allow entry 3 match-value C srtp support allow action next-table 45 srtp caller mandate srtp interworking allow srtp media interworking allow entry 4 match-value D srtp support allow action next-table 45 srtp caller mandate srtp interworking allow srtp media interworking allow cac-table 45 table-type limit dst-adjacency entry 1 match-value A srtp support allow action cac-complete srtp callee forbid srtp interworking allow srtp media interworking allow entry 2 match-value B srtp support allow action cac-complete srtp callee forbid srtp interworking allow srtp media interworking allow entry 3 match-value C srtp support allow action cac-complete srtp callee mandate srtp interworking allow srtp media interworking allow entry 4 match-value D srtp support allow action cac-complete srtp callee mandate srtp interworking allow srtp media interworking allow complete end

show sbc sbc1 sbe cac-policy-set 44 detail





Implementing QoS (Marking)

Cisco Unified Border Element (SP Edition) supports quality of service (QoS) profiles that the integrator configures for IP packet marking on the data path. IP packet marking is used in Cisco Unified Border Element (SP Edition) in the following contexts:

- Configuring media packet real-time transport protocol (RTP) and real-time control protocol (RTCP) marking based on a per call scope.
- Supporting Differentiated Services Code Point (DSCP) marking as well as IP precedence/Type of Service (ToS) marking for voice service.
- Enabling the unique marking of media packets depending on the branch of the call (either the caller branch or the callee branch) on which the packets are sent.
- Supporting signaling and media packet marking based on Session Initiation Packet (SIP) resource priority header.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 3.3S	The Flow Statistics Enhancements feature was introduced in the Cisco ASR 1000 Series Routers.
Cisco IOS XE Release 3.4S	The QoS Demarcation feature was introduced in the Cisco ASR 1000 Series Routers. The Flow Statistics Enhancements feature mentioned in the preceding row is part of the QoS Demarcation feature.

Feature History for Implementing QoS (Marking)

Contents

This chapter contains the following sections:

- Prerequisites for Implementing QoS, page 38-2
- Information About Implementing QoS, page 38-2
- How to Implement QoS, page 38-2
- Implementing QoS Demarcation, page 38-10
- Configuration Examples of QoS Profiles, page 38-23

Prerequisites for Implementing QoS

The following is the prerequisite to implement QoS on Cisco Unified Border Element (SP Edition): Before implementing QoS, Cisco Unified Border Element (SP Edition) must already be configured.

Information About Implementing QoS

To implement QoS marking on Cisco Unified Border Element (SP Edition), the user configures Cisco Unified Border Element (SP Edition) with a number of QoS profiles, which are given unique names to identify them. These QoS profiles are used exclusively for marking packets.

Each QoS profile contains the following mutually exclusive parameters.

- A 6-bit DSCP value to mark packets that match the QoS.
- A 3-bit IP precedence value and a 4-bit ToS value to mark packets that match the QoS.

Note

A default QoS profile that can be neither modified nor deleted is preconfigured on Cisco Unified Border Element (SP Edition). If the user does not define a QoS profile, the default QoS profile is used for marking packets.

QoS signaling profiles are currently supported only for SIP signaling.

How to Implement QoS

To implement QoS marking on Cisco Unified Border Element (SP Edition), follow the procedures in the following sections:

- Configuring QoS Profiles
- Selecting a QoS Profile Using CAC

Configuring QoS Profiles

This task configures a signaling QoS profile to use an IP precedence value of 1 and a ToS value of 12 to mark packets that match the QoS.

<u>Note</u>

QoS signaling profiles are currently supported only for SIP signaling.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. qos sig name
- 5. marking *type*
- 6. ip precedence *value*
- 7. ip tos value

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc Router(config-sbc)#	• Use the <i>sbc-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of an signaling border element (SBE) entity within a SBC service.
	Example: Router(config-sbc)# sbe Router(config-sbc-sbe)#	
Step 4	gos sig name Example:	Enters the mode of configuring a QoS profile. The <i>name</i> parameter must be the name of an existing QoS profile. The string "default" is reserved.
	Router(config-sbc-sbe)# qos sig residential Router(config-sbc-sbe-qos-sig)#	
Step 5	marking type	Configures whether the QoS policy marks packets with a DSCP value or an IP precedence and ToS value or a policy that does not mark. The type can be one of the following:
	Example:	that does not mark. The type can be one of the following.
	Router(config-sbc-sbe-qos-sig)# marking ip-precedence	• dscp
	ip precedence	• ip-precedence
		• passthrough—creates a QoS policy that does not mark packets.
		The no version of this command removes the QoS policy.

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	Command or Action	Purpose
Step 6	ip precedence value	Configures an IP precedence with which to mark IP packets belonging to the given QoS profile. The range of IP precedence values is 0 to 7.
	Router(config-sbc-sbe-qos-sig)# ip precedence 1	The no version of this command sets the default IP precedence value to 0.
		Note If the QoS profile is configured to mark packets DSCP value takes precedence.
Step 7	ip tos value	Configures an IP ToS with which to mark IP packets belonging to the given QoS profile. The <i>value</i> parameter is
	Example: Router(config-sbc-sbe-qos-sig)#	a bit field consisting of one or more of the following bits linked together using an arithmetic OR:
	ip tos 12	• 8—Minimize delay
		• 4—Maximize throughput
		• 2—Maximize reliability
		• 1—Minimize monetary cost

Analyzing the SIP Resource-Priority Header

Users can configure Cisco Unified Border Element (SP Edition) to map SIP packets with Resource-Priority header strings to the following SBC priority values:

- Routine
- Priority
- Immediate
- Flash
- Flash override
- Critical

The Call Admission Control (CAC) uses the assigned priority value to choose the QoS profile.

The following task configures Cisco Unified Border Element (SP Edition) to assign priority value"flash" to a SIP packet with Resource-Priority header string "dsn.flash."

- 1. configure terminal
- **2. sbc** *service name*
- 3. sbe
- 4. resource-priority-set name
- 5. resource-priority string value
- **6. priority** *priority-value*

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of an SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	resource-priority-set name	Enters the mode to map SIP Resource-Priority header string to SBC priority values.
	Example: Router(config-sbc-sbe)# resource-priority-set dsn	
Step 5	resource-priority string value	Enters the mode to configure the priority of the Resource-Priority header string.
	Example: Router(config-sbc-sbe-rsrc-pri-set)# resource-priority dsn.flash	
Step 6	priority priority-value	Sets the SBC priority value of the Resource-Priority header string.
	Example: Router(config-sbc-sbe-rsrc-pri)# priority flash	The SBC priority value must be one of the following:routine
		• priority
		• immediate
		• flash
		• flash-override
		• critical

Configuring a Resource Priority Set on a SIP Adjacency

The following task configures the SIP adjacency "SipToIsp42" to use resource-priority-set "dsn."

- 1. configure terminal
- 2. sbc service name
- 3. sbe
- 4. adjacency sip *adjacency-name*

5. resource-priority-set name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the SBC.
Step 3	sbe	Enters the mode of an SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures the SIP adjacency that is to be used with the specified resource priority set.
	Example: Router(config-sbc-sbe)# adjacency sip SipToIsp42	
Step 5	resource-priority-set name	Sets the SIP adjacency that is to be used with the specified resource priority set.
	Example: Router(config-sbc-sbe-adj-sip)# resource-priority-set dsn	

Selecting a QoS Profile Using CAC

This task configures calls from the account *cisco* to use the voice QoS profile *enterprise* for packets sent from Cisco Unified Border Element (SP Edition) to the original caller.

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This command can only be run at the per-call scope. The CAC policy does not get activated if this command is run at any other scope.

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name

- 6. first-cac-table table-name
- 7. cac-table table-name
- 8. table-type *limit* list of limit tables
- 9. entry entry-id
- 10. match-value key
- **11.** caller-voice-qos-profile profile-name
- **12.** caller-video-qos-profile profile-name
- **13.** caller-sig-qos-profile profile name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of Call Admission Control (CAC) policy set configuration within an SBE entity, creating a new
	Example: Router(config-sbc-sbe)# cac-policy-set 1	policy set, if necessary.

	Command or Action	Purpose
Step 5	first-cac-scope scope-name	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)#	The <i>scope-name</i> argument configures the scope at which limits should be initially defined. Possible values are:
	first-cac-scope call	• adj-group
		• call
		• dst-account
		• dst-adj-group
		• dst-adjacency
		• dst-number
		• global
		• src-account
		• src-adj-group
		• arc-adjacency
Step 6	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table MyCacTable	
Step 7	cac-table table-name	Enters the mode for configuration of an admission control table (creating one, if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table MyCacTable	SBE policy set.

	Command or Action	Purpose
Step 8	table-type limit list of limit tables	Configures a CAC Limit table-type within the context of an SBE policy set.
	Example:	<i>list of limit tables</i> can be one of the following values:
	Router(config-sbc-sbe-cacpolicy-cactable)#	• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all—No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
Step 9	entry entry-id	Enters the mode for configuring an entry in an admission control table, creating the entry, if necessary.
	Example: Router(config-sbc-sbe-cacpolicy- cactable)# entry 1	
Step 10	match-value key	Configures the match value of an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cac-table-ent)# match-value cisco</pre>	

	Command or Action	Purpose
Step 11	caller-voice-qos-profile profile-name	Configures the QoS profile to use for voice media packets sent to the original caller.
	<pre>Example: Router(config-sbc-sbe-cacpolicy- cac-table-ent)# caller-voice-qos-profile enterprise</pre>	
Step 12	caller-video-qos-profile profile-name	Configures the QoS profile to use for packets sent to the original caller.
	Example:	
	Router(config-sbc-sbe-cacpolicy- cac-table-ent)# caller-video-qos-profile enterprise	
Step 13	caller-sig-qos-profile profile-name	Configures the QoS profile to use for signaling packets sent to the original caller.
	Example:	
	Router(config-sbc-sbe-cacpolicy-	
	cac-table-ent)# caller-sig-qos-profile enterprise	

Implementing QoS Demarcation

In the context of a network, a QoS demarcation point is a transit point within the network that provides features for measuring call quality and fixing problems that affect call quality. You can configure the SBC as a QoS demarcation point to meet the following objectives:

- Generate an alert when a problem in the network affects call quality.
- Provide information that can be used to determine the location of the problem.
- Calculate statistics that can assist in diagnosing and fixing the problem.

The quality of a group of calls (or media streams processed by the SBC) can be determined by measuring parameters such as the following along the packet data path:

- Media packets lost while in transit from the sender to the receiver
- · Media packets dropped from the set of media packets received
- Jitter in the media packets received
- Network latency in the media streams

Using these measurements, the SBC can calculate the following QoS statistics:

• Average local media packet jitter

See RFC 3550 for the definition of local media packet jitter.

• Average remote media packet jitter

See RFC 3550 for the definition of remote media packet jitter.

- Average round trip delay
- Mean Opinion Score for Conversational Quality, Estimated (MOS-CQE) score

The MOS-CQE score provides an overall view of the statistics listed earlier. See Recommendations G.107 and G.113 of the ITU-T for the definition of the MOS-CQE score.



The International Telecommunication Union (ITU) coordinates and assists in the development of telecommunications standards. The ITU Telecommunication Standardization Sector (ITU-T) is a division of the ITU. Recommendations G.107 and G.113 that are published by the ITU-T explain the MOS-CQE score and the method for calculating it. For more information about these recommendations, visit the ITU-T website at http://www.itu.int/ITU-T/index.html.

Of the various factors defined in Recommendations G.107 and G. 113 for calculation of the MOS-CQE score, you can specify values for the following factors:

- Advantage (A) factor, which is specified at the per-adjacency level
- Equipment Impairment (Ie) factor, which is specified at the per-codec level
- Packet-Loss Robustness (Bpl) factor, which is specified at the per-codec level

Detailed information about these factors is available on the ITU-T website.

• Ratio of unanswered calls to the total number of calls

Note

The ratio of unanswered calls is not based on the measurements listed earlier.

- Ratio of media packets that are lost to the total number of media packets sent
- Ratio of media packets that are dropped to the total number of media packets received



Stored QoS statistics data is lost after an RP failover.

For each statistic, you can configure a combination of the following alerts to denote the state of the statistic:

- Critical
- Major

• Minor

For each alert, you specify a minimum (low) value and a maximum (upper) value. For statistics for which a higher value signifies an adverse effect on call quality, the alert changes when the upper limit of the earlier alert level is crossed. The following example illustrates how this works:

You specify the following alert levels for the Local Media Packet Jitter statistic:

- Major Low alert level: 60
- Major Upper alert level: 70
- Critical Low alert level: 71
- Critical Upper alert level: 80

A higher value of local media packet jitter indicates an adverse effect on call quality. While this statistic is being monitored, if the value of the statistic is increasing and crosses 80, the alert level changes to Critical. In contrast, if the value is decreasing and crosses 60, the alert changes to Normal. If the value is decreasing but does not go below 60, the alert level remains at Major.

<u>Note</u>

You can configure the SBC to generate an SNMP trap in response to changes in alert levels.

The reverse is true for the MOS-CQE score. If the alert levels listed earlier are specified for the MOS-CQE score, a MOS-CQE score higher than the specified Minor Upper alert level is classified as Normal. A value lower than the Critical Low alert level signifies that the statistic is in the Critical state.

You can specify the following time intervals at which a statistic must be measured. Because these are moving-average statistics, their values do not change suddenly over the boundaries of the time interval that you specify.

- Current 5 minutes—Statistics for the current 5-minute interval
- Current 15 minutes—Statistics for the current 15-minute interval
- Current hour—Statistics for the current 60-minute interval
- Current day—Statistics for the current day, starting from midnight
- Indefinitely—Statistics for the period starting from the last explicit reset

The following sections describe the procedures to configure the SBC for calculating the QoS statistics. Note that stored QoS statistics data is lost after an RP failover.

- Configuring the Calculation of the Local Jitter Ratio, page 38-12
- Configuring the G.107 Factors, page 38-15
- Configuring the Calculation of the MOS-CQE Score, page 38-18
- Configuring Alert Levels for the QoS Statistics, page 38-20
- Configuring SNMP Notifications for the QoS Statistics, page 38-23

Configuring the Calculation of the Local Jitter Ratio

Prior to Cisco IOS XE Release 3.3S, the Media Packet Forwarder (MPF) used the RTCP Sender Reports (SR) and Receiver Reports (RR) exchanged between a caller and a callee, and performed its own measurements on the media stream. QoS-related information was then passed to the media stream. Local jitter was not calculated because it requires tracking of the packet inter-arrival time, which is a processor-intensive operation in the MPF.

From Cisco IOS XE Release 3.3S, the SBC can be configured to calculate local jitter by tracking the percentage of calls that match criteria such as the source adjacency or destination adjacency. Local jitter is calculated according to the method specified in RFC 3550. The calculation is performed for both RTP streams and SRTP streams.

This task explains how to specify the percentage of calls for which the SBC must calculate the local jitter ratio. This task is one of the prerequisites for calculation of the MOS-CQE score.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency { sip | h323 } adjacency-name
- 5. local-jitter-ratio call-percentage
- 6. end
- 7. show sbc sbc-name sbe adjacencies adjacency-name detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example:	
	Router(config) # sbc mySbc	
Step 3	sbe	Enters the SBE entity mode within the SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Specifies whether you want to configure an SBE SIP adjacency mode or an SBE H.323 adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency h323 adj1	• <i>adjacency-name</i> —Specifies the name of the SIP adjacency or H.323 adjacency.

	Command or Action	Purpose
Step 5	local-jitter-ratio call-percentage	Specifies the percentage of calls that must be used to calculate the local jitter ratio.
	Example: Router(config-sbc-sbe-adj-h323)# local-jitter-ratio 205	• <i>call-percentage</i> —Specifies the percentage of calls. The value is expressed as an integer in the range from 0 to 1000. For example, if you enter 205 as the value of <i>call-percentage</i> , the SBC uses 20.5 percent of the calls for measuring local jitter.
		The default value is 0 because jitter determination is a performance drain on the MPF. When the value is 0, the jitter ratio and MOS-CQE are not calculated for the adjacency.
Step 6	end	Exits the SBE H.323 adjacency mode, and enters the privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-h323)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays details of the specified adjacency. The output also includes the local jitter ratio.
	Example: Router# show sbc mySbc sbe adjacencies h323adj detail	

The following example displays details of an adjacency using the **show sbc sbe adjacencies detail** command. The output also includes the *call-percentage* parameter value.

Router# show sbc mySbc sbe adjacencies adj1 detail

```
SBC Service "mySbc"
 Adjacency adj1 (H.323)
   Status:
                               Attached
   Signaling address: 1.0.0.3:1720 (default)
   Signaling-peer: 40.40.40.4:1720 (default)
   Admin Domain: None
   Account:
   Media passthrough: Yes
   Group:
   Hunting triggers: Global Triggers
   Hunting mode: Global Mode
   Technology Prefix:
   H245 Tunnelling: Enabled
   Fast-Slow Interworking: None
   Trust-level: Untrusted
   Call-security: Insecure
   Realm: None
   Warrant Match-Order: None
   Local Jitter Ratio: 205/1000
   Calc Moscqe: 0/1000
   G107A factor: 0
   H225 address block: Disabled (default)
   H225 address usage: h323id (default)
```

Configuring the G.107 Factors

The Advantage (A) factor, Equipment Impairment (Ie) factor, and Packet-Loss Robustness (Bpl) factor are used in the calculation of the MOS-CQE score. From Cisco IOS XE Release 3.4S, you can specify values for these factors.

This task explains how to configure the Advantage factor, Equipment Impairment factor, and Packet-Loss Robustness factor.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency { sip | h323 } adjacency-name
- 5. g107a-factor factor-number
- 6. exit
- 7. codec system sys-codec id payload-id
- 8. g107 ie factor-number
- 9. g107 bpl factor-number
- 10. end
- 11. show sbc sbc-name sbe adjacencies adjacency-name detail
- **12**. **show sbc** *sbc-name* **sbe codecs name** *codec-name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the SBE entity mode within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Specifies whether you want to configure an SBE SIP adjacency mode or an SBE H.323 adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency h323 adj1	• <i>adjacency-name</i> —Name of the SIP adjacency or H.323 adjacency.

	Command or Action	Purpose
Step 5	g107a-factor factor-number	Sets the Advantage factor.
	Example: Router(config-sbc-sbe-adj-h323)# g107a-factor 10	 <i>factor-number</i>—Value of the Advantage factor. The range is from 0 to 20. The default value is 0. See Recommendation G.107 for information about the significance of this default value.
Step 6	exit	Exits the SBE H.323 adjacency mode or the SBE SIP adjacency mode, and enters the SBE entity mode.
	Example: Router(config-sbc-sbe)# exit	
Step 7	codec system sys-codec id payload-id	Enters the codec definition mode.
	Example: Router(config-sbc-sbe)# codec system PCMU id 0	
Step 8	g107 ie factor-number	Sets the Equipment Impairment factor.
	Example: Router(config-sbc-sbe-codec-def)# g107 ie 20	• <i>factor-number</i> —Value of the Equipment Impairment factor. The range is from 0 to 50.
		See Appendix I of Recommendation G.113 for information about the values that you can set for various codecs. If you have a custom codec, you can set a value that best matches the impairment introduced by the codec.
		The default value is 0. See Appendix I of Recommendation G.113 for information about the significance of this default.
Step 9	g107 bp1 factor-number	Sets the Packet-Loss Robustness factor.
	Example:	• <i>factor-number</i> —Specifies the value of the Packet-Loss Robustness factor, which can range from 1 to 40.
	Router(config-sbc-sbe-codec-def)# g107 bpl 30	See Appendix I of Recommendation G.113 for information about the values that you can set for various codecs. If you have a custom codec, you can set a value that best matches the Packet-Loss Robustness factor for the codec.
		The default value is 1. See Appendix I of Recommendation G.113 for information about the significance of this default.
Step 10	end	Exits the codec definition mode, and enters the privileged EXEC mode.
	Example: Router(config-sbc-sbe-adj-h323)# end	

	Command or Action	Purpose
Step 11	show sbc sbc-name sbe adjacencies adjacency-name detail	Displays details of the specified adjacency. The output includes the value set for the Advantage factor.
	Example: Router# show sbc mySbc sbe adjacencies h323adj detail	
Step 12	show sbc sbc-name sbe codecs name codec-name Example:	Displays details of the specified codec. The output includes the Equipment Impairment factor and Packet-Loss Robustness factor values.
	Router#show sbc Mysbc sbe codecs name PCMU	

The following example displays the details of a specified adjacency using the **show sbc sbe adjacencies detail** command. The output includes the Advantage factor value.

Router# show sbc mySbc sbe adjacencies adj1 detail

```
SBC Service "mySbc"
 Adjacency adj1 (H.323)
    Status: Attached
    Signaling address: 1.0.0.3:1720 (default)
    Signaling-peer: 40.40.40.4:1720 (default)
   Admin Domain: None
   Account:
   Media passthrough: Yes
   Group:
   Hunting triggers: Global Triggers
   Hunting mode: Global Mode
   Technology Prefix:
   H245 Tunnelling: Enabled
    Fast-Slow Interworking: None
   Trust-level: Untrusted
   Call-security: Insecure
   Realm: None
   Warrant Match-Order: None
   Local Jitter Ratio: 1000/1000
   Calc Moscqe: 305/1000
   G107A factor: 10
   H225 address block: Disabled (default)
   H225 address usage: h323id (default)
```

The following example displays the details of a specified codec using the **show sbc sbe codecs name codec** command. The output includes the Equipment Impairment factor and Packet-Loss Robustness factor values.

Router# show sbc mySbc sbe codecs name PCMU

codec name	=	PCMU
static_payload_id	=	0
codec_type	=	sample
clock_rate	=	8000
packet_time	=	10
bandwidth	=	64000
sample_size	=	8
num_channels	=	1
max_fpp	=	20
media_type	=	audio
g107 bpl	=	40
g107 ie	=	50

options

= transcode, inband-dtmf

Configuring the Calculation of the MOS-CQE Score

This section describes the procedure to configure a target MOS-CQE score.

The following are prerequisites for calculation of the MOS-CQE score:

- Performing the procedure described in the ?\$paranum>Configuring the Calculation of the Local Jitter Ratio? section on page 38-12. Note that it is optional to configure caculation of the local jitter ratio. If you do not perform the procedure, the default value set for the percentage of calls for which the local jitter is to be calculated is used in the calculation of the MOS-CQE score.
- Performing the procedure described in the ?\$paranum>Configuring the G.107 Factors? section on page 38-15. Note that it is optional to configure the G.107 factors. If you do not perform this procedure, the default values that are set for these factors are used to calculate the MOS-CQE score.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. calc-moscqe call-percentage
- 6. end
- 7. show sbc sbc-name sbe adjacencies adjacency-name detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the SBE entity mode within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency {sip h323} adjacency-name	Specifies whether you want to configure an SBE SIP adjacency mode or an SBE H.323 adjacency mode.
	Example: Router(config-sbc-sbe)# adjacency h323 adj1	• <i>adjacency-name</i> —Name of the SIP adjacency or H.323 adjacency.

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	Command or Action	Purpose
Step 5	calc-moscqe call-percentage	Specifies the percentage of calls that must be used to calculate the MOS-CQE score.
	Example: Router(config-sbc-sbe-adj-h323)# calc-moscqe 305	• <i>call-percentage</i> —Percentage of calls. The range is from 0 to 1000. For example, if you enter 305 as the value of <i>call-percentage</i> , the SBC uses 30.5 percent of the calls for calculating the MOS-CQE score.
		The default value is 0. When the value is 0, the MOS-CQE score is not calculated. Note that the MOS-CQE score also depends on the local jitter ratio. If you do not perform the procedure described in the ?\$paranum>Configuring the Calculation of the Local Jitter Ratio? section on page 38-12, the MOS-CQE score is not calculated.
Step 6	end	Exits the SBE SIP adjacency mode or SBE H.323 adjacency mode, and returns to the privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 7	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays details of the specified adjacency. The output also includes the value that is set for the <i>call-percentage</i> parameter.
	Example: Router# show sbc mySbc sbe adjacencies h323adj detail	 <i>sbc-name</i>—Name of the SBC. <i>adjacency-name</i>—Name of the adjacency.

The following example displays the details of a specified adjacency using the **show sbc sbe adjacencies detail** command. The output also includes the *call-percentage* parameter value.

```
Router# show sbc mySbc sbe adjacencies adj1 detail
```

```
SBC Service "mySbc"
 Adjacency adj1 (H.323)
   Status: Attached
   Signaling address: 1.0.0.3:1720 (default)
   Signaling-peer: 40.40.40.4:1720 (default)
   Admin Domain: None
   Account:
   Media passthrough: Yes
   Group:
   Hunting triggers: Global Triggers
   Hunting mode: Global Mode
   Technology Prefix:
   H245 Tunnelling: Enabled
   Fast-Slow Interworking: None
   Trust-level: Untrusted
   Call-security: Insecure
   Realm: None
   Warrant Match-Order: None
   Local Jitter Ratio: 1000/1000
   Calc Moscqe: 305/1000
   G107A factor: 0
   H225 address block: Disabled (default)
   H225 address usage: h323id (default)
```



You can display the MOS-CQE score calculated by the SBC by running the **show sbc sbe call-stats per-adjacency** command. A sample output of this command is provided later in this chapter.

Configuring Alert Levels for the QoS Statistics

This task explains how to configure alert levels for the QoS statistics.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. statistics {lcl-jit | mos-cqe | mpd-pct | mpl-pct | rmt-jit | rtd | ucr}
- 5. currenthour {adjacency adjacency-name {critical low value upper value | major low value upper value [critical low value upper value] | minor low value upper value [[critical low value upper value] | [major low value upper value [critical low value upper value]]]} | default {critical low value upper value | major low value upper value [critical low value upper value] | minor low value upper value [critical low value upper value] | minor low value upper value [critical low value upper value] | minor low value upper value [critical low value upper value] | minor low value upper value [critical low value upper value] | minor low value upper value] | minor low value upper value [critical low value upper value] | [major low value upper value]]]}



Depending on your requirement, you can use **current15mins**, **currentday**, or **currentindefinite** instead of **currenthour**. The time interval for which statistical data is monitored varies according to the command that you run. The syntax is the same for all these commands. The time intervals for which these commands are applicable are described at the start of the ?\$paranum>Implementing QoS Demarcation? section on page 38-10.

- 6. end
- 7. show sbc sbc-name sbe call-stats per-adjacency adjacency-name period

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode.
	Example: Router(config)# sbc mySbc	• soc-nume—ivanie of the SBC.

	Command or Action	Purpose
Step 3	sbe	Enters the SBE configuration mode.
	Example: Router(config-sbc)# sbe	
Step 4	statistics {lcl-jit mos-cqe mpd-pct mpl-pct rmt-jit rtd ucr}	Specifies the statistic for which you want to set alert levels.
		You can specify one of the following statistics:
	Example: Router(config-sbc-sbe)# statistics lcl-jit	• lcl-jit —Specifies the average local media packet jitter.
		• mos-cqe —Specifies the MOS-CQE score.
		• mpd-pct —Specifies the ratio of media packets that are dropped to the total number of media packets received.
		• mpl-pct —Specifies the ratio of media packets that are lost to the total number of media packets sent.
		• rmt-jit —Specifies the average remote media packet jitter.
		• rtd —Specifies the average round trip delay.
		• ucr —Specifies the ratio of unanswered calls to the total number of calls.

	Command or Action	Purpose	
Step 5	<pre>currenthour {adjacency adjacency-name {critical low value upper value major low value upper value [critical low value upper value] minor low value upper value [major low value upper value [critical low value upper value]] } default {critical low value upper value major low value upper value [critical low value upper value] minor low value upper value [major low value upper value [critical low value upper value]] }} Example: Router(config-sbc-sbe-stats)# currenthour default critical low 30 upper 50</pre>	Specifies that statistical data must be monitored for the next hour for the QoS statistic specified by the statistics command.	
		Note Depending on your requirement, you can use current15mins, currentday, or currentindefinite instead of currenthour. The time interval for which statistical data is monitored varies according to the command that you run. The syntax is the same for all these commands. The time intervals for these commands is described at the start of the ?\$paranum>Implementing QoS Demarcation? section on page 38-10.	
		• adjacency —Specifies that alert levels must be set for the specified adjacency.	
		• <i>adjacency-name</i> —Name of the adjacency.	
		• critical —Specifies the lower and upper limits for the Critical alert level.	
		• low —Specifies the lower limit for the alert level.	
		• <i>value</i> —Value of the lower limit or upper limit.	
		• upper —Specifies the upper limit for the alert level.	
		• major —Specifies the lower limit and upper limit for the Major alert level.	
		• minor —Specifies the lower limit and upper limit for the Minor alert level.	
		• default —Specifies that alert levels must be set for all adjacencies on the SBC.	
Step 6	end	Exits the SBE entity mode, and enters the privileged EXEC mode.	
	Example: Router(config-sbc-sbe-adj-h323)# end		
Step 7	<pre>show sbc sbc-name sbe call-stats per-adjacency adjacency-name period</pre>	 Displays QoS statistics for the specified adjacency. <i>sbc-name</i>—Name of the SBC. <i>adjacency-name</i>—Name of the adjacency. 	
	Example: Router# show sbc mySbc sbe call-stats per-adjacency adj1 currentindefinite	• <i>period</i> —Interval for which the statistics must be displayed.	

The following example displays the call statistics pertaining to the adjacency (since the last explicit reset) using the **show sbc sbe call-stats per-adjacency** command. The output also includes QoS statistics.

Router# show sbc Mysbc sbe call-stats per-adjacency adj1 currentindefinite

Statistics for the current hour for adjacency adj1

Stats Reset Timestamp:

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```
Timestamp when stats for this summary period were reset =
                                                            2011/04/08 04:05:09
Current count of Media Packets Lost =
                                                            0
Current count of Media Packets Dropped =
                                                            1
Current count of Media Packets Sent =
                                                            116
                                                            116
Current count of Media Packets Received =
Current count of RTCP Packets Sent =
                                                            Ο
                                                            0
Current count of RTCP Packets Received =
                                                            21 secs 16 ms
Average Call Duration =
Average of the Unanswered Call Ratio per thousand call =
                                                            0
Average of the Round Trip Delay =
                                                            0 ms
Average of the locally calculated jitter =
                                                            77 ms
Average of the remotely calculated jitter =
                                                            0 ms
Average of the received media dropped per thousand pkts =
                                                            8
Average of the sent media lost per thousand pkts =
                                                            0
Average of Mean Opinion Score =
                                                            20
Current alert level of the Unanswer Seizure Ratio =
                                                           NONE
Current alert level of the Round Trip Delay =
                                                           NORMAL
Current alert level of the locally calculated Jitter =
                                                            MINOR
Current alert level of the remotely calculated Jitter =
                                                            NORMAL
Current alert level of the media packet dropped =
                                                            MAJOR
Current alert level of the sent packets lost =
                                                           NORMAL
Current alert level of the Media Opinion Score =
                                                            MINOR
```

Configuring SNMP Notifications for the QoS Statistics

To configure SNMP notifications for the QoS statistics, perform the procedure described in the ?\$paranum>Configuring SNMP Notifications? section on page 5-3 for running the following command:

snmp-server enable traps sbc qos-statistics

Configuration Examples of QoS Profiles

This section provides the following configuration examples:

- Configuring a QoS Voice Profile Using IP Precedence Marking: Example
- Configuring a QoS Voice Profile Using DSCP Marking: Example
- Choosing a QoS Profile Using CAC: Example
- Configuring a SIP Adjacency Using a Resource-Priority Set: Example

Configuring a QoS Voice Profile Using IP Precedence Marking: Example

This task configures a QoS voice profile to use an IP precedence value of 1 and a ToS value of 12 to mark packets that match the QoS.

```
configure
sbc mysbc
sbe
qos voice residential
marking ip-precedence
ip precedence 1
ip tos 12
```

Configuring a QoS Voice Profile Using DSCP Marking: Example

This task configures a QoS voice profile to mark packets with a DSCP value of 10.

```
configure
sbc mysbc
sbe
qos voice residential
marking dscp
dscp 10
```

Choosing a QoS Profile Using CAC: Example

This task configures calls from the account "cisco" to use the voice QoS profile "enterprise" for packets sent from Cisco Unified Border Element (SP Edition) to the original caller.

```
configure
 sbc mysbc
  sbe
   cac-policy-set 1
    first-cac-scope call
     first-cac-table MyCacTable
      cac-table MyCacTable
      table-type limit src-account
      entry 1
      match-value cisco
      caller-voice-qos-profile enterprise
      caller-video-gos-profile enterprise
sbc mysbc
sbe
  cac-policy-set 1
   first-cac-scope call
   first-cac-table MyCacTable
   cac-table MyCacTable
    table-type limit src-account
    entry 1
    match-value cisco
    caller-video-gos-profile enterprise
     caller-voice-gos-profile enterprise
   1
  1
 !
```

Configuring a SIP Adjacency Using a Resource-Priority Set: Example

The following example shows how to configure a SIP adjacency using a resource-priority set:

```
configure
sbc mysbc
sbe
adjacency sip SipToIsp42
resource-priority-set dsn
```



Implementing Transcoding

Transcoding is the process of translating a media stream encoded using one codec into a media stream encoded using another codec. For example, translating a media stream encoded as Pulse Code Modulation u-law (PCMU) into one encoded as ITU-T G.726-32.

The primary reason for transcoding configurations is to configure the capabilities of external media transcoding devices when these devices cannot be discovered automatically. In-band auto discovery of transcoder capabilities is currently not supported. Therefore, this step must be done when configuring all connections to all current remote transcoding devices.

Note

Transcoding configurations can be skipped altogether if the described reason does not apply.

Media gateways are allowed to connect whether or not configuration has been supplied for them. To help avoid configuration errors, the signaling border element (SBE) logs a warning if an incoming connection is received from a media gateway that is not a data border element (DBE) and does not have transcoding configured.



The Transcoding feature is supported in the unified model for Cisco IOS XE Release 2.5 and later.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Implementing SBC Transcoding

Release	Modification
Cisco IOS XE Release 2.5	This feature was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.3S	The Voice Transcoding Per Adjacency Statistics feature was added to the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 3.11S	The Blended Transcoding feature was added to the Cisco ASR 1000 Series Aggregation Services Routers.

Γ

Contents

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- Prerequisites for Implementing Transcoding, page 40-2
- Restrictions for Implementing Transcoding, page 40-2
- Restrictions for Media Gateway-Assisted DMTF Interworking, page 40-3
- Information About Transcoding, page 40-3
- Configuration Examples for Implementing Transcoding, page 40-13
- Verification, page 40-15
- Voice Transcoding Per Adjacency Statistics, page 40-16
- Configuring the Voice Transcoding Per Adjacency Statistics, page 40-16
- Media Gateway-Assisted DTMF Interworking, page 40-19
- Blended Transcoding, page 40-27

Prerequisites for Implementing Transcoding

The following prerequisites are required to implement SBC transcoding:

- Before implementing these features, Cisco Unified Border Element (SP Edition) must already be configured.
- All SBE and DBE configurations required to make simple calls must already be configured. Transcoding configurations follow these configurations.

Restrictions for Implementing Transcoding

The following are restrictions of the Implementing Transcoding feature:

- The H.323 fast-start calls will be dropped to slow-start procedure if transcoding is required. This can be achieved by the callee side rejecting the fast-start request.
- No transcoding support for H.323 to SIP interworked calls.
- No transcoding support for H.323 to H.323 interworked calls.
- The only codecs supported for H.323 transcoding are G.711 (PCMU and PCMA) and G.729 (with and without annex B).
- When audio transcoding is in operation, the SBC does not support sending and receiving RFC 2833 in-band packets to and from the SBC and interworking RFC 2833 packets with out-of-band SIP INFO or SIP NOTIFY Relay messages on the other call leg.

The following are DTMF interworking restrictions when transcoding is used:

• Signaling and media DTMF interworking is not supported when transcoding is performed on the call
When audio transcoding is in operation, the SBC does not support sending and receiving RFC 2833 in-band packets to and from the SBC and interworking RFC 2833 packets with out-of-band SIP INFO or SIP NOTIFY Relay messages on the other call leg.

Restrictions for Media Gateway-Assisted DMTF Interworking

Following are the restrictions of the Media Gateway-Assisted DMTF interworking feature:

- The SBC supports the use of transcoders, such as a Cisco MGX 8880, only in SIP-SIP calls. DTMF interworking with transcoders are not supported for H.323 calls.
- The SBC cannot interwork DTMF with transcoders that cannot pass through DTMF.
- When a Cisco MGX 8880 is not used as transcoders, only SIP-SIP calls are supported.

Information About Transcoding

Transcoding is the process of translating a media stream encoded using one codec into a media stream encoded using another codec. For example, translating a media stream encoded as PCMU into one encoded as G.726-32.

Transcoding is supported using external digital signal processor (DSP) hardware. A Cisco MGX 8880 Media Gateway can be used to provide the transcoding function for one or more SBCs.

The SBC supports two types of transcoding:

- Transcoding After Rejection, page 40-3
- Codec Filtering, page 40-6
- Configuring Transcoding After Rejection, page 40-7
- Configuring Codec Filtering Transcoding, page 40-10

Transcoding After Rejection

The SBC automatically brings the transcoding device into use for any call requiring transcoding between these codecs, as long as the Call Admission Control (CAC) policy configuration does not preclude the transcoder service from being supplied for the call. When a call that requires transcoding is set up, the SBE goes through the following steps:

- Receives an initial signaling request from the calling endpoint. This triggers the SBC to perform initial call setup on the incoming and outgoing local media termination points. The SBC then forwards the set up request towards the callee.
- Receives a response from the called endpoint that indicates that none of the codecs in the initial request are acceptable. These responses include:
 - 415—Unsupported media type (SIP)
 - 488—Not acceptable here (SIP)
 - Failure to identify common codec during Terminal Capability Exchange procedure of H.245 protocol.

This triggers the SBC to bring a transcoder into the call that is inserted in the media path between the incoming and outgoing local media terminations. A new request is sent to the called endpoint, indicating the new codec type generated by the transcoder.

• SBE may then have to iterate through the list of codecs the transcoder supports until it finds one that is acceptable to the called endpoint. When this is done, the call is connected and media transmission begins.

Figure 40-1 shows where the transcoder sits in the network, and the path taken by the media in a transcoded call.

Figure 40-1 Transcoding Configuration





Although Figure 40-1 shows two DBEs, transcoding is possible with a single DBE. With a single DBE, the media flows through the DBE twice, once on its way from the sending endpoint to the transcoder and a second time as it flows from the transcoder to the receiving endpoint.

For the Session Border Controller (SBC) to program the transcoder, it must be registered. The transcoding device acts as an H.248 media gateway, so it needs to be configured with the IP address and port of the SBE or SBC to connect to. The SBE or SBC acts as an H.248 Media Gateway Controller. See the documentation for your transcoder device for notes on how to do this. The documentation for the Cisco MGX 8880 Media Gateway can be found at:

http://www.cisco.com/en/US/docs/switches/wan/mgx/software/mgx_r5.0/data/configuration/guide/scg. html

In addition, the SBE must have the following specific configuration:

- An H.248 control address and port must be configured (using the **sbe control address ipv4** and **sbe control address h248 port** commands). By default, this is on port 2944, and it is the address and port to which the transcoder must connect.
- An explicit media gateway needs to be configured (using the **sbe media-gateway ipv4** command). The explicit media gateway must have its list of supported codecs defined so that the SBC knows which codecs the transcoder can translate between, and it must be identified as a transcoder (using the **sbe media-gateway ipv4 codecs** and **sbe media-gateway ipv4 transcoder** commands).
- The **show sbc sbe media-gateway-associations** command can be used to check that the transcoder has correctly registered with the SBE. If this has happened, the transcoder should appear in the list of known media gateways with an active association.

For configuration step information, see the ?\$paranum>Configuring Transcoding After Rejection? section on page 40-7.

Troubleshooting Tip for Media-Timeout Transcoded Call Using a VXSM card

A Cisco MGX 8880 equipped with one or more Cisco Voice Switch Service Module (VXSM) card sets can operate as a media gateway. In a network where the SBC uses the Cisco MGX 8880 as a transcoding device to act as an H.248 media gateway, some additional configuration is required on the VXSM card for media-timeout to work properly in a transcoded call.

The following additional steps need to be configured on the VXSM card in the Cisco MGX 8880 Media Gateway:

- **Step 1** Enable the RTCP control with the following command. InteropMGX.4.VXSM.a > cnfdspparam -control 1
- **Step 2** Set the RTCP timer control to startRtpOrRtcpPktRcvd with the following command:

```
InteropMGX.4.VXSM.a > cnfdspparam -rtcptm 3
```

Step 3 Verify that the settings are correct using the following command to show a list of DSP parameters:

```
InteropMGX.4.VXSM.a > dspdspparam
```

_____ List DSP Parameters _____ SID Payload Type decimal : <=== RTCP control enabled RTCP Control : true RTCP Interval(milliseconds) : 5000 RTCP Interval Multiplier : 5 VAD Adaptive : false G.711 PLC : none DTMF Power Level (0.1 dBm) : -120 DTMF Power Twist (0.1 dB) : 0 startRtpOrRtcpPktRcvd <=Timer Control properly set</pre> RTCP Timer Control : VOM Control disable : RTCPXR Control : enable RTCPXR Report Frequency : 1 VQM Default Minimum Gap 16 : RTCPXR external R factor 127 : 50 SES Threshold (ms) : Voice IPIP mode normal :

For more information on the VXSM card, see the "VXSM as a Transcoding Gateway" chapter in the *Cisco Voice Switch Service Module (VXSM) Configuration Guide Release 5.5* at: http://www.cisco.com/en/US/docs/switches/wan/mgx/software/mgx_r5.5/voice/vxsm/configuration/guide/config5.html

Codec Filtering

The SBC allows you to restrict which codecs a particular call, caller and callee are allowed to use by whitelisting certain codecs. Initially all recognized codecs are on the whitelist. If a codec is requested which is absent from the call, caller, or callee codec whitelist, then the call still proceeds, but the forbidden codecs are removed from the offer and media gate configuration.

By supporting caller and callee codec lists, the SBC is able to make more intelligent transcoding decisions. If the codec support of either the calling or the called endpoint is known, then setting the caller and/or callee lists in a CAC policy is appropriate. However it may be that other considerations, such as the source adjacency, will affect the codec decision, in which case the per-call codec list can still be used.

For example, if the caller and callee codec lists are set to 'A and B', then all calls would use codec A. However, if a call had come across a transit network X (as indicated by the source adjacency) that only supported codec B, then the user could have an extra policy matching on source adjacency X, setting the per-call codec list to B. Calls crossing network X would then be forced to use codec B.

You can also limit the minimum packetization period of each codec, by configuring a value for the lowest acceptable minimum packetization period for each permitted codec. If a session is requested with a packetization period below this limit, the call still proceeds, but SBC increases the packetization period to the configured minimum value.

For configuration step information, see the ?\$paranum>Configuring Codec Filtering Transcoding? section on page 40-10.

Configuring Transcoding After Rejection

In this configuration area, the user supplies a configuration for a list of remote media gateways that may need to be managed by the SBE. This is not required when transcoding is not needed.

The primary reason for transcoding configurations is to configure the capabilities of external media transcoding devices when these devices cannot be discovered automatically. In-band auto-discovery of transcoder capabilities is currently not supported. Therefore, this step must be done when configuring all connections to all current remote transcoding devices.

Note

Transcoding configurations can be skipped if the described reason does not apply.

By default, media gateways are allowed to connect whether or not configuration has been supplied for them. To help avoid configuration errors, the SBE logs a warning if an incoming connection is received from a media gateway that is not a DBE and does not have transcoding configured.

The basic steps for implementing transcoding are as follows:

- 1. Configure the IP address, port, and transport protocol for H.248 media gateway controller on SBC. This step may not be required if the Media Gateway Controller has already been configured.
- 2. Configure the media gateway IP address.
- **3.** Configure the codecs to be transcoded (for example, between ITU-T G.711 ulaw and ITU-T G.729A).
- 4. Specify the media gateway as a transcoder.
- 5. Activate SBE.

This task implements transcoding for SBC.

Once configured, the SBC automatically brings the transcoding device into use for any call requiring transcoding between the codecs as long as the call admission control (CAC) policy configuration does not preclude the transcoder service from being supplied for the call using the **transcode-deny** command (See the Configuring Call Admission Control Policy Sets, CAC Tables, and Global CAC Policy Sets section in the ?\$paratext[CT_ChapTitle]>? module).



In an H.323 adjacency configuration, you must use the **h245-tunnel disable** command for H.323 FastStart transcoded calls.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. control address h248 index index-number
- 5. port port-number
- 6. ipv4 ipv4_IP_address
- 7. transport [transport-type]
- 8. exit
- 9. media-gateway ipv4 IPv4-IP-address
- **10.** codecs codec-list
- 11. transcoder
- 12. exit
- 13. activate
- 14. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	control address h248 index index-number	Configures an SBE to use a given IPv4 H.248 control address
	Example: Router(config-sbc-sbe)# control address h248 index 0	

	Command or Action	Purpose
Step 5	port port-number	Configures an SBE to use a given IPv4 H.248 port for H.248 communications.
	Example: Router(config-sbc-sbe-ctrl-h248)# port 2000	
Step 6	ipv4 ipv4-IP-address	Configures an SBE to use a given IPv4 H.248 control address.
	Example: Router(config-sbc-sbe-ctrl-h248)# ipv4 1.1.1.1	
Step 7	transport [transport-type]	Configures transport type for H.248 communications.
	<pre>Example: Router(config-sbc-sbe-ctrl-h248)# transport udp</pre>	
Step 8	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-ctrl-h248)# exit	
Step 9	media-gateway ipv4 IPv4-IP-address	Configures a media gateway.
	Example: Router(config-sbc-sbe)# media-gateway ipv4 10.0.0.1	
Step 10	codecs codec-list	Configures the codecs supported by the media gateway. Enters into media gateway codecs configuration mode.
	<pre>Example: Router(config-sbc-sbe-mg)# codecs m=audio 1234 RTP/AVP 0 18,a=rtpmap:0 PCMU/8000,a=rtpmap:18 G729A/8000</pre>	
Step 11	transcoder	Configures the media gateway with transcoder support.
	Example: Router(config-sbc-sbe-mg-codecs)# transcoder	
Step 12	exit	Exits media gateway codecs configuration mode to the sbe command mode level.
	Example: Router(config-sbc-sbe-mg-codecs)# exit	
Step 13	activate	Initiates the SBC service after all SBE address configuration has been successfully committed.
	Example: Router(config-sbc-sbe-mg)# activate	
Step 14	end	Ends the configuration session.
	Example: Router(config-sbc)# end	

Configuring Codec Filtering Transcoding

Configure codec filtering transcoding as shown below.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. media-gateway ipv4 IPv4_IP_address
- 5. codecs codec-list
- 6. transcoder
- 7. exit
- 8. cac-policy-set
- 9. first-cac-table
- 10. cac-table
- 11. table-type policy-set
- 12. entry entry-num
- 13. caller-codec-list list-name
- 14. exit
- 15. exit
- 16. exit
- 17. codec-list list-name
- **18.** codec codec-name
- 19. exit
- 20. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.

	Command or Action	Purpose
Step 3	sbe	Enters the mode of an SBE entity within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	media-gateway ipv4 IPv4-IP-address	Configures a media gateway.
	<pre>Example: Router(config-sbc-sbe)# media-gateway ipv4 10.0.0.1</pre>	
Step 5	codecs codec-list	Configures the codecs supported by the media gateway.
	Example: Router(config-sbc-sbe-mg)# codecs m=audio 1234 RTP/AVP 0 18,a=rtpmap:0 PCMU/8000,a=rtpmap:18 G729A/8000	
Step 6	transcoder	Configures the media gateway with transcoder support.
Step 7	Example: Router(config-sbc-sbe-mg-codecs)# transcoder exit	Exits the media gateway configuration mode.
	Example: Router(config-sbc-sbe-mg-codecs)# exit	
Step 8	cac-policy-set	Enters the CAC policy submode.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 9	first-cac-table	Creates or configures the first admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table 1</pre>	
Step 10	cac-table	Creates or configures an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table 1	
Step 11	table-type policy-set	Configures the Policy Set table type of Call Admission Control (CAC) table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	
Step 12	entry	Creates or modifies an entry in a table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 13	caller-codec-list list-name	Lists the codecs which the caller leg of a call is allowed to use.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-codec-list my_codecs</pre>	
Step 14	exit	Exits the CAC table entry configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 15	exit	Exits the CAC table configuration mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 16	exit	Exits the CAC policy configuration mode.
	Example: Router(config-sbc-sbe-cacpolicy)# exit	
Step 17	<pre>codec-list list-name</pre>	Creates a codec list and enters the Codec list configuration mode
	Example: Router(config-sbc-sbe)# codec-list my_codecs	
Step 18	codec codec-name	Adds a codec to a codec list.
	Example: Router(config-sbc-sbe-codec-list)# codec PCMU	
Step 19	exit	Exits the Codec list configuration mode.
	Example: Router(config-sbc-sbe-codec-list)# exit	
Step 20	end	Ends the configuration session.
	Example: Router(config-sbc-sbe)# end	

Configuration Examples for Implementing Transcoding

The example below is a configuration of transcoding after rejection.

```
Router(config)# sbc mySbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# control address h248 index 1
Router(config-sbc-sbe-ctrl-h248)# port 2000
Router(config-sbc-sbe-ctrl-h248)# ipv4 88.88.133.2
Router(config-sbc-sbe-ctrl-h248)# transport udp
Router(config-sbc-sbe-ctrl-h248)# exit
Router(config-sbc-sbe)# media-gateway ipv4 10.0.0.1
Router(config-sbc-sbe-mg)# codecs m=audio 1234 RTP/AVP 0 18,a=rtpmap:0
PCMU/8000,a=rtpmap:18 G729A/8000
Router(config-sbc-sbe-mg-codecs)# transcoder
Router(config-sbc-sbe-mg-codecs)# exit
Router(config-sbc-sbe-mg-codecs)# exit
Router(config-sbc-sbe-mg)# activate
Router(config-sbc)# end
```

Below is an example of codec filtering transcoding.

```
Router(config)# ip route 10.0.20.33 255.255.255.255 10.130.10.33
Router(config)# ip route 0.0.0.0 0.0.0.0 10.74.50.114
Router(config)# ip route 0.0.0.0 0.0.0.0 10.130.10.1
```

Router(config)# snmp-server community cisco group Network-Monitor Router(config)# snmp-server community public group Network-Monitor Router(config)# snmp-server community private group Network-Monitor

```
Router# configure terminal
Router(config)# sbc sbc-11
Router(config-sbc)# sbe
Router(config-sbc-sbe)# media-gateway ipv4 10.100.181.2
Router(config-sbc-sbe-mg)# codecs m=audio 20000 RTP/AVP 0 8 18,a=rtpmap:0
PCMU/8000,a=rtpmap:8 PCMA/8000,a=rtpmap:18 G729/8000
Router(config-sbc-sbe-mg)# transcoder
```

```
Router(config-sbc-sbe)# control address h248 index 1
Router(config-sbc-sbe-ctrl-h248)# ipv4 10.130.10.4
Router(config-sbc-sbe-ctrl-h248)# transport udp
```

```
Router(config-sbc-sbe)# adjacency sip SIPP81
Router(config-sbc-sbe-adj-sip)# nat force-off
Router(config-sbc-sbe-adj-sip)# preferred-transport udp
Router(config-sbc-sbe-adj-sip)# redirect-mode pass-through
Router(config-sbe-adj-sip)# authentication nonce timeout 300
Router(config-sbe-adj-sip)# signaling-address ipv4 10.130.10.4
Router(config-sbc-sbe-adj-sip)# signaling-port 5060
Router(config-sbc-sbe-adj-sip)# remote-address ipv4 10.0.244.81 255.255.255.255
Router(config-sbc-sbe-adj-sip)# signaling-peer 10.0.244.81
Router(config-sbc-sbe-adj-sip)# signaling-peer 10.0.244.81
Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060
Router(config-sbc-sbe-adj-sip)# dbe-location-id 0
Router(config-sbc-sbe-adj-sip)# reg-min-expiry 3000
Router(config-sbc-sbe-adj-sip)# attach
Router(config-sbc-sbe)# adjacency sip SIPP91
```

```
Router(config-sbc-sbe-adj-sip)# nat force-off
Router(config-sbc-sbe-adj-sip)# preferred-transport udp
Router(config-sbc-sbe-adj-sip)# redirect-mode pass-through
Router(config-sbc-sbe-adj-sip)# authentication nonce timeout 300
Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 10.130.10.4
```

```
Router(config-sbc-sbe-adj-sip)# signaling-port 5060
Router(config-sbc-sbe-adj-sip)# remote-address ipv4 10.0.244.91 255.255.255.255
Router(config-sbc-sbe-adj-sip)# signaling-peer 10.0.244.91
Router(config-sbc-sbe-adj-sip)# signaling-peer-port 5060
Router(config-sbc-sbe-adj-sip)# dbe-location-id 0
Router(config-sbc-sbe-adj-sip)# reg-min-expiry 3000
Router(config-sbc-sbe-adj-sip)# attach
Router(config-sbc-sbe) # sip inherit profile preset-core
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-table table
Router(config-sbc-sbe-cacpolicy)# first-cac-scope call
Router(config-sbc-sbe-cacpolicy)# averaging-period 60
Router(config-sbc-sbe) # cac-table table
Router(config-sbc-sbe-cacpolicy-cactable)# match-type adjacency
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIPP81
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 64009 Gbps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-updates 4294967295
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-channels 4294967295
Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-type full-duplex
Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-timeout 0
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-codec-list allow711u
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-hold-setting standard
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-hold-setting standard
Router(config-sbc-sbe-cacpolicy-cactable)# entry 2
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIPP91
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 64009 Gbps
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-updates 4294967295
Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-channels 4294967295
Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-type full-duplex
Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-timeout 0
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-codec-list allowg729
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-hold-setting standard
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-hold-setting standard
Router(config-sbc-sbe-cacpolicy-cactable-entry)# complete
Router (config-sbc-sbe) # active-cac-policy-set 1
Router (config-sbc-sbe)# retry-limit 3
Router (config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# first-call-routing-table table
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency SIPP91
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 318X
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# prefix
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # complete
Router (config-sbc-sbe)# active-call-policy-set 1
Router(config-sbc-sbe)# sip max-connections 2
Router(config-sbc-sbe) # sip timer
Router(config-sbc-sbe-sip-tmr)# tcp-idle-timeout 120000
Router(config-sbc-sbe-sip-tmr)# tls-idle-timeout 3600000
```

Router(config-sbc-sbe-sip-tmr)# udp-response-linger-period 32000

```
Router(config-sbc-sbe-sip-tmr)# udp-first-retransmit-interval 500
Router(config-sbc-sip-tmr)# udp-max-retransmit-interval 4000
Router(config-sbc-sbe-sip-tmr)# invite-timeout 180
Router (config-sbc-sbe)# codec-list allow711u
Router(config-sbc-sbe-codec-list)# codec PCMU
Router (config-sbc-sbe) # codec-list allowg729
Router(config-sbc-sbe-codec-list)# codec G729
Router(config-sbc-sbe) # h323
Router(config-sbc-sbe-h323)# ras timeout arg 5000
Router(config-sbc-sbe-h323)# ras retry arg 2
Router(config-sbc-sbe-h323)# ras timeout brg 3000
Router(config-sbc-sbe-h323)# ras retry brg 2
Router(config-sbc-sbe-h323)# ras timeout drg 3000
Router(config-sbc-sbe-h323) # ras retry drg 2
Router(config-sbc-sbe-h323) # ras timeout grg 5000
Router(config-sbc-sbe-h323)# ras retry grq 2
Router(config-sbc-sbe-h323) # ras timeout rrg 3000
Router(config-sbc-sbe-h323)# ras retry rrq 2
Router(config-sbc-sbe-h323)# ras rrg ttl 60
Router(config-sbc-sbe-h323) # ras timeout urg 3000
Router(config-sbc-sbe-h323)# ras retry urg 1
Router(config-sbc-sbe-h323)# h225 timeout proceeding 10000
Router(config-sbc-sbe-h323) # h225 timeout establishment 180000
Router(config-sbc-sbe-h323) # h225 timeout setup 4000
Router(config-sbc-sbe-h323) # ras rrg keepalive 45000
Router(config-sbc-sbe)# h323
Router(config-sbc-sbe-adj-h323)# adjacency timeout 30000
Router(config-sbc-sbe) # blacklist
Router(config-sbc-sbe-blacklist)# global
Router(config-sbc-sbe) # blacklist
Router(config-sbc-sbe-blacklist)# address-default
Router(config-sbc-sbe) # redirect-limit 2
Router(config-sbc-sbe) # deact-mode normal
Router (config-sbc-sbe)# activate
Router(config-sbc) # dbe
Router(config-sbc-dbe) # location-id 0
Router(config-sbc-dbe) # media-timeout 360
Router(config-sbc-dbe) # deact-mode normal
```

Verification

Use the following **show sbc sbe media-gateway-associations** command to display a list of known media gateways with an active association and to verify the operation:

The following example shows the SBC and media communications.

Router# show sbc slt-n2 sbe media-gateway-associations

```
SBC Service "slt-n2"
Media gateway 192.169.125.1:2944
Gateway Protocol = megaco
Transport Protocol = UDP
```

Router (config-sbc-dbe)# activate

Local	Address	= 22.46.0.11:2944		
Requests Replies	Sent 117 2	Received 2 116	Failed 1 -	Retried 7 0
	-			-

Voice Transcoding Per Adjacency Statistics

The Voice Transcoding Per Adjacency Statistics feature provides the transcoding statistics to the user for voice calls at both global and adjacency levels. The feature analyzes the consumption of the cards, such as the DSP cards, that provide the transcoding functions.

The transcoding statistics include the following information:

- The number of active transcoding media stream for each codec pair over several summary periods at global and adjacency scopes. The statistic also provides a high water mark for the corresponding codec pair.
- The number of active transcoding calls both per-adjacency and globally are listed. The statistics can be listed both at global and adjacency scopes for the list of codec pairs.
- The statistics display the codec names for the following codecs, if the transcoding call uses any other codecs, the codec name is displayed as *Other*:
 - **-** G711A
 - **-** G711U
 - G729
 - GSM
 - **-** T38
 - CLEAR

Configuring the Voice Transcoding Per Adjacency Statistics

This task shows how to configure the Voice Transcoding Per Adjacency Statistics feature, list the transcoding statistics as per the scope and summary period, and also reset the transcoding statistics.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. transcoding-stats enable
- 5. end
- 6. show sbc *sbc-name* sbe transcoding-stats {global | adjacency *adjacency-name*} {current15mins | current5mins | currentday | currenthour | current-indefinite | previous15mins | previous5mins | previousday | previoushour}
- 7. clear sbc *sbc-name* sbe transcoding-stats [global | adjacency *adjacency-name*] [all | current-indefinite]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the SBC service mode. Use the <i>sbc-name</i> argument to define the name of the service.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the SBE entity mode within a SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	transcoding-stats enable	Enables or disables the transcoding related statistics for the SBC.
	Example: Router(config-sbc-sbe)# no transcoding-stats enable	The following warning is issued and the user needs to confirm y (Yes) or n (No) to enable or disable the transcoding statistics:
		Note This will re-activate the SBC, and existing calls will be impacted[confirm]
		By default, the transcoding related statistics for the SBC is enabled.
Step 5	end	Exits the SBE entity mode and enters the Privileged Exec mode.
	Example: Router(config-sbc-sbe)# end	

	Command or Action	Purpose
Step 6	<pre>show sbc sbc-name sbe transcoding-stats {global adjacency adjacency-name} {current15mins current5mins currentday currenthour current-indefinite previous15mins previous5mins previousday previoushour}</pre>	Lists the voice transcoding statistics for the required scope and summary period.
		• adjacency —Lists the transcoding statistics for the specified adjacency.
	Example: Router# show sbc mySBC sbe transcoding-stats adjacency SIPP current5mins	• global —Lists the globally scoped statistics for the entire SBC.
		• current15mins —Lists the statistics for the current 15 minute interval.
		• current5mins —Lists the statistics for the current 5 min. interval.
		• currentday —Lists the statistics for the current day from midnight.
		• currenthour —Lists the statistics for the current hour.
		• currentindefinite —Lists the statistics for the period since the last explicit reset.
		• previous15mins —Lists the statistics for the previous 15 minute interval.
		• previous5mins —Lists the statistics for the previous 5 min. interval.
		• previousday —Lists the statistics for the previous day.
		• previoushour —Lists the statistics for the previous hour.
Step 7	<pre>clear sbc sbc-name sbe transcoding-stats [global adjacency adjacency-name] [all current-indefinite]</pre>	Clears the transcoding statistics for all or current-indefinite summary period.
		• adjacency —Clears statistics for the adjacency.
	Example:	• global —Clears the global transcoding statistics.
	Router# clear sbc mySBC sbe adjacency SIPP all	• all —Clears statistics for all summary periods.
		• currentindefinite —Clear statistics for only the current-indefinite period.

The following example shows the output of the **show sbc sbe transcoding-stats adjacency current15mins** command:

Router# show sbc mySBC sbe transcoding-stats adjacency SIPP current15mins

Codec1Codec2Transcoded StreamHWM of TranscodedStreamLast ResetG711AG711U4102010/09/1019:27:15

Media Gateway-Assisted DTMF Interworking

The SBC enables inband DTMF interworking using media gateway switches such as the MGX 8880. A Cisco MGX 8880 is used with DTMF interworking in the following scenarios:

- As a transcoder—DTMF interworking between media plane and signaling is supported.
- As an inband DTMF extractor or injector.

The SBC supports two types of media plane DTMF:

- RFC2833 (telephone-event)
- Inband DTMF—DTMF inband audio stream, such as G.711. To support inband DTMF, MGX performs the following tasks:
 - Monitors the audio stream.
 - Extracts the DTMF signal.
 - Reports or injects the DTMF signal into the voice band, and vice versa.

DTMF Interworking with MGX as Transcoder

When the SBC uses an external transcoder, such as MGX, DTMF interworking is supported for the following:

- Between media and signaling in a call.
- For both, negotiated transcoding and transcoding provisioned through the use of codec lists.
- Between supported media formats, such as RFC2833, and supported SIP signaling formats, such as INFO or NOTIFY.

Inband DTMF Support—Interworking Without a Transcoder

The SBC supports a call or adjacency policy to indicate when an inband DTMF tone is monitored. Monitoring an inband DTMF tone can either be forced, or an optional task in the absence of any other DTMF support.

The SBC enables interworking between any two of the three supported DTMF formats, which include:

- Inband
- RFC 2833 telephone events
- Signaling.

In the event of a failover, active calls using any DTMF interworking option are protected, and the interworking capability is retained on restoration.

The SBC provides a per-adjacency option to enforce an inband DTMF-compatible codec negotiation if no other methods are available for receiving or sending DTMF.

Configuring Inband DTMF Interworking

To configure inband DTMF interworking, perform the following steps.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name
- 6. first-cac-table table-name
- 7. cac-table table-name
- 8. table-type {policy-set | limit {list of limit tables}}
- 9. entry entry-id
- **10.** cac-scope {list of scope options}
- **11.** callee inband-dtmf-mode {always | inherit | maybe | never}
- **12.** caller inband-dtmf-mode {always | inherit | maybe | never}
- 13. complete
- 14. active-call-policy-set policy-set-id
- 15. end
- 16. show sbc service-name sbe cac-policy-set id table name entry entry

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	

	Command or Action	Purpose
Step 3	sbe	Configures the submode of the SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the submode of the CAC policy.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	first-cac-scope scope-name	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope global	Note The first-cac-scope definition is relevant only if the table type configured by the first-cac-table command is a Limit table. In that case, the scope of the first-cac-table is determined by first-cac-scope. If the first-cac-table is a Policy Set table, the first-cac-scope is ignored, and defaults to Global.
		The <i>scope-name</i> argument configures the scope at which limits should be defined initially. Possible <i>scope-name</i> values are:
		• adj-group
		• call
		• category
		• dst-account
		• dst-adj-group
		• dst-adjacency
		• dst-number
		• global
		• src-account
		• src-adj-group
		• src-adjacency
		• src-number
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacency.
Step 6	first-cac-table table-name	Configures the name of the first policy table to be processed when performing the admission control stage of the policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table first_policy_table	

	Command or Action	Purpose
Step 7	<pre>cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# cac-table first policy table</pre>	Enters the mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set.

	Command or Action	Purpose
Step 8	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	The <i>list of limit tables</i> argument controls the syntax of the match-value fields of the entries in the table. Possible <i>list of limit tables</i> values are:
		• account —Compare with name of the account.
		• adj-group —Compare with name of the adjacency group.
		• adjacency —Compare with name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare with number analysis-assigned category.
		• dst-account —Compare with name of the destination account.
		• dst-adj-group —Compare with name of the destination adjacency group.
		• dst-adjacency —Compare with name of the destination adjacency.
		• dst-prefix —Compare with beginning of the dialed digit string.
		• event-type—Compare with CAC policy event types.
		• src-account —Compare with name of the source account.
		• src-adj-group —Compare with name of the source adjacency group.
		• src-adjacency —Compare with name of the source adjacency.
		• src-prefix —Compare with beginning of the calling number string.
		Note For Limit tables, the event, message, call matches only a single entry.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches with either the source or the destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
		Note For Policy Set tables, the event, call or message, is applied to all entries in this table.

	Command or Action	Purpose
Step 9	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	
Step 10	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each entry at which limits are applied in a Policy Set table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # cac-scope call</pre>	Only per-call scope can be configured when using the codec-restrict-to-list command.
		<i>list of scope options</i> —Specifies one of the following strings used to match events:
		• account —Events that are from the same account.
		• adjacency —Events that are from the same adjacency.
		• adj-group —Events that are from members of the same adjacency group.
		• call —Scope limits are per single call.
		• category —Events that have the same category.
		• dst-account —Events that are sent to the same account.
		• dst-adj-group —Events that are sent to the same adjacency group.
		• dst-adjacency —Events that are sent to the same adjacency.
		• dst-number —Events that have same destination.
		• global —Scope limits are global.
		• src-account —Events that are from the same account.
		• src-adj-group —Events that are from the same adjacency group.
		• src-adjacency —Events that are from the same adjacency.
		• src-number —Events that have the same source number.
		• sub-category —The limits specified in this scope apply to all the events sent to or received from members of the same subscriber category.
		• sub-category-pfx —The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		• subscriber —The limits specified in this scope apply to all the events sent to or received from individual subscribers
		A device that is registered with a Registrar server.

	Command or Action	Purpose	
Step 11	callee inband-dtmf-mode {always inherit maybe never} Example:	To configure a DTMF inband mode for the callee side, use the callee inband-dtmf-mode command in the CAC table configuration mode. To deconfigure the DTMF inband mode for the callee side, use the no form of this command.	
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee inband-dtmf-mode always	The callee inband-dtmf-mod e specifies one of the following strings:	
		• always —The inband DTMF tones are always in use by an endpoint.	
		• inherit —The inband DTMF mode for an endpoint is not affected by the CAC entry.	
		• maybe —The inband DTMF tones are used by an endpoint unless signaling indicates that an alternative format for a DTMF is in use.	
		• never —An endpoint never uses inband DTMF.	
Step 12	caller inband-dtmf-mode {always inherit maybe never} Example:	To configure the DTMF inband mode for the caller side, use the caller inband-dtmf-mode command in the CAC table configuration mode. To deconfigure the DTMF inband mode for the caller side, use the no form of this command.	
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller inband-dtmf-mode never	The caller inband-dtmf-mod e specifies one of the following strings:	
		• always —The inband DTMF tones are always in use by an endpoint.	
		• inherit —The inband DTMF mode for an endpoint is not affected by the CAC entry.	
		• maybe —The inband DTMF tones are used by an endpoint unless signaling indicates that an alternative format for a DTMF is in use.	
		• never —An endpoint never uses inband DTMF.	
Step 13	complete	Completes the CAC-policy.	
	Example: Router(config-sbc-sbe-cacpolicy)# complete		
Step 14	active-call-policy-set policy-set-id	Sets the active routing policy set within an SBE entity.	
	Example: Router(config-sbc-sbe-cacpolicy)# cac-policy-set global 1		

	Command or Action	Purpose
Step 15	end	Exits the cac-policy-set configuration mode and enters the Privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# end	
Step 16	<pre>show sbc service-name sbe cac-policy-set id table name entry entry</pre>	Displays detailed information, including DTMF interworking, pertaining to a specific entry in a CAC policy table.
	Example:	
	Router# show sbc mysbc sbe cac-policy-set 1 table standard_policy_list entry 1	

Configuring Codecs to Support Inband DTMF

To configure the codecs to support inband DTMF, perform the following tasks:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. codec system system-name id
- 5. options {none | transrate | transcode | inband-dtmf}
- 6. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode of an SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Configures the submode of the SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	codec system system id	Specifies the name of the system, analog-to-digital codec (enCOder/DECoder), and enters the Codec definition mode.
	Example: Router(config-sbc-sbe)# codec system GSM id 3	

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	Command or Action	Purpose
Step 5	options {none transrate transcode inband-dtmf}	Configures the codec that will support voice inband DTMF. The values for the options are:
	Example: Router(config-sbc-sbe-codec-def)# options inband-dtmf	 none transrate transcode inband-dtmf
Step 6	end	Exits the cac-policy-set configuration mode and enters the Privileged EXEC mode.
	Example: Router(config-sbc-sbe-cacpolicy)# end	

Blended Transcoding

The Blended Transcoding feature enables the SBC to establish sessions without transcoding.

Do not enable the Blended Transcoding feature in the following situations:

- When using H.323 or SIP-H.323 interworking calls
- When the calls are under transcoding video streams
- When the calls are in fax transcoding

The Blended Transcoding feature does not work with the following features:

- Media Bypass
- H.323 Calls and SIP-H.323 Interworking
- Late-Early Interworking
- Downstream Forking with Codec Change
- Local Call Transfer
- IMS (Gq and Rx)

Before you enable the Blended Transcoding feature, make sure that the DSP farm codec is already configured. For more information about the DSP farm codec configuration, see the *Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model* at:

http://www.cisco.com/en/US/partner/docs/routers/asr1000/configuration/guide/sbcu/sbc_spadsp.html# wp1157164

Enabling Blended Transcoding

To enable the Blended Transcoding feature, perform the following steps:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe

- 4. codec list list-name
- 5. codec codec-name
- 6. cac-policy-set cac-policy-name
- 7. first-cac-table table-name
- 8. cac-table cac-table-name
- 9. table-type limit adjacency
- 10. entry entry-id
- **11.** match-value string-value
- 12. blended-transcode
- 13. blended-codec-list codec-list-name
- 14. action cac-complete
- 15. complete
- 16. cac-policy-set global policy-set-id

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enables the SBC service.
	Example: Router(config)# sbc mysbc	
Step 3	sbe	Enters the mode of the SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	codec list list-name	Creates a codec list.
	Example: Router(config-sbc-sbe)# codec list my_codecs	
Step 5	codec codec-name	Adds a codec to the codec list.
	Example: Router(config-sbc-sbe-codec-list)# codec PCMU	
Step 6	cac-policy-set policy-set-id	Creates a new call admission control (CAC) policy set.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	

	Command or Action	Purpose	
Step 7	first-cac-table table-name	Specifies the admission control table that should be processed first.	
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table BlendedTranscodeTable		
Step 8	cac-table cac-table-name	Creates an admission control table.	
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table BlendedTranscodeTable		
Step 9	table-type limit adjacency	Configures a CAC table type that determines the priority of the call to be used as a criterion in the CAC policy.	
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit adjacency		
Step 10	entry entry-id	Creates an entry in the CAC table.	
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1		
Step 11	match-value string-value	Specifies the adjacency that is to be enabled with the Blended Transcoding feature.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value SIP3</pre>		
Step 12	blended-transcode	Enables the Blended Transcoding feature.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # blended-transcode</pre>		
Step 13	blended-codec-list codec-list name	Configures a blended codec list.	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # blended-codec-list my_codecs</pre>		
Step 14	action cac-complete	Configures the action to be performed after the CAC entry in an admission control table; indicates that this CAC policy	
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	is complete.	

	Command or Action	Purpose
Step 15	complete	Ends the configuration of the CAC table, and goes back to the SBC SBE configuration mode.
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # complete	
Step 16	cac-policy-set global policy-set-id	Activates the global CAC policy set within the SBE entity.
	Fxamnle [,]	
	Router (config-sbc-sbe)# cac-policy-set global	
	1	

Configuration Example for Blended Transcoding

The following example shows how to enable the Blended Transcoding feature:

```
codec list codec-a
        codec PCMU
        codec G729
cac-policy-set 1
        first-cac-table BlendedTranscodingTable
        cac-table BlendedTranscodingTable
        table-type limit adjacency
        entry 1
        match-value SIP3
        blended-transcode
        blended-codec-list codec-a
        action cac-complete
        complete
cac-policy-set global 1
```



Cisco Unified Border Element (SP Edition)—SPA DSP Services

The shared port adapter (SPA) digital signal processor (DSP) is a single-width, half-height, high-power, SPA module that can be used across multiple Cisco platforms. The SPA DSP is designed for DSP-based voice and video solutions in the SPAs on the Cisco mid-range and high-end routers.

In Cisco IOS XE Release 3.2S, the following SPA DSP features have been deployed on the Cisco ASR 1000 Series Router for the session border controller (SBC):

- Associating SBC configuration with a DSP farm profile.
- Voice transcoding and transrating support using onboard DSP services.
- Dual tone multifrequency (DTMF) interworking using onboard DSP services.
- VoIPv4 and VoIPv6 transcoding and transrating support.
- Transcoding, transrating, and DTMF interworking call control and signaling control.

Cisco Unified Border Element (SP Edition) was earlier known as Integrated Session Border Controller, and is referred to as SBC in this document.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all the Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or the Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 3.2S	The SPA DSP onboard services were introduced on the Cisco ASR 1000 Series Routers.
Cisco IOS XE Release 3.3S	The Call Recovery feature was added.
Cisco IOS XE Release 3.8S	The AMR-WB feature was supported on the SBC on the Cisco ASR 1000 Aggregation Services Routers.

Feature History of SPA DSP on the Cisco Unified Border Element (SP Edition)

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- Restrictions, page 41-2
- Prerequisites for the SPA DSP Services, page 41-2
- Information About the SPA DSP Services, page 41-2
- Configuring the SPA DSP Services for SBC, page 41-7
- Configuring the Unified SBC, page 41-10
- Configuration Examples of the SPA DSP Services for the SBC, page 41-34
- Configuration Examples of Unified SBC, page 41-36

Restrictions

The following restrictions are applicable to a SPA DSP:

- Voice, audio, and video conferencing are not supported.
- HA, system-level In-Service Software Upgrade (ISSU), and Nonstop Forwarding (NSF) are not supported.
- Video codecs are not supported.
- Although Online Insertion and Removal (OIR) is supported, the sessions going through a SPA at the time of removal are lost.
- The Cisco Unified Communications Manager is not supported.

Prerequisites for the SPA DSP Services

The DSP farm definition and SBC configuration and activation must be completed before transcoding the SBC calls. For more information about SPA configuration, see the "Configuring the Cisco DSP SPA for the ASR 1000 Series" chapter in *Cisco ASR 1000 Series Aggregation Services Routers SIP and SPA Software Configuration Guide* at:

http://www.cisco.com/en/US/docs/interfaces_modules/shared_port_adapters/configuration/ASR1000/a srcfgdsp.html

Information About the SPA DSP Services

A SPA DSP contains digital signal processors and related hardware to provide voice transcoding capability for the SBC. In addition, Cisco Unified Border Element, Enterprise can use a SPA DSP for simple voice transcoding services.

You can find more information on terminating and generating the RTCP by the SPA-DSP at: http://www.cisco.com/en/US/docs/interfaces_modules/shared_port_adapters/configuration/ASR1000/ ASRovdsp.html#wp1296621

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Transcoding the SBC

SBC transcoding is used for codec translation between two VoIP networks as part of the Data Border Element (DBE) functions. Figure 41-1 shows how a SPA DSP performs codec transcoding for unified SBC and Figure 41-2 shows how a SPA DSP performs codec transcoding for distributed SBC.







The SPA DSP allows the translation of one type of media stream or codec to another type of media stream that uses different media encoding and decoding technologies. Other translation activities include:

- Translation between different codecs
- Translation between different packetization settings (transrating)
- DTMF interworking

Transcoding the Distributed SBC

Transcoding is inferred from a Session Description Protocol (SDP) that is used to program a call. Programming terminations in the same call containing different codecs implicitly instruct the distributed SBC to perform transcoding.

Transrating the Distributed SBC

Transrating is inferred from the SDP that is used to program a call. Programming terminations in the same call with different ptime implicitly instruct the distributed SBC to perform transrating.



Transrating is supported only for the different rates using the same codec, not across codecs. Therefore, transrating and transcoding cannot be performed simultaneously.

RTP Telephone-Event Codec-to-SIP Interworking

When an RTP packet is marked as DTMF using the telephone-event codec, the RTP packet is removed from the stream. The DBE sends an H.248 message to the signaling border element (SBE), indicating that a DTMF event has occurred, and that the RTP packet should be converted into a SIP DTMF event.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is present in side A of the SDP, but not in side B.
- The dd/etd event is subscribed for side A, but not for side B.

SIP-to-RTP Telephone-Event Codec Interworking

When an endpoint generates a SIP signal, the SIP DTMF signals arrive completely out of band. An endpoint that supports SIP DTMF generates the signals to be sent to the SBE. In turn, the SBE recognizes that this is a DTMF message and sends an H.248 message to the DBE, indicating that a DTMF tone is required to be inserted into the RTP stream. The DBE then inserts the RTP DTMF packets into the audio stream using telephone-event codec.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is present in side B of the SDP, but not in side A.
- The dd/etd event is subscribed for side B, but not for side A.

RTP Telephone-Event Codec-to-RTP In-Band Waveform

After the RTP packet is marked as DTMF using the telephone-event codec, the RTP packet is removed from the stream, and an RTP stream containing the DTMF waveform is sent to the other endpoint.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is present in side A of SDP, but not in side B.
- The dd/etd event is subscribed for side A and side B.

RTP In-Band Waveform-to-RTP Telephone-Event Codec

After the DTMF is sent as part of the voice waveform, the RTP packets are removed from the stream, and the DBE inserts the a new RTP packet with the payload-type telephone event into the audio stream.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is present in side B of the SDP, but not in side A.
- The dd/etd event is subscribed for side A and side B

SIP-to-RTP In-Band Waveform

After an endpoint generates a SIP signal, the SIP DTMF signals arrive completely out of band. The endpoint that supports SIP DTMF generates the signals to be sent to the SBE. In turn, the SBE recognizes that this is a DTMF message, and sends an H.248 message to the DBE, indicating that a DTMF tone is required to be inserted into the RTP stream. The DBE then inserts a stream containing the DTMF waveform.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is not present on either side A or side B.
- The dd/etd event is subscribed for side B.

RTP In-Band Waveform-to-SIP

When the DTMF is sent as part of the voice waveform, the RTP packets are removed from the stream, and the DBE sends an H.248 message to the SBE, indicating that a DTMF event has occurred, and that the RTP packets should be converted into a SIP DTMF event.

The call must meet the following conditions:

- The telephone-event codec (for RFC 2833) is not present on either side A or side B.
- The dd/etd event is subscribed for side A.

Call Recovery

From Cisco IOS XE Release 3.3S, calls on a partially crashed SPA DSP can be recovered within the call outage time of 2.5s.

When part of a SPA DSP crashes, a crash recovery process runs, and then the RP reprograms the crashed part of the SPA DSP with all calls that were previously on it. For example, a simple transcoding scenario, a-law to u-law transcoding, can represent up to 129 calls that require reprogramming.

Depending on the part of the SPA DSP that crashes, the total recovery time may be longer because it might have to recover more components and also reprogram more calls. However, the entire media path outage time for all the recovered calls is less than 2.5s.

In all cases of the SPA DSP call recovery, the call recovery occurs on the same SPA DSP where the call existed prior to the crash. The calls are not moved to another SPA DSP.

The SPA DSP failure call recovery can be disabled or rendered ineffective if the SPA DSP crash dumps are enabled. It can push the call outage time beyond 2.5s.

The show voice dsp group all command indicates when a SPA DSP is undergoing call recovery.

Router# show voice dsp group all

```
Show DSP group all
DSP groups on slot 0 bay 0:
dsp 1:
   State: UP
   HA State : DSP_HA_STATE_PENDING1
   Max signal/voice channel: 43/43
   Max credits: 645
   num_of_sig_chnls_allocated: 43
   Transcoding channels allocated: 43
   Group: FLEX_GROUP_XCODE, complexity: LOW
```

```
Note
```

The **show voice dsp group all** command displays the output **HA STATE : DSP_HA_STATE_PENDING1** only during the recovery process which can be upto a few milliseconds.

AMR-WB Transcoding Support

Adaptive Multi-Rate Wideband (AMR-WB) is a patented speech coding standard based on Adaptive Multi-Rate encoding, using a methodology that is similar to the Algebraic code-excited linear prediction (ACELP). AMR-WB, which was specified by 3GPP, provides improved speech quality due to a wider speech bandwidth of 50 to 7000Hz compared to narrowband speech coders what are in general optimized for Plain old telephone service (POTS) wireline quality of 300 to 3400 Hz.

AMR-WB is codified as G.722.2, an ITU-T standard speech codec, formally known as Wideband coding of speech at around 16 kbps using AMR-WB. G.722.2 AMR-WB is the same codec as the 3GPP AMR-WB.

AMR-WB operates like AMR with nine different bit rates. The lowest bit rate providing excellent speech quality in a clean environment is 12.65 kbps. Higher bit rates are useful in background noise conditions and for music. Also, lower bit rates of 6.60 and 8.85 kbps provide reasonable quality, especially compared to narrowband codecs.

Note

The AMR-WB feature requires DSP firmware with AMR-WB codec support.

Shared credits: 0, reserved credits: 645 Transcoding channels allocated: 24 Credits used (rounded-up): 360

Rate Mode	AMR Bit-Rate (kbps)	AMR-WB/G.722.2 Bit-Rate (kbps)
0	4.75	6.60
1	5.15	8.85
2	5.90	12.65
3	6.70	14.25
4	7.40	15.85
5	7.95	18.25
6	10.20	19.85
7	12.20	23.05
8	SID ¹	23.85
9		SID

Table 41-1 shows the relationship between the AMR rate mode and bit-rate.

 Table 41-1
 Relationship Between the AMR Rate Mode and Bit-Rate

1. SID: Silence Indicator

Configuring the SPA DSP Services for SBC

This section describes the tasks to involved in configuring the SPA DSP services for the SBC:

- Setting Up a SPA DSP for DSP Farm Services, page 41-7
- Configuring a DSP Farm Profile, page 41-8

Setting Up a SPA DSP for DSP Farm Services

Use the following procedure to set up the SPA DSP in the DSP farm mode for the DSP services:

SUMMARY STEPS

- 1. configure terminal
- 2. voice-card slot number/subslot number
- 3. dsp services dspfarm
- 4. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	voice-card slot number/subslot number	Specifies the slot number of the voice card and enters the voice card interface configuration mode.
	Example: Router(config)# voice-card 0/2	
Step 3	dsp services dspfarm	Allows DSP farm services on the SPA DSP voice card.
	Example: Router(config-voicecard)# dsp services dspfarm	
Step 4	end	Exits the voice card interface configuration mode.
	Example: Router(config-voicecard)# end	

For more information about configuring DSP farm services on a SPA DSP, see the "Configuring the Cisco DSP SPA for ASR 1000 Series" chapter in the *Cisco ASR 1000 Series Aggregation Services Routers SIP and SPA Software Configuration Guide* at:

http://www.cisco.com/en/US/docs/interfaces_modules/shared_port_adapters/configuration/ASR1000/a srcfgdsp.html

Configuring a DSP Farm Profile

Use the following steps to configure a DSP farm profile:

SUMMARY STEPS

- 1. configure terminal
- 2. dspfarm profile *profile-identifier* {conference | mtp | transcode}
- 3. description profile-description-text
- 4. codec codec-name
- 5. associate application {cube | sbc | sccp}
- 6. maximum session number
- 7. no shutdown
- 8. end
DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	<pre>dspfarm profile profile-identifier {conference mtp transcode}</pre>	Enables the DSP farm service for the specified DSP farm profile, and enters a DSP farm profile configuration mode.
	Example: Router(config)# dspfarm profile 20 transcode	 The service options are: conference—Enables conferencing. mtp—Enables media termination point. transcode—Enables transcoding of information.
		service is supported.
Step 3	description profile-description-text	Specifies a description for a defined profile.
	Example: Router(config-dspfarm-profile)# description <i>enables transcoding</i>	
Step 4	<pre>no codec codec-name Example: Router(config-dspfarm-profile)# codec g711ulaw Router(config-dspfarm-profile)# codec g711alaw Router(config-dspfarm-profile)# codec g729ar8 Router(config-dspfarm-profile)# codec g729abr8 Router(config-dspfarm-profile)# codec g729r8 Router(config-dspfarm-profile)# codec g723r63 Router(config-dspfarm-profile)# codec ilbc Router(config-dspfarm-profile)# codec g726r32 Router(config-dspfarm-profile)# codec g729br8</pre>	Adds codecs or removes the codec from a codec list. The codec must be present in the list of codecs that the SBE is hard-coded to recognize.
Step 5	associate application { cube sbc sccp } profile-description-text	Associates an application to the profile. The applications that can be associated are:
	Example: Router(config-dspfarm-profile)# associate application sbc	 cube—Associates the Cisco Unified Border Element application to a defined profile in the DSP farm. sbc—Associates the SBC application to a defined profile in the DSP farm.
		• sccp —Associates the client control protocol application to a defined profile in the DSP farm.
		Note The sbc application keyword is available only when a DSP farm profile transcode service is used.

	Command or Action	Purpose
Step 6	<pre>maximum session number Example: Router(config-dspfarm-profile)# maximum session 300</pre>	Establishes the maximum number of sessions that can be assigned to a defined profile. The maximum number of sessions is dependent upon the number of SPA DSPs in the router, and the codecs configured. For a fully populated Cisco ASR 1013 Series Router with 23 SPA DSPs and only the G711 codec, the maximum number of sessions would be 20769.
Step 7	no shutdown	Enables or disables a DSP farm profile.
	Example: Router(config-dspfarm-profile)# no shutdown	
Step 8	end	Exits the DSP farm profile.
	Example: Router(config-dspfarm-profile)# end	

Configuring the Unified SBC

This section explains the various ways in which to configure the SBC for the SPA DSP voice card:

- Associating the Unified SBC with a DSP Farm Profile, page 41-10
- Configuring the Unified SBC to Enable Transcoding, page 41-11
- Configuring the Unified SBC to Enable Transrating, page 41-17
- Configuring the Unified SBC to Enable SRTP and Transcoding, page 41-22
- Configuring the Unified SBC for Inband DTMF Interworking, page 41-28
- Configuring the Unified SBC to Support AMR-WB, page 41-33

Associating the Unified SBC with a DSP Farm Profile

Association of the SBC to the DSP farm profiles is possible only after the corresponding DSP farm profile is created. Use the **associate dspfarm profile** command in the global configuration mode.

SUMMARY STEPS

- 1. show dspfarm {all | dsp | profile}
- 2. configure terminal
- 3. sbc sbc-name
- 4. **associate dspfarm profile** {*profile-number* | **all**}
- 5. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	show dspfarm {all dsp profile	Displays the DSP farm configuration information:
	profile-identifier}	• all —Displays the DSP farm global information.
	Evample [,]	• dsp —Displays information pertaining to all the DSPs.
	Router# show dspfarm profile all	• profile —Displays the DSP farm profiles.
Step 2	configure terminal	Enables the global configuration mode.
	Example: Router# configure terminal	
Step 3	sbc sbc-name	Creates the SBC service on the SBC, and enters the SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 4	associate dspfarm profile {profile-number	Associates the SBC to a DSP farm profile:
	all}	• <i>profile-number</i> —Specifies the profile number to be associated.
	Example: Router(config-sbc)# associate dspfarm profile 1	• all —Allows the SBC to pick the most appropriate DSP farm profile from the profiles associated to the SBC for the transcoding session.
Step 5	end	Exits the configuration mode.
	Example: Router(config-sbc-sbe)# end	

Configuring the Unified SBC to Enable Transcoding

This task configures the SBC for enabling the transcoding feature.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name

- 6. first-cac-table table-name
- 7. cac-table *table-name*
- 8. table-type limit list of limit tables
- 9. entry entry-id
- 10. match-value key
- **11.** callee-codec-list *list-name*
- **12.** caller-codec-list *list-name*
- 13. media police strip | reject | degrade
- 14. action cac-complete
- 15. complete
- 16. cac-policy-set global cac-policy-num
- **17.** codec-list *list-name*
- **18.** codec codec-nam
- 19. exit
- 20. codec-list list-name
- 21. codec codec-nam
- 22. exit
- 23. end
- 24. show sbc *sbc-name* sbe call-stats global current5min

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters the SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the signaling border element (SBE) function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary:
	Example: Router(config-sbc-sbe)# cac-policy-set 1	• <i>policy-set-id</i> —Integer chosen by a user to identify the policy set. The range is from 1 to 2147483647.

	Command or Action	Purpose
Step 5	<pre>first-cac-scope scope-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope dst-adjacency</pre>	Configures the scope at which limits should be initially defined to perform tasks at the admission control stage of the policy. Each CAC policy has a scope that can be applied to it. This CAC policy is applicable on a per call basis.
		<i>scope-name</i> has one of the following values:
		• adj-group —Limits for events from members of the same adjacency group.
		• call —Limits are per single call.
		• category —Limits per category.
		• dst-account —Limits for events sent to the same account.
		• dst-adj-group —Limits for events sent to the same adjacency group.
		 dst-adjacency—Limits for events sent to the same adjacency. dst-number—Limits for events that have the same adjacency number.
		• src-adj-group —Limits for events from the same adjacency group.
		• arc-adjacency —Limits for events from the same adjacency.
		• src-number —Limits for events that have the same source number.
Step 6		<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table codec-dst-acc</pre>
	• <i>table-name</i> —The admission control table that should be processed first.	
Step 7	<pre>cac-table table-name</pre>	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table codec-dst-acc	 <i>table-name</i>—Name of the admission control table.

	Command or Action	Purpose
Step 8	table-type limit list of limit tables	Configures a new CAC Limit table type in which the criteria used to match the entries must be entered.
	Example:	list of limit tables can be one of the following values:
	Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit dst-adjacency	• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
Step 9	entry entry-id	Enters the CAC table entry mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	• <i>entry-id</i> —Specifies the table entry.
Step 10	match-value key	Configures the match value of an entry in a CAC Limit table type.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value nava</pre>	
Step 11	callee-codec-list list-name	Lists the codecs that the callee leg of a call is allowed to use.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee-codec-list PCMU</pre>	

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	Command or Action	Purpose
Step 12	caller-codec-list list-name	Lists the codecs that the caller leg of a call is allowed to use.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-codec-list PCMA	
Step 13	media police strip reject degrade	Configures the manner in which the SBC will handle the media streams that exceed the bandwidth limit for media calls.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # media police strip	
Step 14	action cac-complete	When an event matches, the CAC policy is considered complete.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	
Step 15	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# complete</pre>	
Step 16	cac-policy-set global policy-num	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Example: Router(config-sbc-sbe)# cac-policy-set global 1	 <i>policy-num</i>—The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 17	codec-list <i>list-name</i>	Creates a codec list, and enters the Codec list configuration mode.
	Example: Router(config-sbc-sbe)# codec-list PCMU	
Step 18	codec <i>codec-name</i>	Adds a codec to a codec list.
	Example: Router(config-sbc-sbe-codec-list)# codec PCMU	
Step 19	exit	Exits the codec list configuration mode.
	Example: Router(config-sbc-sbe-codec-list)# exit	
Step 20	codec-list <i>list-name</i>	Creates a codec list, and enters the Codec list configuration mode.
	Example: Router(config-sbc-sbe)# codec-list PCMA	
Step 21	codec <i>codec-name</i>	Adds a codec to a codec list.
	Example: Router(config-sbc-sbe-codec-list)# codec PCMA	

	Command or Action	Purpose
Step 22	exit	Exits the codec list configuration mode.
	Example: Router(config-sbc-sbe-codec-list)# exit	
Step 23	end	Ends the configuration session.
	Example: Router(config-sbc-sbe)# end	
Step 24	<pre>show sbc sbc-name sbe call-stats global current5min</pre>	Lists the statistics for all the calls on the specified SBE.
	Example: Router# show sbc mySBC sbe call-stats global current5min	

The following example shows an output of the **show sbc sbe call-stats global current5min** command that lists the count of the active transcoded and transrated calls.

Router# show sbc mySBC sbe call-stats global current5min

SBC Service "mySBC"		
Statistics for the current 5 mins for global counters		
Call count totals:		
Total call attempts =	0	
Total active calls =	1	
Total active IPv6 calls =	0	
Total activating calls =	0	
Total de-activating calls =	0	
Total active emergency calls =	0	
Total active e2 emergency calls =	0	
Total IMS rx active calls =	0	
Total IMS rx call renegotiation attempts =	0	
Total SRTP-RTP interworked calls =	0	
Total active calls not using SRTP =	1	
Total active transcoded calls =	1	
Total active transrated calls =	al active transrated calls = 0	
General call failure counters:		
Total call setup failures =	l call setup failures = 0	
Total active call failures =	active call failures = 0	
Total failed call attempts =	0	
Total failed calls due to update failure =	0	
Total failed calls due to resource failure =	0	
Total failed calls due to congestion =	0	
Total failed calls due to media failure =	0	
Total failed calls due to signaling failure =	0	
Total failed calls due to IMS rx setup failure =	0	
Total failed calls due to IMS rx renegotiation failure =	0	
Total failed calls due to RTP disallowed on call leg =	0	
Total failed calls due to SRTP disallowed on call leg =	0	

Configuring the Unified SBC to Enable Transrating

Note

Transrating is supported only for different rates using the same codec, not across codecs. Therefore, transrating and transcoding cannot be performed simultaneously.

This section describes how to enable transrating using either of the following methods:

- Transrating Using the Same Codec Policy, page 41-17
- Transrating Using a New Codec Policy, page 41-21

Transrating Using the Same Codec Policy

This task configures the SBC for enabling the transrating using the same codec policy.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table table-name
- 7. table-type {policy-set | limit {*list of limit tables*}}
- 8. entry entry-id
- **9.** cac-scope {list of scope options}
- **10.** callee ptime *0-100*
- **11.** caller ptime *0-100*
- 12. media police strip | reject | degrade
- 13. action cac complete
- 14. complete
- 15. cac-policy-set global cac-policy-num
- 16. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters the SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)#</pre>	Configures the name of the first policy table to be processed. A CAC policy may have many tables configured. To start applying the CAC policy, the first table that is used must be defined:
	first-cac-table Transrate	• <i>table-name</i> —The admission control table that should be processed first.
Step 6	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set:
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table Transrate	• <i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
		list of limit tables can be one of the following values:
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type—Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches either the source adjacency group or the destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each of the entries in which limits are applied in a CAC Policy Set table.
Step 8	entry entry-id	Enters the CAC table entry mode to create or modify an entry in an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Enables the selection of a scope at which CAC limits are applied within each entry in a Policy Set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	<i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope call	• <i>account</i> —Events that are from the same account.
		• <i>adjacency</i> —Events that are from the same adjacency.
		• <i>adj-group</i> —Events that are from members of the same adjacency group.
		• <i>call</i> —Scope limits are per single call.
		• <i>category</i> —Events that have the same category.
		• <i>dst-account</i> —Events that are sent to the same account.
		• <i>dst-adj-group</i> —Events that are sent to the same adjacency group.
		• <i>dst-adjacency</i> —Events that are sent to the same adjacency.
		• <i>dst-number</i> —Events that have the same destination.
		• global—Scope limits are global.
		• <i>src-account</i> —Events that are from the same account.
		• <i>src-adj-group</i> —Events that are from the same adjacency group.
		• <i>src-adjacency</i> —Events that are from the same adjacency.
		• <i>src-number</i> —Events that have the same source number.
Step 10	callee ptime <0-100>	Configures the packetization time on the callee side that is forced for calls using this CAC entry.
	Example:	By default, 0 ms is configured, which means no transrating
	<pre>Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee ptime 30</pre>	occurs.
Step 11	caller ptime <0-100>	Configures the packetization time on the caller side that is forced for calls using this CAC entry.
	Example:	By default, 0 ms is configured, which means no transrating
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller ptime 40	occurs.
Step 12	media police strip reject degrade	Configures the manner in which the SBC handles the media streams that exceed the bandwidth limit for media calls.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # media police strip</pre>	
Step 13	action cac-complete	When an event matches, this CAC policy is complete.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac complete</pre>	

	Command or Action	Purpose
Step 14	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# complete</pre>	
Step 15	cac-policy-set global policy-num	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Router(config-sbc-sbe)# cac-policy-set global 1	• <i>policy-num</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 16	end	Exits the CAC configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Transrating Using a New Codec Policy

This task configures the SBC for enabling the transrating feature. This is an alternative mechanism to that described in the ?\$paranum>Transrating Using the Same Codec Policy? section on page 41-17 section for configuring transrating.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. codec list list-name
- 5. policy {minimum | transrating}
- 6. codec codec-name packetization-period packet-period [priority priority-value]
- 7. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters into the SBC configuration mode.
	Example: Router(config)# sbc mySBC	

	Command or Action	Purpose
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	codec list list-name	Creates the codec list and enters the codec list mode.
	Example: Router(config-sbc-sbe)# codec list PCMU	
Step 5	<pre>policy {minimum transrating}</pre>	Configures the packetization period policy that is to be specified as either of the following:
	Example:	• minimum —Packetization period is the minimum.
	Router(config-sbc-sbe-codec-list)# policy minimum	• transrating —Packetization period is transrating.
Step 6	codec codec-name packetization-period packet-period [priority priority-value]	Adds a codec to a codec list, and sets a minimum packetization period and priority value for the codec.
	Example:	
	Router(config-sbc-sbe-codec-list)# codec PCMU packetization-period 200 priority 1	
Step 7	end	Exits the CAC configuration mode, and returns to privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe-codec-list)# end	

Configuring the Unified SBC to Enable SRTP and Transcoding

Although Secure Real-time Transport Protocol (SRTP) is independent of transcoding, both can be configured to be used simultaneously.

This task configures the unified SBC to enable the SRTP and transcoding features.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table table-name

- 7. table-type {policy-set | limit {*list of limit tables*}}
- 8. entry entry-id
- **9.** cac-scope {*list of scope options*}
- 10. srtp support allow
- 11. srtp caller forbid | mandate | allow | prefer
- 12. srtp callee forbid | mandate | allow | prefer
- 13. srtp interworking forbid | allow
- 14. srtp media interworking forbid | allow
- **15.** action next-table goto-table-name
- 16. exit
- 17. exit
- **18.** cac-table table-name
- **19.** table-type limit list of limit tables
- 20. entry entry-id
- **21**. match-value *key*
- 22. callee-codec-list list-name
- 23. action cac-complete
- 24. complete
- 25. cac-policy-set global cac-policy-num
- 26. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters into the SBC configuration mode.
	Example:	
	Router(config)# sbc mySBC	
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example:	
	Router(config-sbc-sbe)# cac-policy-set 3	

	Command or Action	Purpose
Step 5	<pre>first-cac-table table-name Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table C3</pre>	 Configures the name of the first policy table to be processed. A CAC policy may have many tables configured. To start applying the CAC policy, the first table that is used must be defined: <i>table-name</i>—The admission control table that should be processed first.
Step 6	cac-table table-name Example:	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set:
	Router(config-sbc-sbe-cacpolicy)# cac-table C3	• <i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	list of limit tables can be one of the following values:
		• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type—Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacency. The adj-group table type matches on either the source adjacency group or the destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry in which limits are applied in a CAC Policy Set table.
Step 8	entry entry-id	Enters the mode to create or modify an entry in an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Choose a scope at which CAC limits are applied within each entry in a Policy Set table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)	<i>list of scope options</i> —Specifies one of the following strings used to match events:
	# cac-scope global	• <i>account</i> —Events that are from the same account.
		• <i>adjacency</i> —Events that are from the same adjacency.
		• <i>adj-group</i> —Events that are from members of the same adjacency group.
		• <i>call</i> —Scope limits are per single call.
		• <i>category</i> —Events that have the same category.
		• <i>dst-account</i> —Events that are sent to the same account.
		• <i>dst-adj-group</i> —Events that are sent to the same adjacency group.
		• <i>dst-adjacency</i> —Events that are sent to the same adjacency.
		• <i>dst-number</i> —Events that have the same destination.
		• global—Scope limits are global.
		• <i>src-account</i> —Events that are from the same account.
		• <i>src-adj-group</i> —Events that are from the same adjacency group.
		• <i>src-adjacency</i> —Events that are from the same adjacency.
		• <i>src-number</i> —Events that have the same source number.
Step 10	srtp support allow	Configures SRTP support.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp support allow</pre>	
Step 11	<pre>srtp caller forbid mandate allow prefer</pre>	Configures SRTP for the caller side of the call with one of the following SRTP settings:
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp caller mandate	• forbid —SRTP is not supported on the caller side of the call.
		• mandate —SRTP is mandatory on the caller side of the call.
		• allow —SRTP is optional on the caller side of the call.
		• prefer —SRTP is preferred on this adjacency. Both RTP and SRTP are accepted inbound, but only SRTP is offered outbound.

	Command or Action	Purpose
Step 12	<pre>srtp callee forbid mandate allow prefer</pre>	Configures SRTP for the callee side of the call:
	Example:	• forbid —SRTP is not supported on the callee side of the call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp callee mandate	• mandate —SRTP is mandatory on the callee side of the call.
		• allow —SRTP is optional on the callee side of the call.
		• prefer —SRTP is preferred on this adjacency. Both RTP and SRTP are accepted inbound, but only SRTP is offered outbound.
Step 13	<pre>srtp interworking forbid allow</pre>	Configures SRTP-to-RTP interworking.
	Example:	• forbid —Prohibits SRTP-to-RTP interworking on a call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp interworking allow	• allow —Allows SRTP-to-RTP interworking on a call.
Step 14	<pre>srtp media interworking forbid allow</pre>	Configures SRTP-to-RTP media interworking.
	Example:	• forbid —Prohibits SRTP-to-RTP media interworking on a call.
	Router(config-sbc-sbe-cacpolicy-cactable-entry) # srtp media interworking allow	• allow —Allows SRTP-to-RTP media interworking on a call.
Step 15	action next-table goto-table-name	Configures the action to be taken when the routing entry is chosen.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action next-table xcode</pre>	• <i>goto-table-name</i> —Specifies the next routing table to be processed when an event matches the entry.
Step 16	exit	Exits the CAC table entry configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 17	exit	Exits the CAC table configuration mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 18	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table xcode	 <i>table-name</i>—Name of the admission control table.
Step 19	table-type limit list of limit tables	Configures the limit of the table types to be matched by the match-value command. For the example provided here, use the following table type:
	Example:	the following table type:
	Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit dst-adjacency	• <i>dst-adjacency</i> —Compares the name of the destination adjacency.

	Command or Action	Purpose
Step 20	entry entry-id	Enters the CAC table entry mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	• <i>entry-id</i> —Specifies the table entry.
Step 21	match-value key	Configures the match-value of an entry in a Call Admission Control (CAC) Limit table:
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value nav4B</pre>	• <i>key</i> —Specifies the keyword used to match events. The format of the key is determined by the table-type limit.
Step 22	callee-codec-list list-name	Lists the codecs that the callee leg of a call is allowed to use:
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee-codec-list PCMU	• <i>list-name</i> —Specifies the name of the codec list. The maximum size is 30 characters.
Step 23	action cac-complete	When the event matches, this CAC policy is complete.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	
Step 24	complete	Completes the CAC policy set when you have committed the full set.
	<pre>Example: Router(config-sbc-sbe-cacpolicy)# complete</pre>	
Step 25	cac-policy-set global policy-num	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Router(config-sbc-sbe)# cac-policy-set global 3	• <i>policy-num</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 26	end	Exits the CAC configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Configuring the Unified SBC for Inband DTMF Interworking

A SPA DSP can be used to detect the DTMF tones, called inband, that are played in the real-time transport protocol (RTP) stream. Inband DTMF interworking uses SPA DSP resources, and can be used for plain calls and transcoded calls.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-table table-name
- 6. cac-table table-name
- 7. table-type {policy-set | limit {*list of limit tables*}}
- 8. entry entry-id
- **9.** cac-scope {*list of scope options*}
- 10. callee inband-dtmf-mode always
- 11. caller inband-dtmf-mode never
- **12.** action next-table goto-table-name
- 13. complete
- 14. cac-policy-set global cac-policy-num
- 15. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC, and enters into the SBC configuration mode.
	Example: Router(config)# sbc mySBC	
Step 3	sbe	Enters the SBE function mode of the SBC.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	cac-policy-set policy-set-id	Enters the CAC policy set configuration mode within an SBE entity, creating a new policy set, if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 4	
Step 5	first-cac-table table-name Example:	Configures the name of the first policy table to be processed. A CAC policy may have many tables configured. To start applying the CAC policy, the first table that is used
	Router(config-sbc-sbe-cacpolicy)# first-cac-table C4	 <i>table-name</i>—The admission control table that should be processed first.
Step 6	cac-table table-name	Enters the CAC table mode for configuration of an admission control table (creating one, if necessary) within the context of an SBE policy set:
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table C4	• <i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose
Step 7	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBC policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	list of limit tables can be one of the following values:
		• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type—Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacency. The adj-group table type matches either the source adjacency group or destination adjacency group.
		When the policy-set keyword is specified, use the cac-scope command to configure the scope within each entry at which limits are applied in a CAC Policy Set table.
Step 8	entry entry-id	Enters the CAC table entry mode to create or modify an entry in an admission control table.
	Example:	
	Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 9	<pre>cac-scope {list of scope options}</pre>	Choose the scope at which CAC limits are applied within each entry in a Policy Set table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # cac-scope global</pre>	<i>list of scope options</i> —Specifies one of the following strings used to match events:
		• <i>account</i> —Events that are from the same account.
		• <i>adjacency</i> —Events that are from the same adjacency.
		• <i>adj-group</i> —Events that are from members of the same adjacency group.
		• <i>call</i> —Scope limits are per single call.
		• <i>category</i> —Events that have the same category.
		• <i>dst-account</i> —Events that are sent to the same account.
		• <i>dst-adj-group</i> —Events that are sent to the same adjacency group.
		• <i>dst-adjacency</i> —Events that are sent to the same adjacency.
		• <i>dst-number</i> —Events that have the same destination.
		• global—Scope limits are global
		• <i>src-account</i> —Events that are from the same account.
		• <i>src-adj-group</i> —Events that are from the same adjacency group.
		• <i>src-adjacency</i> —Events that are from the same adjacency.
		• <i>src-number</i> —Events that have the same source number.
Step 10	callee inband-dtmf-mode {always inherit maybe never}	Configures the DTMF inband mode for the callee side.
		• always —The inband DTMF tones are always in use by the endpoint.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee inband-dtmf-mode always</pre>	• inherit —The inband DTMF mode for the endpoint is not affected by this CAC entry.
		• maybe —The inband DTMF tones are used by the endpoint unless signaling indicates that an alternative format for DTMF is in use.
		• never —The endpoint never uses inband DTMF.
Step 11	caller inband-dtmf-mode {always inherit maybe never}	Configures the DTMF inband mode for the caller side.
		• always —The inband DTMF tones are always in use by the endpoint.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller inband-dtmf-mode never</pre>	• inherit —The inband DTMF mode for the endpoint is not affected by this CAC entry.
		• maybe —The inband DTMF tones are used by the endpoint unless signaling indicates that an alternative format for DTMF is in use.
		• never —The endpoint never uses inband DTMF.

	Command or Action	Purpose
Step 12	action next-table goto-table-name	Configures the action to take when this routing entry is chosen.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action next-table xcode</pre>	• <i>goto-table-name</i> —Specifies the next routing table to be processed when an event matches the entry.
Step 13	complete	Completes the CAC policy set when you have committed the full set.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# complete	
Step 14	cac-policy-set global policy-num	Activates the global CAC policy set. The CAC policy set must be in a complete state before it can be assigned as the default policy.
	Router(config-sbc-sbe)# cac-policy-set global 4	• <i>policy-num</i> —The call policy set number, ranging from 1 to 2147483647. The policy set must be in a complete state before it can be assigned as the default policy.
Step 15	end	Exits the CAC configuration mode and returns to privileged EXEC mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # end</pre>	

Configuring the Unified SBC to Support AMR-WB

This section explains how to configure the Unified SBC to support AMR-WB.

SUMMARY STEPS

- 1. configure terminal
- 2. dspfarm profile profile-identifier transcode
- 3. codec amr-wb
- 4. **sbc** *sbc*-*name*
- 5. associate dspfarm profile profile-identifier
- 6. activate

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	dspfarm profile profile-identifier transcode	Enters the DSP farm profile configuration mode, and defines a profile for DSP farm services.
	Example: Router(config)# dspfarm profile 20 transcode	
Step 3	codec amr-wb	Specifies the AMR-WB codec in the DSP farm profile.
	Example: Router(config-dspfarm-profile)#codec amr-wb	
Step 4	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mySBC dbe	
Step 5	associate dspfarm profile profile-identifier	Associates a DSP farm profile to a Cisco Call Manager group.
	<pre>Example: Router(config-sbc-dbe))# associate profile 20</pre>	
Step 6	activate	Initiates the DBE service of the SBC.
	Example: Router(config-sbc-dbe)# activate	

Configuration Examples of the SPA DSP Services for the SBC

This section contains the following examples:

- Example: Enabling DSP Farm Service on the SPA DSP, page 41-34
- Example: Configuring a DSP Farm Profile, page 41-35
- Example: Viewing a DSP Farm Profile Configuration and Status, page 41-35

Example: Enabling DSP Farm Service on the SPA DSP

The following example shows how to enable DSP farm services on the SPA DSP:

```
enable
configure terminal
voice-card 0/2
dsp services dspfarm
end
```

Configuration Examples of the SPA DSP Services for the SBC

Example: Configuring a DSP Farm Profile

The following example shows how to configure a DSP farm profile:

```
enable
configure terminal
dspfarm profile 1 transcode
description enables transcoding
   codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g723r63
codec ilbc
 codec gsmamr-nb
codec g726r32
codec g729br8
associate application sbc
maximum session 300
end
```

Example: Viewing a DSP Farm Profile Configuration and Status

After a DSP farm profile is created, use the show command to display a DSP farm profile configuration and status. The following examples show the output of the show commands:

```
Router# show running-config
voice-card 2/0
no dspfarm
dsp services dspfarm
!
dspfarm profile 20 transcode
codec g711ulaw
codec g711alaw
codec g729r8
codec g729ar8
codec g729br8
codec g729abr8
rsvp
maximum sessions 5
associate application SBC
1
Router# show dspfarm profile 20
Dspfarm Profile Configuration
Profile ID = 20, Service = TRANSCODING, Resource ID = 1
Profile Description :
Profile Admin State : UP
Profile Operation State : ACTIVE
Application : SBC Status : ASSOCIATED
Resource Provider : FLEX_DSPRM Status : UP
Number of Resource Configured : 5
Number of Resource Available : 5
Codec Configuration
Codec : g729abr8, Maximum Packetization Period : 60
Codec : g711alaw, Maximum Packetization Period : 30
Codec : g711ulaw, Maximum Packetization Period : 30
Codec : g729r8, Maximum Packetization Period : 60
```

```
Codec : g729ar8, Maximum Packetization Period : 60
Codec : g729br8, Maximum Packetization Period : 60
RSVP : ENABLED
I.
Router# show dspfarm all
DSPFARM Configuration Information:
Admin State: UP, Oper Status: ACTIVE - Cause code: NONE
Transcoding Sessions: 0(Avail: 0), Conferencing Sessions: 2 (Avail: 2)
Trans sessions for mixed-mode conf: 0 (Avail: 0), RTP Timeout: 600
Connection check interval 600 Codec G729 VAD: ENABLED
Total number of active session(s) 0, and connection(s) 0
SLOT DSP CHNL STATUS USE TYPE SESS-ID CONN-ID PKTS-RXED PKTS-TXED
0 0 1 UP FREE conf - -
0 0 2 UP FREE conf- -
      3 UP FREE conf - -
0
   0
0
   0
      4
          UP FREE conf - -
0
   0
      5
          UP FREE conf - -
0
  0
      6 UP FREE conf - -
```

Configuration Examples of Unified SBC

This section contains the following examples:

- Example: Associating the Unified SBC with a DSP Farm Profile, page 41-36
- Example: Configuring the Unified SBC to Enable Transcoding, page 41-36
- Example: Configuring the Unified SBC to Enable Transrating, page 41-37
- Example: Configuring the Unified SBC to Enable SRTP and Transcoding, page 41-38
- Example: Configuring the Unified SBC for In-Band DTMF Interworking, page 41-38
- Example: Configuring the Unified SBC to Support AMR-WB, page 41-39

Example: Associating the Unified SBC with a DSP Farm Profile

The following example shows how to associate the Unified SBC with a DSP farm profile:

```
enable
configure terminal
sbc mySBC
associate dspfarm profile 1
end
```

Example: Configuring the Unified SBC to Enable Transcoding

The following example shows how to configure the unified SBC to enable transcoding.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

```
enable
configure terminal
 sbc mySBC
  sbe
  cac-policy-set 1
  first-cac-scope dst-adjacency
  first-cac-table codec-dst-acc
  cac-table codec-dst-acc
  table-type limit dst-adjacency
   entry 1
   match-value nava
   caller-codec-list PCMU
   callee-codec-list PCMA
   media police strip
   action cac-complete
   complete
   cac-policy-set global 1
codec-list PCMU
codec PCMU
exit
codec-list PCMA
codec PCMA
exit
end
```

Example: Configuring the Unified SBC to Enable Transrating

```
<u>Note</u>
```

Transrating is supported only for different rates using the same codec, not across codecs. Therefore, transrating and transcoding cannot be performed simultaneously.

The following example shows how to configure the unified SBC for enabling the transrating feature using the same codec policy:

```
enable
configure terminal
 sbc mySBC
  she
  cac-policy-set 2
  first-cac-table Transrate
  cac-table Transrate
  table-type policy-set
  entry 1
   cac-scope call
   callee ptime 30
   caller ptime 20
   media police strip
   action cac complete
   complete
  cac-policy-set global 2
   end
```

The following example shows how to configure the Unified SBC for enabling the transrating feature using the same codec policy:

enable configure terminal sbc MySBC sbe codec list PCMU

Γ

```
policy transrating
codec PCMU packetization-period 200 priority 1
end
```

Example: Configuring the Unified SBC to Enable SRTP and Transcoding

The following example shows how to configure SBC to enable the SRTP and transcoding features.

enable configure terminal sbc mySBC sbe cac-policy-set 3 first-cac-table C3 cac-table c3 table-type policy-set entry 1 cac-scope global srtp support allow srtp caller mandate srtp callee mandate srtp interworking allow srtp media interworking allow action next-table xcode exit exit cac-table xcode table-type limit dst-adjacency entry 1 match-value nav4b callee-codec-list PCMU action cac-complete complete cac-policy-set global 3 end

Example: Configuring the Unified SBC for In-Band DTMF Interworking

The following example shows how to configure the unified SBC for inband DTMF transmission.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

```
enable
configure terminal
sbc mySBC
sbe
cac-policy-set 4
first-cac-table c4
cac-table c4
table-type policy-set
entry 1
cac-scope global
callee inband-dtmf-mode always
caller inband-dtmf-mode never
```

```
action next-table xcode
exit
exit
cac-table xcode
table-type limit dst-adjacency
entry 1
match-value spab
callee-codec-list PCMU
action cac-complete
complete
cac-policy-set global 4
end
```

Example: Configuring the Unified SBC to Support AMR-WB

The following example shows how to configure the Unified SBC to support AMR-WB:

```
enable
configure terminal
sbc mySBC
sbe
cac-policy-set 1
first-cac-scope dst-adjacency
first-cac-table codec-dst-acc
cac-table codec-dst-acc
table-type limit dst-adjacency
entry 1
match-value nava
caller-codec-list AMRWB
callee-codec-list PCMA
media police strip
action cac-complete
complete
cac-policy-set global 1
codec-list AMRWB
codec AMR-WB
exit
codec-list PCMA
codec PCMA
exit
```







Tracking Policy Failure Statistics

Users can track the number of calls that Cisco Unified Border Element (SP Edition) rejected based on the rules established in the number analysis policies, routing policies, or Call Admission Control (CAC) policies. Users can also view and query the policy failure statistics associated with these rejected calls, which can help them determine whether changes need to be made to the existing policies.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Feature History for Policy Failure Statistics

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for Tracking Call Policy Failure Statistics, page 42-2
- Information About Policy Failure Statistics, page 42-2

Restrictions for Tracking Call Policy Failure Statistics

Review the following restrictions for policy failure statistics:

- Only new call failures are tracked by this feature.
- Only call failures associated with local policy are recorded. Calls rejected by downstream signaling devices are not included in this statistics.

Information About Policy Failure Statistics

The section provides information on the following:

- Policy Failure Statistics for a Specified Time Interval, page 42-2
- Policy Set and Per-Entry Statistics, page 42-2
- Automatic Tracking of Policy Failure Statistics, page 42-3
- Policy Failure Statistics and Hunting, page 42-4

Policy Failure Statistics for a Specified Time Interval

Table 42-1 lists the command to view and clear the failure statistics on the specified signaling border element (SBE) for a certain time interval.

Table 42-1	Commands for Time-Based Policy Failure Statistics
------------	---

clear sbc service-name sbe policy-failure-stats	Clears the policy failure statistics for the
	current and previous time interval.

Policy Set and Per-Entry Statistics

To determine whether calls failed due to policies configured in the routing, number validation, or CAC tables, users can view the policy failure statistics for a specific policy table or table entry. Table 42-2 lists the commands to view and clear the statistics on policy tables associated with a policy set.

Table 42-2	Commands for Statistics for PolicyTables in a Policy Set
------------	--

show sbc service-name sbe cac-policy-set policy set-id tables	Displays a summary of the CAC policy tables associated with the given policy set, including the number of failed calls.
clear sbc service-name sbe cac-rejection-stats	Clears all CAC policy failure statistics.
show sbc <i>service-name</i> sbe call-policy-set <i>policy set-id</i> tables	Displays a summary of routing policy tables associated with the given policy set, including the number of failed calls.
clear sbc service-name sbe call-rejection-stats	Clears all routing and number analysis policy rejection statistics.
show sbc service-name sbe cac-policy-set policy set-id table name entries	Displays the specified entire CAC policy table.

Table 42-3 lists the commands to view the detailed information for a specific entry in a CAC policy table and routing table.

Table 42-3Per-Entry Statistics Commands

show sbc service-name sbe cac-policy-set policy set-id table name entries	Displays a summary of the entries associated with the given CAC table.
show sbc service-name sbe call-policy-set policy set-id table name entry entry	Displays detailed statistics for the given entry in the routing table.
show sbc service-name sbe call-policy-set policy set-id table name entries	Displays a summary of the entries associated with the given routing table.

Automatic Tracking of Policy Failure Statistics

Cisco Unified Border Element (SP Edition) automatically tracks policy failure statistics for call attribute sets representing the following:

- · Per source adjacency statistics for all configured adjacencies
- Per destination adjacency statistics for all configured adjacencies
- Per source account statistics for all configured accounts
- Per destination account statistics for all configured accounts

Table 42-4 lists the commands to view and clear automatically tracked policy failure statistics.

Table 42-4 Automatically Tracked Statistics Commands

show sbc service-name sbe policy-failure-stats src-adjacency table-name period	Displays the policy statistics of the specified source adjacency for the specified time interval. The value for the <i>period</i> parameter must be one of the following:
	• current5mins
	• previous5mins
	• current15mins
	• previous15mins
	• currenthour
	• previoushour
	• currentday
	• previousday
clear sbc service-name sbe policy-failure-stats src-adjacency table-name	Clears the policy statistics of the specified source adjacency.

The statistics are collected at 5 minute intervals past the hour (that is, at 0, 5, 10, 15 minutes, and so on past the hour). For example, the periods covered by the various buckets at 12:43 would be as follows:

- current five minutes: 12:40-12:43
- previous five minutes: 12:35-12:40

- current 15 minutes: 12:30-12:43
- previous 15 minutes: 12:15-12:30
- current hour: 12:00-12:43
- last hour: 11:00-12:00
- current day: 00:00-12:43
- last day: 00:00-24h 00:00

A counter can keep increasing. It keeps a count of events that have completed. When reporting the value of a counter, it's the sum total of events that happened in the period. Some examples of counters are total call attempts, failed call attempts, and active call failures.

A gauge is a counter that can go up and down. It typically tells you how many of something there are now. When reporting the value of a gauge, it's either the current value, or when measured over a longer time period, it's the average value during the period. Some examples of gauges are active calls, activating calls, and deactivating calls.

Policy Failure Statistics and Hunting

If the CAC module refuses a call or if a call cannot be signaled to the chosen destination adjacency because of a negative or no response, call hunting occurs. Call hunting is the process of selecting an alternative adjacency from the routing tables and retrying the call using the newly selected destination adjacency.

Hunting continues until one of the following conditions is fulfilled:

- The call gets connected.
- No further adjacencies are available for retry.
- The call has been hunted too many times.

Global Statistics and Call Hunting

If a call gets connected after hunting, Cisco Unified Border Element (SP Edition) does not include it in any of the following global statistics:

- Total call setup failures
- Total call setups failed due to number analysis
- Total call setups failed due to routing
- Total call setups failed due to CAC
- CAC failure due to number of calls limit
- CAC failure due to call rate limit
- CAC failure due to media channels limit
- CAC failure due to bandwidth limit

If a call fails after number analysis, hunting does not occur. Cisco Unified Border Element (SP Edition) includes it in the following global statistics:

- Total call setup failures
- Total call setups failed due to number analysis
If a call fails the first time it is routed because no destination adjacency is found in the routing table, then Cisco Unified Border Element (SP Edition) includes it in the following global statistics:

- Total call setup failures
- Total call setups failed due to routing

If a call fails because a CAC policy refused it permission to proceed, Cisco Unified Border Element (SP Edition) includes the failure in the total call setups failed due to CAC statistics. Additionally, the call is included in one of the following statistics depending on the nature of the CAC limit:

- CAC failure due to number of calls limit
- CAC failure due to call rate limit
- CAC failure due to media channels limit
- CAC failure due to bandwidth limit

Per-table and Per-entry Statistics and Call Hunting

If a call undergoes N iterations of hunting, then it traverses the number analysis tables once, and the routing and the CAC tables N times. But the CAC tables can reject the call each time it traverses the CAC table. For each time the CAC table rejects the call, Cisco Unified Border Element (SP Edition) finds the table and entry that was responsible for setting the CAC limit, and increments the following:

- Number of calls refused by the CAC table
- Number of calls refused by the table entry

Per-adjacency and Per-Account Statistics and Call Hunting

If a call gets connected after hunting, Cisco Unified Border Element (SP Edition) does not include it in the following per-account or per-adjacency statistics:

- total call setup failures
- total call setups failed due to number analysis
- total call setups failed due to CAC
- CAC failures due to rate limit
- CAC failures due to media channels limit
- CAC failures due to bandwidth limit

If a call fails due to number analysis, then hunting does not occur and Cisco Unified Border Element (SP Edition) includes the call in the following per-account and per-source adjacency statistics:

- total call setup failures
- total call setups failed due to number analysis

If a call fails in the routing tables before hunting occurs, Cisco Unified Border Element (SP Edition) includes the call in the following per-source account and per-source-adjacency statistics:

- total call setup failures
- total call setups failed due to routing

A call included in the total call setup failures statistics is included in the per-source adjacency, per-destination-adjacency, per-source-account adjacency, and per-destination account statistics. Additionally, if the most recent hunting attempt failed because a CAC policy refused the call permission to proceed, Cisco Unified Border Element (SP Edition) includes the failure in the total call setups failed

due to CAC statistics in the per-source-adjacency, per-destination-adjacency, per-source-account, and per-destination-account statistics. The call is also included in one of the following statistics depending on the nature of the CAC limit depending on the nature of the CAC limit:

- CAC failure due to number of calls limit
- CAC failure due to call rate limit
- CAC failure due to media channels limit
- CAC failure due to bandwidth limit



Implementing SNMP

Simple Network Management Protocol (SNMP) for Cisco Unified Border Element (SP Edition) is defined in the following MIBs:

- CISCO-SESSION-BORDER-CONTROLLER-EVENT-MIB—Defines SNMP notifications and alarms that are generated by Cisco Unified Border Element (SP Edition). This MIB sends the notifications and traps that are generated by Cisco Unified Border Element (SP Edition) to the SNMP manager.
- CISCO-SESSION-BORDER-CONTROLLER-STATS-MIB—Defines the SNMP statistics information for Cisco Unified Border Element (SP Edition). The two types are call statistics and media statistics. The calls are categorized as Session Initiation Protocol (SIP) calls and H.323 calls; the media statistics refer to RTP.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Implementing SNMP

Release	Modification
Cisco IOS XE Release 2.4	Implementing Simple Network Management Protocol (SNMP) was introduced on the Cisco IOS XR.

Contents

- Prerequisites for Implementing SNMP, page 43-2
- Information About Implementing SNMP, page 43-2
- Implementing SNMP for Cisco Unified Border Element (SP Edition), page 43-3
- Configuration Example for Implementing SNMP, page 43-5

Prerequisites for Implementing SNMP

The following prerequisites are required to implement SNMP for Cisco Unified Border Element (SP Edition):

- You must have sufficient user privileges to modify the running configuration of the router.
- You must have configured Cisco Unified Border Element (SP Edition).
- Before you can access the SNMP MIBs to perform SNMPv2 polling for the SBC MIB statistics or to configure the SNMP users and groups for SNMPv3 polling, you must configure the SNMP read-only community string by using the **snmp-server community** command in Cisco IOS XR. For more information about the **snmp-server community** command, see the *Cisco IOS Network Management Command Reference*.

Information About Implementing SNMP

This section describes how to implement SNMP for SBC:

- SNMP Notifications, page 43-2
- SNMP Statistics, page 43-3

SNMP Notifications

Table 43-1 lists the types of SNMP notifications.

Туре	Description
Source Alert Notification	Specifies that the media is received from an unexpected source.
Blacklist Notification	Adds or removes the source from the blacklist table.
Adjacency Status Notification	Attaches or detaches the adjacency from the SBC.
SLA Violation Notification	Specifies that the violations of the Service Level Agreement (SLA) are described in the policy tables. SLAs can include the maximum number of calls allowed, maximum call rate, maximum bandwidth, and so forth.
Radius Connection Status Notification	Specifies that the SBC connection to a RADIUS server is either lost or restored.
H.248 Controller Status	Specifies if an H.248 controlled entity, either a data border element (DBE) or remote transcoder, is connected or detached from the SBC. See other sections of the documentation for supported H.248 controlled entities.

SNMP Statistics

Table 43-2 lists the types of SNMP statistics.

Table 43-2 List of SNMP Statistics

Туре	Description	
Global call statistics	Represents the global call-related statistics such as call rates, media flows, signaling flows, and so forth. The show sbc dbe media-stats command displays output for the global call statistics.	
Periodic statistics	Represents the information for the SBC call statistics for a particular time interval such as current 5 minutes, previous 5 minutes, current 15 minutes, previous 15 minutes, current hour, and previous hour. The show sbc sbe call-stats command displays output for the periodic statistics.	
Per flow statistics	Represents the SBC media flow statistics. These media statistics are used for each of the current ongoing call flows. The show sbc dbe media-flow-stats command displays output for the per-flow statistics.	
H.248 statistics	Represents the information for the H.248 call-related statistics when the H.248 controller is associated with SBC. The show sbc dbe controllers command displays output for the H.248 statistics.	

Implementing SNMP for Cisco Unified Border Element (SP Edition)

This section describes how to implement SNMP for Cisco Unified Border Element (SP Edition):

• Configuring SNMP Notifications, page 43-3

Configuring SNMP Notifications

Perform this task to configure SNMP notifications for Cisco Unified Border Element (SP Edition).

SUMMARY STEPS

- 1. configure terminal
- 2. snmp-server enable traps sbc {adj-status | blacklist | h248-ctrlr-status | qos-statistics | radius-conn-status | sla-violation | source-alert}
- 3. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	<pre>smmp-server enable traps sbc {adj-status blacklist congestion-alarm h248-ctrlr-status media-source qos-statistics radius-conn-status sla-violation source-alert) Example: Router(config)# snmp-server enable traps sbc blacklist </pre>	Specifies the SBC notification type to be enabled.
		• Use the adj-status keyword to enable the SNMP SBC Adjacency Status trap when an adjacency is attached to or detached from the SBE.
		• Use the blacklist keyword to enable the SNMP SBC Dynamic Blacklist trap when a source is added or removed from the blacklist table.
		• Use the congestion-alarm keyword to enable the SNMP SBC Congestion Alarm trap.
		• Use the h248-ctrlr-status keyword to enable the SNMP SBC H.248 Controller Status trap. For a distributed deployment model, a DBE is attached or detached from the SBC.
		• Use the media-source keyword to enable the SNMP SBC Media Source Alert traps.
		• Use the qos-statistics keyword to enable the QoS statistics traps. See the ?\$paranum>Implementing QoS Demarcation? section on page 38-10 for more information about the procedure.
		• Use the radius-conn-status keyword to enable the SNMP SBC Radius Connection Status trap when the connection is changed for the RADIUS server.
		• Use the sla-violation keyword to enable the SNMP SBC SLA Violation trap when there is an SLA violation in the policy tables. SLAs include the maximum number of calls allowed, maximum call rate, maximum bandwidth, and so on.
		• Use the source-alert keyword to enable the SNMP SBC Source Alert trap when media is received from an unexpected source.
		See the ?\$paranum>Alarm Logs? section on page 44-6 for information about configuring logging for some of these alarms.
Step 3	end	Exits the configuration command.
	Example: Router(config)# end	

Configuration Example for Implementing SNMP

This section provides the following configuration example for implementing SNMP for Cisco Unified Border Element (SP Edition):

• Configuring SNMP Notifications: Example, page 43-5

Configuring SNMP Notifications: Example

The following example shows how to configure the SNMP blacklist notification for Cisco Unified Border Element (SP Edition):

```
configure terminal
snmp-server enable traps sbc blacklist
end
```





Logging Support

Cisco Unified Border Element (SP Edition) provides various features for working with logs. Logging can be configured so that logs are generated under specified conditions. Logs can also be generated on demand. Information derived from the logs can be used for analyzing and troubleshooting issues pertaining to the operation of the network and for identifying areas for improvement in the network.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to *Cisco Unified Border Element* (SP Edition) Command Reference: Unified Model at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Release	Modification
Cisco IOS XE Release 2.x	The Syslog feature was introduced in a release earlier than Cisco IOS XE Release 3.1S.
Cisco IOS XE Release 3.5S	The Call Log Correlation feature was introduced to enable all the correlation logs associated with a particular call to be linked together using a correlator ID.
	The Alarms feature was enhanced to include new features for working with alarm logs.

Feature History for Logging Support

Contents

This chapter contains the following sections:

- Syslog Capabilities, page 44-2
- Call Log Correlation, page 44-4
- Alarm Logs, page 44-6

Syslog Capabilities

All the Cisco Unified Border Element (SP Edition) debug messages that are displayed on the console are recorded in the Cisco IOS syslog. All the Cisco IOS syslog commands that configure log size, persistence, and redirection can be used for managing the syslog.

In addition to the console messages, Cisco Unified Border Element (SP Edition) records a log in its own internal buffer. This is known as the problem determination log and is saved in the event of a software-forced reload or as a result of using the **sbc dump-diagnostics** command. When you compile the problem reports, the problem determination log file is included as part of the problem reports.

Internal Log Levels

The Session Border Controller (SBC) application uses an internal log level to control the verbosity of the console and the PD log. Although both the console and problem determination log levels can be changed independently, we do not recommend changing the problem determination log level because the problem determination log buffer is of limited size and important logs may be lost.

The default SBC problem determination logging level is 63 for the console and 60 for the buffer. You can change the default SBC problem determination logging level using the **debug sbc log-level console** command, the **debug sbc log-level filter** command, or the **debug sbc log-level buffer** command.

Log Level	Syslog Level
90	Fatal
80	Error
70	Unexpected
63	Configuration Error
60	Operational
50	Audit
40	Statistics
30	Verbose Operational
20	Verbose Statistics
10	Internal Diagnostic

Enabling the Syslog Functionality

To enable the syslog functionality on the SBC, set the internal log levels, and issue the syslog-specific logging commands. The following example assumes a default problem determination level of 63 (no further action is needed if this is a fresh reboot).

1. Enable logging using the following commands:

```
Router# configure
Router(config)# logging enable
Router(config)# logging standby
```

The logging standby command allows the synchronization of the active and standby syslog Note settings. 2. Configure the location to which you want the syslog messages to be sent. Locations can be one of the following: - Console: logging console <1-7> Router(config) # logging console severity-level - Buffer: logging buffer <1-7> Router(config) # logging buffered severity-level Note Use the **show logging** command to view the logging statistics and the logging buffer. Use the clear logging command to clear the logging buffer. - Syslog server: logging trap <1-7> Router(config) # logging host ip_address [tcp[/port] | udp[/port]] Router(config) # logging trap severity-level Router(config) # logging device-id {hostname | ipaddress interface_name | string text | context-name} Router(config) # logging facility number Note The logging device-id command allows the customization of syslog messages when sending the log to a remote server.

- Telnet sessions: logging monitor <1-7>

```
Router(config)# logging monitor severity-level
Router# terminal monitor
```

- SNMP management station: logging history <1-7>

```
Router(config) # logging history severity-level
```

Supervisor: logging supervisor <1-7>

Router(config) # logging supervisor severity-level

3. Configure specific syslog message manipulation:

```
Router(config)# logging message syslog_id [level severity_level]
Router# show logging message
Router# clear logging
```

4. Configure the global syslog settings:

```
Router(config)# logging queue queue-size
Router# show logging queue
Router(config)# logging timestamp
Router(config)# logging rate-limit {num {interval | level severity_level |
message syslog_id} | unlimited {level severity_level | message syslog_id}}
Router# show logging
```

Call Log Correlation

The Call Log Correlation feature enables all the correlation logs associated with a particular call to be linked together using a correlator ID. This feature also enables real-time filtering of logs on a particular call. A 64-bit diagnostics correlator is assigned to each SIP call, REGISTER, SUBSCRIBE, or NOTIFY messages.

You can set the filters based on the following parameters:

- Dialed or dialing number
- Session Initiation Protocol (SIP) Universal Resource Identifier (URI)
- Remote signaling address
- Remote VPN ID
- Adjacency
- VRF

The logs that match the selected filter type are saved in a separate problem determination trace file and inter process signal (IPS) trace file.

Use the following command to enable the correlation-logs filter:

debug sbc sbc-name correlation-logs filter filter-name [pdtrc-log-level value]

Use the following command to disable the correlation-logs filter:

no debug sbc sbc-name correlation-logs filter filter-name

Use the following command to display the debug logs, filters, and log levels:

show debugging

Problem Determination Log Levels

You can set the problem determination log level in the filter using the **pdtrc-log-level** option in the **debug sbc** *sbc-name* **correlation-logs filter** *filter-name* **[pdtrc-log-level** *value*] command. The problem determination trace log level ranges from 0 to 100. The default log level is 60. A log level of 100 indicates that no logs are output, and 0 indicates that all the logs are output.

Table 44-1 lists the problem determination log levels:

Problem Determination Log Level	Description
90	Critical system errors
80	Major system errors
70	Minor system errors
63	Configuration errors
60	Call errors
55	Call overview
50	Call details
40	Call statistics
30	Verbose operational
20	Verbose statistics
10	Internal diagnostic

Table 44-1 Problem Determination Log Levels

Examples of Call Log Correlation Feature

The following example shows the various filters available for filtering the correlation logs:

Router# debug sbc test correlation-logs filter ?

adjacency	Adjacency, matching calls to or from this adjacency
dn	Dialed/dialing number, matching calls to or from this number
remote-signalling-address	Remote signalling address matching to or from this address
sip-uri	SIP-URI, matching calls to or from this URI
vrf	VRF name

The following example shows the filtering of correlation logs based on the adjacency parameter:

```
Router# debug sbc test correlation-logs filter adjacency abc
Debugging filter log-level set to default level 60
Router# show debugging
SBC correlator filter Adjacency name is abc
IpsTracing is enabled
```

The following example shows the filtering of correlation logs based on the dialing number parameter:

Router# debug sbc test correlation-logs filter dn aa

Debugging filter log-level set to default level 60

Router# show debugging

```
SBC correlator filter DN is aa IpsTracing is enabled
```

The following example shows the filtering of correlation logs based on the remote signalling address parameter:

```
Router# debug sbc test correlation-logs filter remote-signalling-address ipv4 192.0.2.1
```

Debugging filter log-level set to default level 60

Router# show debugging

```
SBC buffer log-level is 0
SBC correlator
Filter Remote signalling-address ipv4 address is 192.0.2.1
IpsTracing is enabled
SBC correlator
Filter DN is abc
Pd loglevel is 70
IpsTracing is enabled
```

The following example shows the filtering of correlation logs based on the SIP URI parameter:

```
Router# debug sbc test correlation-logs filter sip-uri ccc
Debugging filter log-level set to default level 60
```

Router# show debugging

SBC correlator filter Adjacency name is abc IpsTracing is enabled SBC correlator filter Remote signalling-address ipv4 address is 192.0.2.1 IpsTracing is enabled SBC correlator filter SIP-URI is ccc IpsTracing is enabled SBC correlator filter DN is aa IpsTracing is enabled

The following example shows the filtering of correlation logs based on the VRF parameter:

Router# debug sbc test correlation-logs filter vrf new ipv4 rsa 192.0.2.1 pdtrc-log-level 70

Debugging filter log-level set to default level 60

Router# show debugging

```
SBC correlator Filter Remote signalling-address ipv4 address is 192.0.2.1
SBC correlator Filter VRF is new with Vpn(id) = 3
Pd loglevel is 70
IpsTracing is enabled
SBC correlator Filter SIP-URI is 9.0.0.0
Pd loglevel is 0
IpsTracing is enabled
```

Alarm Logs

You can configure the SBC to generate alarms for various types of events associated with the operation of the SBC. You can also configure the SBC to log debugging information, which you can use to monitor and tune the functioning of the system. On the basis of the alarms, you can take corrective and preventive action to ensure that the SBC continues functioning according to your business requirements. It is also

important to monitor the alarms generated by the SBC over a period of time and analyze this information. To address this requirement, you can configure the SBC to generate, display, and store alarm logs. The information provided in the alarm logs can help resolve some common issues, such as interoperability problems and incorrect configurations. These logs can also be used to identify issues that might potentially require escalation and investigation by more specialized support staff. Information in the logs can be used to improve the overall efficiency of the system.



All alarm log information is lost after a route processor failover.

You can use any combination of the following commands to configure alarm logs:

- Use the **debug sbc alarm-filter** command to specify the alarm types for which alarm logs must be generated.
- Use the **debug sbc alarm-log-level** command to specify the output mode and the alarm severity level for which alarms must be logged.
- The buffer that is used to store alarm logs may run out of free space while log files are written to it. In addition, you may want to store alarm logs for future reference. Use the **sbc periodic-dump-alarms** command to configure periodic movement of alarm log files from the buffer to a file system.
- Use the **sbc dump-alarms** command to move the alarm logs from the buffer to either a file system that you specify or the default file system configured on the router.

Configuring Alarm Logs

This task explains the commands that you can use to configure alarm logs. Note that it is not mandatory to use any particular command described in this task. You can use any combination of these commands to configure alarm logs.

SUMMARY STEPS

- 1. debug sbc sbc-name alarm-filter alarm-type
- 2. debug sbc sbc-name alarm-log-level [buffer | console] severity-level
- 3. sbc periodic-dump-alarms {dump-location file-system [time-period time-period] | time-period time-period}
- 4. sbc dump-alarms [file-system]
- 5. show debugging

DETAILED STEPS

	Command or Action	Purpose
Step 1	debug sbc sbc-name alarm-filter alarm-type	Configures the alarm types for which alarm logs must be generated.
	Example:	• <i>sbc-name</i> —Name of the SBC.
	Router# debug sbc MySbc alarm-filter audit-congestion	• <i>alarm-type</i> —One of the following alarms:
		- audit-congestion—Call audit congestion.
		- blacklist-alert—Blacklist alert.
		- blacklist-event—Blacklist event.
		- h248—H248 connection failed.
		- handled-exception—Handled exception.
		 routing-component—Routing component set not active.
		- routing-config—Routing config set not active.
		- routing-invalid—Invalid routing configuration.
		- sip-congestion—SIP congestion detection.
		- sip-peer —SIP peer unavailable.
		 vqm—Voice Quality metrics (VQM) threshold exceeded.
Step 2	<pre>debug sbc sbc-name alarm-log-level [buffer console] severity-level</pre>	Configures the output mode and the alarm severity level for which alarms must be logged.
		• <i>sbc-name</i> —Name of the SBC.
	Example: Router(config)# debug sbc MySbc alarm-log-level console 40	• buffer —Specifies that alarm logs must be stored in the buffer.
		Note The size of a single log file created on the file system cannot exceed 2 MB. When the size of a particular log file reaches 2 MB, a new file is created and logging output is stored in the new file.
		• console —Specifies that logging output must be displayed on the console.
		• <i>severity-level</i> —Alarm severity level for which logs must be generated. The range is from 0 to 100. For alarm logs stored in the buffer, the default is 40. For alarm logs displayed on the console, the default is 80. To disable logging, set the value to 100. If you set the value to 0, logs are generated for all levels of alarm severity.

	Command or Action	Purpose
Step 3	<pre>sbc periodic-dump-alarms {dump-location file-system [time-period time-period] time-period time-period}</pre>	Configures periodic movement of alarm log files from the buffer to a file system.
	Example: Router(config-sbc)# sbc periodic-dump-alarms dump-location bootflash: time-period 120	• dump-location —Specifies that you want the alarm logs to be stored in a file system.
		• <i>file-system</i> —Name of the file system where you want the alarm logs to be moved. For example, <i>file-system</i> can be one of the following:
		– bootflash:
		– flash:
		– fpd:
		– ftp:
		– http:
		– https:
		– obfl:
		– pram:
		– rcp:
		– scp:
		– tftp:
		• time-period <i>time-period</i> —Specifies the periodic time interval, in minutes, after you want the logs to be moved. The range is from 0 to 1440. The default is 60.
		Note When you run the no form of this command, the time period for moving logs is set to 0 and periodic movement of the logs is disabled.

	Command or Action	Purpose
Step 4	<pre>sbc dump-alarms [file-system] Example: Router(config-sbc-sbe-cacpolicy)# sbc dump-alarms bootflash:</pre>	 Moves alarm logs from the buffer to either a file system that you specify or the default file system configured on the router. <i>file-system</i>—Name of the file system where you want the alarm logs to be moved. For example, <i>file-system</i> can be one of the following:
		– flash:
		– fpd:
		– ftp:
		– http:
		– https:
		– obfl:
		– pram:
		– rcp:
		– scp:
		– tftp:
Step 5	show debugging	Displays information about the types of debugging that are enabled on the router.
	Example: Router# show debugging	The output of this command includes debugging settings created by running the debug sbc alarm-filter command and the debug sbc alarm-log-level command.

The following sample output of the **show debugging** command shows the debugging settings created by running the **debug sbc alarm-filter** command and the **debug sbc alarm-log-level** command. In this example, these debug commands have been used to specify that logs must be generated for call audit congestion alarms that are of severity level 60 or higher and that these logs must be moved to the specified file system at 120-minute intervals:

```
Router# show debugging
```

```
SBC:
SBC buffer alarm-log-level : 60
SBC alarm filter 1 : AUDIT CONGESTION
SBC alarm periodic dump time : 120 min
```



SIP 3xx Redirect Responses

This section describes how Cisco Unified Border Element (SP Edition) can be configured to process Session Initiation Protocol (SIP) 3xx responses. 3xx is a class of the response code used in SIP to indicate that further action needs to be taken in order to complete the request. The sender of the request should retry the request, using one or more alternative Uniform Resource Identifiers (URIs), which are presented in the 3xx response.



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP 3xx Redirect Responses

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Information About 3xx Redirect Responses in SIP, page 45-2
- How to Configure Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses, page 45-3
- Examples of Configuring Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses, page 45-5

Information About 3xx Redirect Responses in SIP

This section contains the following subsections:

- 3xx Responses, page 45-2
- Diversion Headers, page 45-3

3xx Responses

3xx responses are usually only expected in session-initiating requests, INVITEs. However, the SIP specification does not preclude sending 3xx responses for other request types. A number of alternative URIs are supplied on the 3xx responses in Contact headers.

The 3xx class of responses includes any response code in the range of 300-399 and indicates a redirection of the call. The redirection requires further action to be taken to complete the request. The following 3xx response codes are defined in SIP.

- **300 Multiple Choices.** The address in the request resolved to several choices, each with its own specific location. The user or user agent (UA) can select a preferred communication end point and redirect the request to that location.
- The response may include a message body containing a list of resource characteristics and location(s), from which the user or UA can choose the most appropriate one, if allowed by the Accept request header field. However, no MIME types have been defined for this message body.
- The choices should also be listed as Contact fields. The response may contain several Contact fields or a list of addresses in a Contact field. UAs may use the Contact header field value for automatic redirection or ask the user to confirm a choice.
- **301 Moved Permanently.** The user can no longer be found at the address in the Request-URI, and the requesting client should retry at the new address given by the Contact header field. The requestor should update any local directories, address books, and user location caches with this new value, and redirect future requests to the addresses listed.
- **302 Moved Temporarily**. The requesting client should retry the request at the new address(es) given by the Contact header field. The Request-URI of the new request uses the value of the Contact header field in the response.

The duration of the validity of the Contact URI can be specified through an **Expires** header field or an **Expires** parameter in the Contact header field. Both proxies and UAs may cache this URI for the duration of the expiration time. If there is no explicit expiration time, the address is valid only once for recursing, and must not be cached for future transactions.

If the URI cached from the Contact header field fails, the Request-URI from the redirected request may be tried again only once.

• **305 Use Proxy**. The requested resource must be accessed through the proxy given by the Contact field. The Contact field gives the URI of the proxy. The recipient is expected to repeat this single request via the proxy. 305 responses must only be generated by the user agent servers (UASs).

380 Alternative Service. The call was not successful, but alternative services are possible. The alternative services are described in the message body of the response. There are no formats currently defined for this information.

In each case, the request should be retried to one of the supplied alternative URIs. The request can be retried by either the originating UA, or by an intermediate back-to-back user agent (B2BUA) or proxy on behalf of the originating UA (and without notifying it).

Cisco Unified Border Element (SP Edition) is a B2BUA, and, therefore, in some deployments it may be necessary for Cisco Unified Border Element (SP Edition) to retry the request instead of sending a negative response back to the initiator of the request.

Diversion Headers

The Diversion header enables the called SIP user agent to identify from whom the call was diverted and why it was diverted. The header notifies the original caller:

- That the call has been redirected to a destination that differs from the original target
- The number to which the call has been redirected
- The reason for the redirection

The diversion header is attached by networking elements that change the final destination of a request.

How to Configure Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses

This section contains the steps for configuring Cisco Unified Border Element (SP Edition) to process SIP 3xx responses.

Configuring Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. redirect-mode mode
- 6. attach
- 7. exit
- 8. redirect-limit limit
- 9. end
- 10. show sbc *sbc-name* sbe adjacencies
- 11. show sbc *sbc-name* sbe redirect-limit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip SipToIsp42	Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	redirect-mode mode	Configures the behavior of the SBC on receipt of a 3xx response to an INVITE from the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip) redirect-mode	• redirect-mode pass-through —SBC passes all 3xx responses back to the caller (the default mode).
	recurse	• redirect-mode recurse—On 300, 301, 302, and 305 INVITE responses (under the redirect-limit, see Step 8), SBC resends the INVITE to the first listed contact address. Otherwise, SBC passes 3xx responses back.
		• no redirect-mode —The no version of this command returns the adjacency to the default behavior.
Step 6	attach	Attaches the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# attach	
Step 7	exit	Exits the adjacency-sip mode and returns to the SBE mode.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# exit</pre>	
Step 8	redirect-limit limit	Configures the maximum number of redirections that the SBC performs on a given call.
	Example: Router(config-sbc-sbe)# redirect-limit 4	• redirect-limit <i>limit</i> —A numeric value, the maximum number of redirections performed before the call is failed (the range is 0-100, the default is 2).
		• no redirect-limit —The no version of this command returns the adjacency to the default behavior.

	Command or Action	Purpose
Step 9	end	Exits the SBE mode and returns to Privileged EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 10	show sbc sbc-name sbe adjacencies	Lists the adjacencies configured on SBEs.
	Example: Router# show sbc mysbc sbe adjacencies	
Step 11	show sbc sbc-name sbe redirect-limit	Displays the current limit on the maximum number of redirections a call can undergo.
	Example: Router# show sbc mysbc sbe redirect-limit	

Examples of Configuring Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses

This section provides two simple configurations for processing SIP 3xx responses.

The following command configures the adjacency "SipToIsp42" to recurse on 300, 301, 302, and 305 INVITE responses.

Router(config)# sbc mySbc sbe adjacency sip SipToIsp42 Router(config-sbc-sbe-adj-sip)# redirect-mode recurse Router(config-sbc-sbe-adj-sip)# end

The following command configures the SBE to perform maximum 4 SIP 3xx redirections per call.

Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# redirect-limit 4
Router(config-sbc-sbe)# end

Examples of Configuring Cisco Unified Border Element (SP Edition) to Process SIP 3xx Responses



SIP Call Hold

The Session Initiation Protocol (SIP) call hold feature in Cisco Unified Border Element (SP Edition) provides a standard telephony service of putting a caller on hold. If a party in a call wants to put the other party on hold, a party re-invites the other by sending an INVITE request with a modified Session Description Protocol (SDP). When a SIP endpoint wishes to place a call on hold or respond to a call hold re-INVITE, it chooses an appropriate method. Cisco Unified Border Element (SP Edition) modifies call hold SDPs to use any available methods in order to maximize inter-operating with SIP devices.

Note

For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Call Hold

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Information About SIP Call Hold in Cisco Unified Border Element (SP Edition), page 46-2
- How to Configure SIP Call Hold, page 46-2
- SDP Call Hold Interworking, page 46-3
- Configuration Examples, page 46-10

Information About SIP Call Hold in Cisco Unified Border Element (SP Edition)

Cisco Unified Border Element (SP Edition) accepts a SIP re-INVITE with an SDP, signaling that the sender wishes to put the call on hold. Cisco Unified Border Element (SP Edition) modifies the SDP offer as needed and replaces remote endpoint addresses with known data border element (DBE) media addresses. Cisco Unified Border Element (SP Edition) then forwards the SIP message, containing the modified SDP to the remote endpoint.

If the re-INVITE is rejected by the endpoint going on hold, then the error response is returned to the holding endpoint (the endpoint that initiated the call hold). The media gate on the DBE continues to be connected and media continues to flow as before.

How to Configure SIP Call Hold

This section contains the steps for configuring the "no media" timeout duration for on-hold calls.

Configuring SIP Call Hold

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. hold-media-timeout timeout
- 5. end
- 6. show sbc service-name sbe hold-media-timeout
- 7. show sbc service-name sbe calls

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure	Enables global configuration mode.
	Example: Router# configure	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.

	Command or Action	Purpose
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	hold-media-timeout timeout	The time an SBE will wait after receiving a media timeout notification from the DBE for an on hold call before tearing that call down.
	Example. Router(config-sbc-sbe)# hold-media-timeout 7200	• When the DBE detects that media has stopped on a call, it will start a timer for the specified duration, using the DBE media timeout command line interface (CLI) command.
		• If no media flows before this timer expires, then the DBE will send a pin-hole timeout event notification to the SBE.
		• If the call is on hold, the SBE will set a timer with a duration matching the configured value using this command.
		• If the call is not taken off hold before the SBE timer expires, then the call will be torn down.
		The default value for this command is off. Unless a specific duration is set, on hold calls never time out.
Step 5	end	Exits the configuration session and enters Privileged EXEC mode.
	Example:	
	Router(config-sbc-sbe)# end	
Step 6	show sbc sbc-name sbe hold-media-timeout	Shows the currently configured duration of the media timeout timer for on-hold calls.
	Example: Router# show sbc mysbc sbe hold-media-timeout	
Step 7	show sbc sbc-name sbe calls	Lists all the calls on the SBE.
	Example: Router# show sbc mysbc sbe calls	

SDP Call Hold Interworking

Cisco IOS XE Release 2.4 introduces support for SDP call hold interworking. With SDP call hold interworking, there are two ways of setting up call hold using SIP. Either the caller or callee can renegotiate the call characteristics using SDP so that either:

- The connection line is set to the null address, c=IN IP4 0.0.0.0.
- Or to the direction attribute for their endpoint so that it does not receive media from the endpoint.
 - If this was previously set to a=sendrecv, the endpoint putting the call on hold sets it to a=sendonly.

 If this was previously set to a=recvonly, the endpoint putting the call on hold sets it to a=inactive.

Some SIP endpoints support setting the connection line to the null address, some support setting the direction, and some support both approaches. Additionally, some endpoints only respect setting the direction attribute to sendonly or inactive.

With SDP call hold interworking, Cisco Unified Border Element (SP Edition) supports interoperating with SIP endpoints that support a subset of the above approaches. When Cisco Unified Border Element (SP Edition) detects that a call is being put on hold in the SDP, it removes any preexisting c=IN IP4 0.0.0.0 or a=direction lines and replaces them with appropriate settings for the endpoint.

If the endpoint putting the call on hold was sendrecv or sendonly, then the default behavior is to send

- c=IN IP4 0.0.0.0
- a=sendonly

If the endpoint putting the call on hold was recvonly or inactive, then the default behavior is to send

- C=IN IP4 0.0.0.0
- a=inactive



Music on hold is supported using SDP call hold interworking.

Configuring SDP Call Hold Interworking

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. first-cac-scope scope-name
- 6. first-cac-table table-name
- 7. cac-table table-name
- 8. table-type {policy-set | limit {list of limit tables}}
- 9. entry entry-id
- **10.** cac-scope {list of scope options}
- **11.** match-value key
- 12. caller-hold-setting {hold-c0 | hold-c0-inactive | hold-c0-sendonly | hold-sendonly | standard }
- **13.** action [cac-complete | next-table goto-table-name]
- 14. exit
- 15. exit
- 16. complete
- 17. active-cac-policy-set policy-set-id

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the submode of CAC policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 5	first-cac-scope <i>scope-name</i>	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope	The <i>scope-name</i> argument configures the scope at which limits should be initially defined. Possible values are:
	global	• adj-group
		• call
		• category
		• dst-account
		• dst-adj-group
		• dst-adjacency
		• dst-number
		• global
		• src-account
		• src-adj-group
		• src-adjacency
		• src-number
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies.

	Command or Action	Purpose
Step 6	first-cac-table table-name	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table RootCacTable	
Step 7	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set.
	Example:	
	Router(config-sbc-sbe-cacpolicy)# cac-table	
	RootCacTable	

	Command or Action	Purpose
Step 8	<pre>table-type {policy-set limit {list of limit tables}}</pre>	Configures the table type of a CAC table within the context of an SBE policy set.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit event-type	list of limit tables can be one of the following values:
		• account —Compare the name of the account.
		• adj-group —Compare the name of the adjacency group.
		• adjacency —Compare the name of the adjacency.
		• all —No comparison type. All events match this type.
		• call-priority —Compare with call priority.
		• category —Compare the number analysis assigned category.
		• dst-account —Compare the name of the destination account.
		• dst-adj-group —Compare the name of the destination adjacency group.
		• dst-adjacency —Compare the name of the destination adjacency.
		• dst-prefix —Compare the beginning of the dialed digit string.
		• event-type —Compare with CAC policy event types.
		• src-account —Compare the name of the source account.
		• src-adj-group —Compare the name of the source adjacency group.
		• src-adjacency —Compare the name of the source adjacency.
		• src-prefix —Compare the beginning of the calling number string.
		Features can be enabled or disabled per adjacency group through CAC configuration the same way this is done per individual adjacencies. The adj-group table type matches on either source or destination adjacency group.
Step 9	entry entry-id	Creates or modifies an entry in a table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	

	Command or Action	Purpose
Step 10	<pre>cac-scope {list of scope options}</pre>	Configures the scope within each of the entries at which limits are applied in a policy set table.
	<pre>Example: Cuter(config-sbc-sbe-cacpolicy-cactable-entry)# cac-scope call</pre>	• <i>list of scope options</i> —Specifies one of the following strings used to match events:
		 account—Events that are from the same account.
		 adjacency—Events that are from the same adjacency.
		 adj-group—Events that are from members of the same adjacency group.
		- call —Scope limits are per single call.
		- category —Events that have same category.
		 dst-account—Events that are sent to the same account.
		 dst-adj-group—Events that are sent to the same adjacency group.
		 dst-adjacency—Events that are sent to the same adjacency.
		 dst-number—Events that have same destination.
		- global—Scope limits are global
		 src-account—Events that are from the same account.
		 src-adj-group—Events that are from the same adjacency group.
		 src-adjacency—Events that are from the same adjacency.
		 src-number—Events that have the same source number.
		 sub-category—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category.
		 sub-category-pfx—The limits specified in this scope apply to all events sent to or received from members of the same subscriber category prefix.
		 subscriber—The limits specified in this scope apply to all events sent to or received from individual subscribers (a device that is registered with a Registrar server).

	Command or Action	Purpose
Step 11	<pre>match-value key Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value call-update</pre>	Specifies the keyword used to match events. The format of the key is determined by the table-type. If you configure either an event-type or call-priority Limit table, then you only see the keyword options that apply for that type of Limit table.
		For Limit event-type tables (table-type limit event-type), the match value keyword options are the following:
		• call-update —Compare the beginning of the calling number string.
		• endpoint-reg —Compare the name of the destination adjacency.
		• new-call —Compare the beginning of the dialed digit string.
		For Limit call-priority tables (table-type limit call-priority), the match value keyword options are the following:
		• critical —Match calls with resource priority 'critical.'
		• flash —Match calls with resource priority 'flash'.
		• flash-override —Match calls with resource priority 'flash-override.'
		• immediate —Match calls with resource priority 'immediate.'
		• priority —Match calls with resource priority 'priority.'
		• routine —Match calls with resource priority 'routine.'
		For all other Limit tables, enter a name or digit string.
		WORD—Name or digit string to match. (Max Size 255).
Step 12	<pre>caller-hold-setting {hold-c0 hold-c0-inactive hold-c0-sendonly hold-sendonly standard}</pre>	Configures the caller hold settings that are supported.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-hold-setting hold-sendonly</pre>	
Step 13	action [cac-complete next-table goto-table-name]	Specifies the action to take if this routing entry is chosen.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete</pre>	

	Command or Action	Purpose
Step 14	exit	Exits the cactable entry configuration mode and enters the cactable mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit</pre>	
Step 15	exit	Exits the cactable configuration mode and enters the cacpolicy mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit	
Step 16	complete	Completes the CAC-policy or call-policy set after committing the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 17	active-cac-policy-set policy-set-id	Sets the active CAC-policy-set within an SBE entity.
	Example:	
Step 17	Example: Router(config-sbc-sbe-cacpolicy)# complete active-cac-policy-set policy-set-id Example: Router (config-sbc-sbe)# active-cac-policy-set 1	Sets the active CAC-policy-set within an SBE entit

Configuration Examples

This section contains configuration examples.

Example of Configuring SIP Call Hold

The following command configures the SBE to wait for two hours after receiving the last media packet on an on-hold call before cleaning up the call resources.

```
Router# configure
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# hold-media-timeout 7200
```

Example of Configuring SDP Call Hold Interworking

In the example below, Fairchild Foods have replaced all the phones in their offices. The new phones support setting a=sendonly and c=IN IP4 0.0.0.0 to place a call on hold; they do not support setting a=inactive. You now want to reconfigure Cisco Unified Border Element (SP Edition) to work with these phones without changing the behavior for other customers. This change creates new policies at the account scope for all events, so that calls in which Fairchild Foods phones are involved are put on hold appropriately.

The following configuration changes will make sure Fairchild phone doesn't receive a=inactive in SDP when Fairchild is the source account and the callee puts the call on hold.

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
```

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```
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router(config-sbc-sbe-cacpolicy)# first-cac-table callhold-src-settings
Router(config-sbc-sbe-cacpolicy) # cac-table callhold-src-settings
Router (config-sbc-sbe-cacpolicy-cactable) # table-type limit src-account
Router(config-sbc-sbe-cacpolicy-cactable) # entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value fairchild
Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-hold-setting hold-c0-sendonly
Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete
Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy) # complete
The following configuration changes will make sure Fairchild phone doesn't receive
a=inactive in SDP when Fairchild is the destination account and the caller puts the call
on hold.
Router# configure terminal
Router(config) # sbc mysbc
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# first-cac-scope global
Router(config-sbc-sbe-cacpolicy)# first-cac-table callhold-dst-settings
Router(config-sbc-sbe-cacpolicy) # cac-table callhold-dst-settings
Router(config-sbc-sbe-cacpolicy-cactable) # table-type limit dst-account
```

Router(config-sbc-sbe-cacpolicy-cactable)# table-type Router(config-sbc-sbe-cacpolicy-cactable)# entry 1

Router (config-sbc-sbe-cacpolicy-cactable-entry) # match-value fairchild

Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-hold-setting hold-c0-sendonly

Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete

Router(config-sbc-sbe-cacpolicy-cactable-entry)# exit

Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy)# complete

```
Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model
```




SIP Call Transfer

Cisco Unified Border Element (SP Edition) supports Session Initiation Protocol (SIP) call transfer, a standard Internet telephony service. Call transfer allows a wide variety of decentralized multiparty call operations. These decentralized call operations form the basis for third-party call control, and are important features for voice over IP (VoIP) and SIP. Call transfer is also critical for conference calling, where calls can transition smoothly between multiple point-to-point links and IP level multicasting. The Cisco Unified Border Element (SP Edition) SIP call transfer feature includes basic in-dialog transfer and advanced call transfer for the following network topologies:

- Central SBC
- Transfer intra network
- Transfer out of network
- Transfer to colleague



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Call Transfer

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for SIP Call Transfer Support, page 47-2
- Information About SIP Call Transfer, page 47-2

Restrictions for SIP Call Transfer Support

The following is a list of restrictions for SIP call transfer support:

- The Configuration feature is expected to be "always on." Therefore, no configuration is required and it is not possible to disable it.
- REFER subscription state is not maintained over failover. Therefore, after a failover, any subsequent NOTIFYs telling the one referring about the progress of the referral are lost. They are bounced back with a 481 SIP error response. This will not prevent calls from being transferred, but may result in a few error logs if diagnostics are enabled.

Information About SIP Call Transfer

REFER Requests

The REFER method has three main roles:

- Originator—User agent that initiates the transfer or REFER request.
- Recipient—User agent that receives the REFER request and is transferred to the final-recipient.
- Final-Recipient—User agent introduced into a call with the recipient.

The REFER method always begins within the context of an existing call and starts with the originator. The originator sends a REFER request to the recipient (user agent receiving the REFER request) to initiate a triggered INVITE request. The triggered INVITE request uses the SIP URL contained in the Refer-To header as the destination of the INVITE request.

The recipient then contacts the resource in the Refer-To header (final-recipient), and returns a SIP 202 (Accepted) response to the originator. The recipient also must notify the originator of the outcome of the REFER transaction—whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the Notify Method, SIP's event notification mechanism.

A Notify message with a message body of SIP 200 OK indicates a successful transfer, while a body of SIP 503 Service Unavailable indicates an unsuccessful transfer. If the call was successful, a call between the recipient and the final-recipient results.

Cisco Unified Border Element (SP Edition) accepts and passes through in-dialog REFER requests. Standard SIP headers are manipulated as normal. The call-transfer specific headers are treated in the following way:

- The Refer-To header is passed through unchanged.
- The Referred-By header:
 - Any received Referred-By header is passed through ignored.

- On the outbound REFER, the following header is written:

Referred-By: <sip:endpoint_dn@sbc_adj_sip_domain_name>

except that,

- If the side of the call on which Cisco Unified Border Element (SP Edition) received the REFER has privacy enabled (configured in CAC), then no Referred-By header is written on the outbound REFER.
- The Replaces header is treated in the same way as for the INVITE requests.

Out-of-dialog REFER requests are rejected. The Target-Dialog header is not explicitly supported, and therefore is stripped or passed through, subject to header and method white/blacklisting configuration.

NOTIFY Messages

When the outcome of the REFER transaction is known, the recipient of the REFER request must notify the originator of the outcome of the REFER transaction—whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the NOTIFY method, SIP's event notification mechanism. The notification contains a message body with a SIP response status line and the response class in the status line indicates the success or failure of the REFER transaction.

Cisco Unified Border Element (SP Edition) accepts and passes through in-dialog NOTIFY requests. Standard SIP headers are manipulated as normal.

- If the NOTIFY contains a body of type message/sipfrag, and if the start of this body can be correctly parsed as a SIP response status line, then the outbound NOTIFY is given a message/sipfrag body containing a SIP response status line with the same response code (and nothing else).
- If there is no body of type message/sipfrag on the NOTIFY, or the first line of the NOTIFY body cannot be correctly parsed as a status line, then the onbound NOTIFY is sent without a body. This includes the case where there is a message/sipfrag body included as part of a mime/multipart body.

Replaces Headers

The processing of Replaces headers is the key logic involved in supporting call transfer across Cisco Unified Border Element (SP Edition). Cisco Unified Border Element (SP Edition) does a lookup on the call IDs and tags in the received Replaces header. If it finds the corresponding call branch (for example, C1), then it looks up the partner call branch (for example, C2). C1 and C2 together make up another call through Cisco Unified Border Element (SP Edition). The Replaces header sent out on the request which is forwarded on might refer to call branch C1 or C2, depending on the request type and other considerations. Any "early-only" flag on the Replaces header is passed through.

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SIP Authentication

Cisco Unified Border Element (SP Edition) supports Session Initiation Protocol (SIP) authentication.



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Authentication

Release	Modification
Cisco IOS XE Release 2.4	Support for SIP authentication was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 2.5	Support for interoperability for SIP authentication of INVITE requests was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.
Cisco IOS XE Release 2.6	Support for interoperability for SIP authentication of outbound out-of-dialogue requests (using the same generation scheme as used by INVITE requests for the Call-ID, From and To dialog tags, and CSeq sequence numbers) was introduced on the Cisco ASR 1000 Series Aggregation Services Routers.

Contents

This module contains the following sections:

- SIP Outbound Authentication, page 48-2
- SIP Inbound Authentication, page 48-6
- Interoperability for SIP Authentication, page 48-11

SIP Outbound Authentication

When network entities communicate using SIP, one entity often needs to challenge another one to determine if it is authorized to transmit SIP signaling into the challenger's network. The SIP authentication model is based on the HTTP digest authentication, as described in the RFC 2617.

The use of basic authentication, where passwords are transmitted unencrypted, is not permitted in SIP.

This section contains the following subsections:

- Prerequisites for Implementing SIP Outbound Authentication, page 48-2
- Restrictions for Implementing SIP Outbound Authentication, page 48-2
- Information About SIP Outbound Authentication, page 48-3
- How to Configure SIP Outbound Authentication, page 48-4
- Examples of Show Commands, page 48-5

Prerequisites for Implementing SIP Outbound Authentication

The following prerequisites are required to implement SIP outbound authentication:

- Configure a SIP adjacency before you specify one or more authentication-realms.
- Configure the Cisco Unified Border Element (SP Edition) with a set of domains (realms) with which it can authenticate itself. Set the username and password to provide when challenged by each of these domains. This configuration is implemented per adjacency.



Multiple realms can be configured per adjacency and there is no limit on the number of these realms aside from memory availability. Different realms may be configured with the same username and password. Also, each realm may be configured with different username and password on different adjacencies. However, any realm can be configured a maximum of one time per adjacency.

Restrictions for Implementing SIP Outbound Authentication

The following restrictions apply to SIP outbound authentication:

• Cisco Unified Border Element (SP Edition) rejects any attempt to configure an authentication-realm with the same domain name as an existing authentication-realm. This restriction is valid per adjacency. Multiple adjacencies may have authentication-realms configured with the same domain.



- The current command line interface (CLI) prohibits the user from configuring two authentication-realms with the same domain for the same adjacency. If this is attempted, the CLI interprets the second authentication-realm configuration as an attempt to reconfigure the first authentication-realm, and updates the user's credentials accordingly.
- Each authentication-realm can only be configured with a single username and password per adjacency.

Information About SIP Outbound Authentication

This section contains the following subsections:

- Configuring Outbound Authentication in Cisco Unified Border Element (SP Edition), page 48-3
- Authenticating the Cisco Unified Border Element (SP Edition) to Remote Devices, page 48-3

Configuring Outbound Authentication in Cisco Unified Border Element (SP Edition)

When a SIP adjacency is configured, the user may specify one or more authentication-realms. Each authentication-realm represents a remote domain, from which Cisco Unified Border Element (SP Edition) receives authentication challenges on the adjacency. When an authentication-realm is configured, the user must specify the correct user name and password that Cisco Unified Border Element (SP Edition) uses to authenticate itself in that realm. Cisco Unified Border Element (SP Edition) stores all valid authentication-realms for each adjacency.

Authenticating the Cisco Unified Border Element (SP Edition) to Remote Devices

Upon receipt of a SIP 401 or 407 response that can be correlated to a request it sent, Cisco Unified Border Element (SP Edition) examines the attached authentication challenge. Cisco Unified Border Element (SP Edition) responds to any authentication challenge received on a given adjacency that matches one of the configured authentication-realms for that adjacency. Any authentication challenge that does not match the configured authentication-realm is passed through unchanged to the SBC's signaling peer for the adjacency, on which the original request was received.

To generate a response to an authentication challenge, Cisco Unified Border Element (SP Edition) does the following:

- 1. First, it looks up the realm parameter of the challenge in its list of configured authentication-realms for the outbound adjacency.
- 2. Second, it finds the password for that authentication-realm and generates an authentication response by combining the password with the nonce parameter from the challenge, and hashing the result.
- **3.** If the challenger has requested **auth-int** quality of protection, Cisco Unified Border Element (SP Edition) also generates a hash of the entire message body and includes it in the response.
- **4.** Cisco Unified Border Element (SP Edition) builds an Authorization (or Proxy-Authorization) header by including the following parameter values (following RFC 2617):
- Nonce from challenge.
- Realm from challenge.
- Digest-URI is set to the SIP URI of the challenged request.
- Message-QOP is set to **auth.**
- Response calculated as described previously.
- Username as specified for the relevant authentication-realm.
- If the challenge contained an opaque parameter, it is returned unchanged on the response.
- If the challenge contained the **qop-directive** parameter, then the **nonce-count** parameter is set to the number of the sent requests, using the response calculated from this nonce.

- Note that the domain parameter is not expected to be included on any challenges that Cisco Unified Border Element (SP Edition) must respond to. This parameter is not used on Proxy-Authenticate challenges, the type of challenge that Cisco Unified Border Element (SP Edition) most often receives. If the domain parameter is included, Cisco Unified Border Element (SP Edition) ignores it.
- 5. Finally, Cisco Unified Border Element (SP Edition) stores its calculated response and the received nonce with the other data for the authentication-realm. This allows Cisco Unified Border Element (SP Edition) to respond rapidly to the subsequent challenges from this realm with the same nonce. If Cisco Unified Border Element (SP Edition) lacks the resources to store its response, it carries on anyway. The next time an authorization challenge is received from this realm, Cisco Unified Border Element (SP Edition) has to recalculate its response. When Cisco Unified Border Element (SP Edition) re-uses a saved response, it updates the nonce count stored along with the nonce-response pair. This allows Cisco Unified Border Element (SP Edition) to correctly fill in the nonce-count field in Authorization responses.

How to Configure SIP Outbound Authentication

This section contains the steps for configuring SIP outbound authentication, allowing the user to add/remove one or more authentication-realms to/from an adjacency.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. authentication-realm inbound domain | outbound domain username password
- 6. end
- 7. show sbc sbc-name sbe adjacency adjacency-name authentication-realms
- 8. show sbc service-name sbe all-authentication-realms

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	

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	Command or Action	Purpose
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip test	• Use the <i>adjacency-name</i> argument to define the name of the service.
Step 5	authentication-realm {inbound domain/outbound domain username password}	Configures a set of outbound authentication credentials for the specified domain on the specified adjacency. This command can be issued either before or after the adjacency has been attached.
	Router(config-sbc-sbe-adj-sip)# authentication-realm outbound example.com usersbc passwrdsbc	The no version of this command deconfigures the authentication-realm on the specified adjacency.
	-	• inbound—Specifies inbound authentication realm.
		• outbound—Specifies outbound authentication realm.
		• domain—Name of the domain for which the authentication credentials are valid.
		• username—User name that identifies the SBC in the specified domain.
		• password—Password to authenticate the username in the specified domain.
Step 6	end	Exits the adj-sip mode and returns to privileged EXEC mode
	Example: Router(config-sbc-sbe-adj-sip)# end	
Step 7	<pre>show sbc sbc-name sbe adjacency adjacency-name authentication-realms</pre>	Shows all currently configured authentication-realms for the specified SIP adjacency.
	Example: Router# show sbc mySbc sbe adjacency SipToIsp42 authentication-realms	
Step 8	<pre>show sbc service-name sbe all-authentication-realms</pre>	Shows all currently configured authentication-realms for all SIP adjacencies.
	Example: Router# show sbc mySbc sbe all-authentication-realms	

Examples of Show Commands

Router# show sbc mySbc sbe adjacency SipToIsp42 authentication-realms

Configured authentication realms Domain Username Password Example.com usersbc passwordsbc

Router# show sbc mySbc sbe all-authentication-realms

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Configured a	uthenticat	ion realms			
Adjacency: S Domain Remote.com	ipToIsp42 Username usersbc	Password sbcpassword	Example.com	usersbc	passwordsbc
Adjacency: S Domain Example.com Other.com	ipToIsp50 Username user2sbc sbcuser	Password password2sbc sbcsbcsbc			

SIP Inbound Authentication

Cisco Unified Border Element (SP Edition) supports two modes of Session Initiation Protocol (SIP) inbound authentication to challenge inbound SIP requests: local and remote. You must select the mode of authentication to configure Cisco Unified Border Element (SP Edition) according to the level of support present in the Remote Authentication Dial-In User Service (RADIUS) servers. If the RADIUS servers are compliant with only draft-sterman-aaa-sip-00 to 01, then select the local mode. If the RADIUS servers are compliant with only RFC 4590, then use the remote authentication mode.

Note

This feature is optional and you can configure the Cisco Unified Border Element (SP Edition) not to challenge the inbound requests.

This section contains the following subsections:

- Prerequisites for Implementing SIP Inbound Authentication, page 48-6
- Restrictions for Implementing SIP Outbound Authentication, page 48-2
- Information About SIP Inbound Authentication, page 48-7
- How to Configure SIP Inbound Authentication, page 48-8
- Examples of Show Commands, page 48-5

Prerequisites for Implementing SIP Inbound Authentication

The following prerequisites are required to implement SIP inbound authentication:

- Configure a SIP adjacency with the intended mode of authentication before you configure Cisco Unified Border Element (SP Edition) to authenticate inbound calls.
- Configure the RADIUS server to specify which mode of inbound authentication is selected.

Restrictions for Implementing SIP Inbound Authentication

The following restrictions and limitations apply to implement SIP inbound authentication:

- Cisco Unified Border Element (SP Edition) supports only one inbound authentication realm per adjacency.
- Cisco Unified Border Element (SP Edition) does not check the validity of nonces generated by a RADIUS server; the RADIUS server must be configured to perform this check.

- Cisco Unified Border Element (SP Edition) does not designate a particular RADIUS server group on an adjacency for inbound authentication.
- Since trust-transference of calls does not occur between inbound authentication, outbound authentication, and Transport Layer Security (TLS) connections, a successful inbound authentication does not ensure that Cisco Unified Border Element (SP Edition) marks the call as secure or implement outbound authentication. Users can, however, configure inbound authentication, outbound authentication, and TLS independently on the same adjacency.

Information About SIP Inbound Authentication

This section contains the following subsections:

- Local Inbound Authentication, page 48-7
- Remote Inbound Authentication, page 48-7
- Interaction with Outbound Authentication, page 48-7
- Failure Modes for Inbound Authentication, page 48-7

Local Inbound Authentication

When configured to perform local inbound authentication, Cisco Unified Border Element (SP Edition) is responsible for challenging an unauthorized request from the remote peer first. Therefore, to be able to challenge the request from the remote peer, the adjacency must already be configured with an authentication realm. After the remote peer has validated the request, it is forwarded to the RADIUS server, which then decides whether to permit the call to pass through or not.

Remote Inbound Authentication

When configured to perform remote inbound authentication, Cisco Unified Border Element (SP Edition) relies on the RADIUS server to challenge an authorized request from the remote peer. Cisco Unified Border Element (SP Edition) forwards the challenge request generated by the RADIUS server to the remote peer, and also forwards the remote peer's authentication request to the RADIUS server.

Interaction with Outbound Authentication

If an adjacency is configured for inbound authentication, then after it successfully authenticates an inbound request, the authorization headers matching the realm for that adjacency are stripped out and not propagated to the outbound signal. Authorization headers for other realms, however, are passed through to the outbound request.

Failure Modes for Inbound Authentication

When the inbound authentication is configured, the following failure modes may occur (in addition to the standard SIP signal failure modes):

Unacceptable Parameters

If the endpoint or RADIUS server specifies a quality of protection parameter other than **auth** or **auth-int**, then the inbound request is rejected and a 403 response is generated. Similarly, Cisco Unified Border Element (SP Edition) generates a 403 response when algorithms other than MD5 and MD5-sess are used.

Access-Request Rejection

If the RADIUS server rejects the Access-Request signal with an Access-Reject response, Cisco Unified Border Element (SP Edition) sends a 403 response to the endpoint.

Insufficient Memory

If Cisco Unified Border Element (SP Edition) does not have sufficient memory to process an inbound authentication request, it rejects the request and sends a 503 response.

No Match on Authentication Realm

If the peer does not return any authentication headers that specify the authentication realm contained in the adjacency's configuration, then Cisco Unified Border Element (SP Edition) rechallenges the request with 401 response.

No Match on Nonce

If the peer's nonce does not match the one generated by Cisco Unified Border Element (SP Edition), then Cisco Unified Border Element (SP Edition) rejects the authentication request and sends a 403 response.

Nonce Timed Out

If the peer's nonce has timed out, then Cisco Unified Border Element (SP Edition) challenges the nonce by sending a 401 response and a new nonce.

No Acceptable RADIUS Servers

If there is no RADIUS server to support a mode configured on the adjacency, then Cisco Unified Border Element (SP Edition) rejects the authentication request with a 501 response and creates a log to alert the user of the inconsistent configuration.

How to Configure SIP Inbound Authentication

This section contains the steps for configuring SIP local inbound authentication a RADIUS server.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. radius [accounting client-name | authentication]
- 5. server server-name

- 6. address
- 7. mode local
- 8. key password
- 9. exit
- 10. activate
- 11. exit
- **12. adjacency sip** *adjacency-name*
- **13.** authentication-realm inbound realm
- 14. authentication mode local
- **15.** authentication nonce timeout time
- 16. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	radius [accounting client-name authentication]	Enters the mode for configuring a RADIUS client for authentication purposes.
	Example: Router(config-sbc-sbe)# radius authentication	
Step 5	server server-name	Enters the mode for configuring the authentication server.
	Example: Router(config-sbc-sbe-auth)# server authserv	
Step 6	address ipv4 ipv4-address	Specifies the IPv4 address of the authentication server.
	Example: Router(config-sbc-sbe-auth-ser)# address ipv4 200.200.200.122	

	Command or Action	Purpose
Step 7	<pre>mode {local remote} or server server-name mode {local remote}</pre>	Configures the RADIUS server for local inbound authentication. By default, the mode is remote.
Step 8	Example: Router(config-sbc-sbe-auth-ser)# mode local key password	Sets the authentication server key.
	<pre>Example: Router(config-sbc-sbe-auth-ser)# key authpass1</pre>	
Step 9	exit	Exits the mode for configuring the authentication server.
Sten 10	Example: Router(config-sbc-sbe-auth-ser)# exit	Activates the PADIUS client
oteh in		Activates the KADIOS citent.
	Example: Router(config-sbc-sbe-auth)# activate	
Step 11	exit	Exits the mode for configuring the RADIUS client and enters the SBE mode.
	<pre>Example: Router(config-sbc-sbe-auth)# exit</pre>	
Step 12	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip test	• Use the <i>adjacency-name</i> argument to define the name of the service.
Step 13	authentication-realm inbound realm	Configures a set of authentication credentials for a specified domain on the specified SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# authentication-realm inbound cisco.com</pre>	Note This is a mandatory parameter for local mode.
Step 14	authentication mode local	Configures the SIP adjacency for local inbound authentication. To configure the SIP adjacency, for
	Example: Router(config-sbc-sbe-adj-sip)# authentication mode local	remote inbound authentication, set the value to remote .
Step 15	authentication nonce timeout time	Configures the value of the authentication nonce timeout in seconds. The range of acceptable values is 0 to 65535 seconds. The default value is 300 seconds
	<pre>Example: Router(config-sbc-sbe-adj-sip)# authentication nonce timeout 10000</pre>	seconds. The default value is 500 seconds.
Step 16	exit	Exits the adj-sip mode and returns to the SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

Examples of Show Commands

Router# show sbc mySbc sbe adjacencies SipToIsp42 detail

SBC server mySbc Adjacency SipToIsp42 Status: Attached Signaling address: 10.2.0.122:5060 Signaling-peer: 200.200.200.179:8888 Force next hop: No Account: core Group: None In Header Profile: Default Out Header Profile: Default In method profile: Default Out method profile: Default In UA option profile: Default Out UA option profile: Default In proxy option profile: Default Priority set name: Default Local-id: None Rewrite REGISTER: Off Target address: None NAT Status: Auto-Detect Reg-min-expiry: 3000 seconds Fast-register: Enabled Fast-register-int: 30 seconds Authenticated mode: Local Authenticated realm: Cisco.com Authenticated nonce life time: 300 seconds IMS visited NetID: NOne Inherit profile: Default Force next hop: No Home network ID: None UnEncrypt key data: None SIPIpassthrough: No Rewrite from domain: Yes Rewrite to header: Yes Media passthrough: No Preferred transport: UDP Hunting Triggers: Global Triggers Redirect mode: Passthrough

Interoperability for SIP Authentication

Cisco Unified Border Element (SP Edition) supports interoperability between SIP devices and third-party soft switch equipments for SIP authentication of all SIP requests. The supported interoperability applies to dialog-creating INVITE requests and out-of-dialog REGISTER and SUBSCRIBE requests only.

Support for interoperability for SIP authentication of INVITE requests was introduced in Cisco IOS XE Release 2.5. Cisco Unified Border Element (SP Edition) uses a generation scheme that generates the Call-ID, From and To dialog tags, and CSeq sequence numbers on the outbound call leg using the inbound request message data which provides both uniqueness and retains the same values for subsequent requests resulting from any challenges.

Support for interoperability for SIP authentication of out-of-dialog requests was introduced in Cisco IOS XE Release 2.6. The same generation scheme used by INVITE requests (based on the inbound request message data) was implemented for out-of-dialog requests.

L

Cisco Unified Border Element (SP Edition) interoperates with third-party soft switch devices for processing SIP authentication of INVITE and out-of-dialog requests in the following way:

- The Call-ID of an authorized SIP request matches that of the initial request.
- The To and From headers of an authorized SIP request match those of the initial request, including the dialog tag (if any) in the From header.
- The CSeq sequence number of an authorized SIP request is one higher than the initial request.

This section contains the following subsections:

- Information About SIP Outbound Authentication, page 48-3
- Information About Interoperability for SIP Authentication, page 48-13

Restrictions for Interoperability for SIP Authentication

The following restrictions apply to support for Interoperability for SIP Authentication on the Cisco Unified Border Element (SP Edition):

- Cisco Unified Border Element (SP Edition) meets the following interoperability requirements only when the received signaling from the upstream call leg also meets the same:
 - The Call-ID of an authorized SIP request matches that of the initial request.
 - The To and From headers of an authorized SIP request match those of the initial request, including the dialog tag (if any) in the From header.
 - The CSeq sequence number of an authorized SIP request is one higher than the initial request.
- Cisco Unified Border Element (SP Edition) depends on the randomness of the values in the received signaling. For example, if a calling endpoint generates insufficiently random values, then the values sent by the SBC on the outbound call leg will also be insufficiently random. However, the SBC uses any and all randomness from the Call-IDs and From tags generated by the caller, and in addition takes steps to avoid Call-ID and tag collisions between the upstream and downstream signaling.
- If the input from certain configuration fields to Call-ID and To/From header generation is changed between forwarding an initial SIP request and the subsequent authorized SIP request to that adjacency, then the headers of the two requests do not match. This can lead to call setup failure, depending on the downstream signaling entities. In particular, note the following:
 - The local-id or signaling-address configuration under adjacency affects Call-ID and From-tag generation.
 - Header rewriting configuration can apply to From and To headers. To meet the interoperability requirements, this rewriting must produce identical results on successive requests.
 - Cisco Unified Border Element (SP Edition) does not issue warnings to the user before accepting such configuration changes.
- Interoperability for SIP authentication affects dialog-creating INVITE requests and out-of-dialogue requests. Requests with other methods are not affected.
- To meet the interoperability requirements, the initial and authorized requests should not be routed out of different adjacencies.

Information About Interoperability for SIP Authentication

This section provides information about interoperability for SIP authentication.

SIP Requests

SIP requests refer to the messages within the scope of a single challenge or response sequence. There can be several sequences before a request is accepted, but the first sequence of any pair of consecutive requests is referred to as the initial request and the second one is referred to as the authorized request.

SIP requests are both dialog-creating INVITE requests and out-of-dialog requests.

The following SIP request terms are used in this chapter:

Initial request—A SIP request with insufficient authentication credentials, which is challenged with a "401–Unauthorized" or a "407–Proxy Authentication Required" response.

Authorized request—The corresponding subsequent request, sent on receipt of the 401/407 challenge response. This request contains an extra Authorization or Proxy-Authorization header.

Call-ID Generation

Cisco Unified Border Element (SP Edition) generates Call-ID values for outbound dialog-creating INVITE requests and out-of-dialogue requests (such as REGISTERS and SUBSCRIBES), based on the Call-ID of the inbound request and on configuration.

The generated Call-ID values are composed of a 32-character hexadecimal MD5 hash of the received Call-ID, an '@' character, and a local-id string representing the SBC itself.

Example:



Call-ID: 4264330abc5106c8ab70ed3fd222b7b2@sbc.home.net

The local-id string is the configuration from the outbound adjacency. If this configuration is absent, the canonical text representation of the signaling-address from the outbound adjacency is used.

From Tag Generation

Cisco Unified Border Element (SP Edition) generates From header dialog tag values for outbound dialog-creating INVITE requests and out-of-dialogue requests (such as REGISTERS and SUBSCRIBES), based on the From tag of the inbound request and on configuration.

The generated From tag values are composed of a local-id string representing the SBC itself, two eight-character hexadecimal MD5 hashes of the received From tag, and a numerical index identifying the internal component responsible for the dialog.

Example:

From: "Fred" <sip:2222222@sbc.home.net>;tag=sbc.home.net+1+a27d9765+b7f0f7e1

The local-id string is generated from configuration in the same way as for Call-IDs.

CSeq Sequence Number Generation

Cisco Unified Border Element (SP Edition) chooses the sequence number for use in the CSeq header of an outbound dialog-creating INVITE request and out-of-dialogue requests (such as REGISTERS and SUBSCRIBES) to be the same as the sequence number on the received inbound request.

Cisco Unified Border Element (SP Edition) continues to choose sequence numbers for subsequent outbound requests on the same dialog by storing the dialog's current sequence number, and incrementing it each time a new transaction is created.

Example:

CSeq: 949005087 INVITE

Pass-Through Authentication

SBC supports passing through authentication challenges and their responses. No configuration is required for this.

- 407 responses are passed through by SBC.
- On challenge responses, a quality of protection (qop) of "auth-int" is stripped out, as SBC necessarily modifies the message, which negates the integrity of the authentication. If this is the only qop offered, the challenge is converted into a 403 Forbidden return code.



Late-to-Early Media Interworking

The late-to-early media interworking feature is supported for Session Initiation Protocol (SIP) calls. In order to interwork between a late media caller and an early media callee, Cisco Unified Border Element (SP Edition) sends an invite to the callee that includes a Session Description Protocol (SDP) offer of media. Two implementations of late-to-early media interworking are available:

- By default, SBC generates the SDP with a single media line that specifies codecs common to both the caller and the callee's codec whitelists.
- SBC can also be configured with a media description using the **sip sdp-media-profile** command to generate a customized offer.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for Late-to-Early Media Interworking

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR.
Cisco IOS XE Release 2.5	The customizable offer for late-to-early media interworking feature was introduced on the Cisco IOS XR.

Contents

This module contains the following sections:

- Restrictions for Late-to-Early Media Interworking Support, page 49-2
- Information about Late-to-Early Media Interworking, page 49-2
- Configuring Late-to-Early Media Interworking, page 49-4
- Configuration Examples for the Late-to-Early Media Interworking Feature, page 49-13
- Verification, page 49-17

Restrictions for Late-to-Early Media Interworking Support

The restrictions for late-to-early media interworking are:

- This feature applies only to SIP-to-SIP calls, it does not apply to SIP-to-H.323 interworking calls.
- This feature applies only to IPv4; you cannot use it with IPv6 addressing.
- If the caller refines the media chosen by the callee, this is sent back to the callee in a PRACK. However, if the callee attempts to refine the media again, the event is logged but it is not passed back to the caller.
- Because Cisco Unified Border Element (SP Edition) generates SDPs, any calls using this feature cannot use media bypass.
- Cisco Unified Border Element (SP Edition) only generates SDPs offering a single audio stream. If the caller and callee want to negotiate video, fax, or other media streams, they can renegotiate this after the call has been established.
- If the callee attempts to send early media either before or without sending a reliable 1XX INVITE, Cisco Unified Border Element (SP Edition) will drop that media. It will not reach the caller.
- The callee must not send unreliable 1XX INVITE responses because the caller would interpret them as an out-of-sequence SDP offer. For late-to-early interworking calls, Cisco Unified Border Element (SP Edition) sets 100rel as mandatory in order to forbid the callee from sending unreliable responses only if the caller side supports 100rel.
- Late-to-early media interworking must not be used with the Gq IMS interface. This interface does not provide Cisco Unified Border Element (SP Edition) with the local media address necessary to create an SDP offer (and will likely result in calls with incorrect media paths).

Information about Late-to-Early Media Interworking

This section includes the following topics:

- Late-to-Early Media Interworking Description, page 49-2
- Customizable Offer for Late-to-Early Media Interworking, page 49-3

Late-to-Early Media Interworking Description

Early Media is the ability of two user agents to communicate before a call is actually established. Early Media can flow when the caller makes a media proposal on the initial call setup request and the callee responds to the offer before the call is connected. Cisco Unified Border Element (SP Edition) provides interoperability between SIP devices that do not provide SDP on their INVITEs and SIP devices that require SDP on INVITEs they receive. This occurs when:

- An endpoint caller wants to negotiate media after the INVITE has been accepted (late media) and does not include an SDP offer on the initial INVITE
- The callee that expects an SDP offer on the initial INVITE, which it then answers with a 1XX response (early media).

The normal negotiation for media is for the caller to include an SDP offer on the initial INVITE and for the callee to accept with a 200 response. However, the following might occur:

• Late media is used by some endpoints, such as call agents that want to allow the callee to select the media used.

• Early media is used by some more recent endpoints that need to support media flow before the call is accepted, such as a pre-call announcement or in-band tones from a Call Hold server.

In order to interwork between a late media caller and an early media callee, Cisco Unified Border Element (SP Edition) sends an invite to the callee that includes an SDP offer of media. Cisco Unified Border Element (SP Edition) then sends appropriate messages between the caller and callee, depending on the responses from each.

Cisco Unified Border Element (SP Edition) supports this interworking on a per-adjacency basis. You can configure each adjacency to require late-to-early media interworking for calls made to that adjacency and/or for calls made from that adjacency.

Customizable Offer for Late-to-Early Media Interworking

By default, SBC generates the SDP with a single media description that specifies codecs common to both the caller and callee's codec whitelists.

The Customizable Offer for Late-to-Early Media Interworking feature provides customized SDPs with one or more media descriptions. You configure the media descriptions in named profiles (SDP media profiles) and associate the profiles to signals by including the profile name in a CAC policy.

To enable a customized offer for late-to-early media interworking:

- Enable late-to-early media interworking per adjacency, as described in the ?\$paranum>Configuring Late-to-Early Media Interworking Per Adjacency? section on page 49-4.
- Create a named SDP media profile containing one or more media description lines which will be inserted into the SDP when SBC is generating the INVITE. SBC will insert the media description lines into the SDP per the sequence number configured.
- Associate this sdp-media-profile with a cac-policy table entry.

When a call requires late-to-early interworking, if the CAC policy entry for that call contains a valid SDP media profile name, then SBC generates a customized SDP. In the absence of such an association, SBC generates the default SDP. In the customized case, SBC inserts the media description lines in the media profile in the SDP when it generates the INVITE. Each entry in the media profile includes a sequence number, which controls the ordering of the lines in the generated SDP.

Rules for Media Lines in SDP Media Profiles

A section of SDP is configured as an entry in the SDP Media profile. An entry can have one or many media description lines. The format of an SDP Media profile is:

```
entry number
   media-line index "media_description"
   media-line index "media_description"
   exit
```

For example:

```
entry 1
  media-line 1 "m=audio 0 RTP/AVP 0"
  media-line 2 "a=rtpmap:0 PCMU/8000"
  exit
```

If more than one media description is created in the same profile, all of the entries are used to generate the same output SDP, in ascending order by entry number.

The *media_description* argument must be enclosed in quotes (" "). The value inside the quotes must be syntactically valid SDP as defined in RFC 2327. The following rules apply:

- An SDP entry must contain exactly one m-line. The m-line must appear first in the entry. The m-line port must be zero. SBC replaces the zero with the appropriate port.
- An SDP entry must not contain a c-line.

The Cisco command line interface handles the contents of *media_description* as a string value. It does not check the syntax of the configured information. If the syntax is incorrect, outbound offers by the SBC are rejected.

Configuring Late-to-Early Media Interworking

This section describes the following configuration scenarios for Late-to-Early Media Interworking:

- Configuring Late-to-Early Media Interworking Per Adjacency, page 49-4
- Configuring Customized Offers for Late-to-Early Media Interworking, page 49-11

Configuring Late-to-Early Media Interworking Per Adjacency

This task shows how to configure late-to-early media interworking per adjacency.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

SUMMARY STEPS

- 1. configure terminal
- **2. sbc** *service-name*
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. nat force-off
- 6. preferred-transport udp
- 7. redirect-mode pass-through
- 8. authentication nonce timeout value
- 9. signaling-address ipv4
- 10. signaling-port
- 11. remote-address ipv4
- 12. signaling-peer
- 13. signaling-peer-port
- 14. dbe-location-id
- 15. account

- 16. reg-min-expiry
- 17. media-late-to-early-iw {incoming | outgoing}
- 18. attach
- 19. exit
- 20. exit
- 21. sip inherit profile
- 22. cac-policy-set
- 23. first-cac-table
- 24. first-cac-scope
- 25. averaging-period
- 26. cac-table
- 27. table-type limit list of limit tables
- 28. entry
- 29. match-value
- **30**. action cac-complete
- 31. max-bandwidth
- 32. max-updates
- 33. max-channels
- 34. early-media-type
- 35. early-media-timeout
- 36. codec-restrict-to-list
- 37. caller-codec-list
- **38**. callee-privacy
- 39. caller-privacy
- 40. exit
- 41. exit
- 42. complete
- 43. exit
- 44. active-cac-policy-set

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the submode for configuring the method profile.
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Configures an adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipGW	
Step 5	nat force-off	Configures a SIP adjacency to assume that all endpoints are behind a NAT device.
	<pre>Example: Router(config-sbe-adj-sip)# nat force-off</pre>	
Step 6	preferred-transport udp	Sets the preferred transport protocol for SIP signaling on an adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# preferred-transport udp	
Step 7	redirect-mode pass-through	Configures the behavior of SBC on receipt of a 3xx response to an invite from the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# redirect-mode recurse	
Step 8	authentication nonce timeout value	Configures the authentication nonce timeout for a SIP adjacency.
	Example: Router(config-sbe-adj-sip)# authentication nonce timeout 10	
Step 9	signaling-address ipv4	Defines the local IPv4 signaling address of a SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# signaling-address ipv4 10.10.10.10</pre>	

	Command or Action	Purpose
Step 10	<pre>signaling-port signaling-port</pre>	Defines the local port of signaling address of a SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-port 5000	
Step 11	remote-address ipv4	Configures a SIP adjacency to restrict the set of remote signaling peers that can be contacted over the adjacency to
	<pre>Example: Router((config-sbc-sbe-adj-sip)# remote-address ipv4 36.36.36.20 255.255.255.0</pre>	those with the given IP address prefix.
Step 12	signaling-peer	Configures a SIP adjacency to use the given remote signaling-peer.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-peer gk andrew	
Step 13	signaling-peer-port	Configures a SIP adjacency to use the given remote signaling-peer's port.
	Example: Router(config-sbc-sbe-adj-sip)# signaling-peer-port 123	
Step 14	dbe-location-id	Configures an adjacency to use a given media gateway DBE location when routing media.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# dbe-location-id 1</pre>	
Step 15	account	Defines a SIP adjacency account on an SBE.
	Example: Router(config-sbc-sbe-adj-sip)# account isp42	
Step 16	reg-min-expiry	Configures the minimum registration period in seconds on the SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# reg-min-expiry 300</pre>	
Step 17	<pre>media-late-to-early-iw {incoming outgoing}</pre>	Configures late-to-early media interworking (iw).
	Example: Router(config-sbe-adj-sip)# media-late-to-early-iw incoming	
Step 18	attach	Attaches an adjacency to an account on an SBE.
	Example: Router(config-sbc-sbe-adj-sip)# attach	

	Command or Action	Purpose
Step 19	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 20	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-adj)# exit	
Step 21	sip inherit profile	Configures a global inherit profile.
	Example: Router(config-sbc-sbe)# sip inherit profile preset-p-cscf-access	
Step 22	cac-policy-set	Enters the submode of CAC policy set configuration within an SBE entity.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	
Step 23	first-cac-table	Configures the name of the first policy table to process when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-table RootCacTable	
Step 24	first-cac-scope	Configures the scope at which to begin defining limits when performing the admission control stage of policy.
	Example: Router(config-sbc-sbe-cacpolicy)# first-cac-scope src-adjacency	
Step 25	averaging-period	Configures the size of the averaging period used by CAC for its rate calculations.
	Example: Router(config-sbc-sbe-cacpolicy)# averaging-period 5	
Step 26	cac-table	Creates or configures an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table MyCacTable	
Step 27	table-type limit list of limit tables	Configures a CAC Limit table type.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type limit call-priority	

	Command or Action	Purpose
Step 28	entry num	Creates or modifies an entry in a table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1</pre>	
Step 29	match-value value-keyword	Configures the match-value of an entry in an admission control table. Use the ? to see a list of valid keywords.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # match-value routine</pre>	
Step 30	action cac-complete	Specifies that when an event matches, this CAC policy is complete.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # action cac-complete</pre>	
Step 31	max-bandwidth	Configures the maximum bandwidth for an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-bandwidth 6000000</pre>	
Step 32	max-updates	Configures the maximum call updates for an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-updates 500</pre>	
Step 33	max-channels	Configures the maximum number of channels for an entry in an admission control table.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-channels 50</pre>	
Step 34	early-media-type {backward-half-duplex forward-half-duplex full-duplex}	Configures the direction of early media to allow for an entry in a call admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # early-media-type full-duplex	
Step 35	early-media-timeout	Configures the amount of time for which to allow early-media before a call is established.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # early-media-timeout 90</pre>	
Step 36	codec-restrict-to-list	Configures the CAC to restrict the codecs used in signaling a call to the set of codecs given in the named list.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # codec-restrict-to-list my_codecs</pre>	

	Command or Action	Purpose
Step 37	caller-codec-list	Lists the codecs which the caller leg of a call is allowed to use.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-codec-list test</pre>	
Step 38	callee-privacy	Configures the level of privacy processing to perform on messages sent from callee to caller.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # callee-privacy always</pre>	
Step 39	caller-privacy	Configures the level of privacy processing to perform on messages sent from caller to callee.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # caller-privacy always</pre>	
Step 40	exit	Exits the current configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit</pre>	
Step 41	exit	Exits the current configuration mode.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable)# exit</pre>	
Step 42	complete	Completes the CAC-policy or call-policy set after committing the full set.
	Example: Router(config-sbc-sbe-cacpolicy)# complete	
Step 43	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-cacpolicy)# exit	
Step 44	active-cac-policy-set	Sets the active CAC-policy-set within an SBE entity.
	Example: Router (config-sbc-sbe)# active-cac-policy-set 1	
Step 45	show sbc sbc-name sbe sip essential-methods	
	Example: Router(config-sbc-sbe)# show sbc mysbc sbe sip essential-methods	

Configuring Customized Offers for Late-to-Early Media Interworking

Prerequisites

Before performing this task, configure late-to-early media interworking per adjacency.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. sip sdp-media-profile profile-name
- 5. entry *entry-num*
- 6. **media-line** *index* "*media_description*"
- 7. (Optional) Repeat the previous step with a different *index* to add more media lines to this entry.
- 8. exit
- **9.** (Optional) Repeat Steps 6 through 9 with a different *entry-num* in Step 6 to add another entry to this profile.
- 10. exit
- 11. exit
- **12.** cac-policy-set policy-set-id
- **13.** cac-table cac-table-name
- 14. entry entry-number
- **15.** sip sdp-media-profile *profile-name*
- 16. Ctrl Z
- 17. show sbc sbc-name sbe sip sdp-media-profile profile-name

DETAILED STEPS

	Command or Action	Purpose	
Step 1	configure terminal	Enables global configuration mode.	
	Example: Router# configure terminal		
Step 2	sbc service-name	Enters the submode for configuring the method profile.	
	Example: Router(config)# sbc mysbc	Use the <i>service-name</i> argument to define the name of the service.	
Step 3	sbe	Enters the mode of an SBE entity within an SBC service.	
	Example: Router(config-sbc)# sbe		

	Command or Action	Purpose
Step 4	sip sdp-media-profile profile-name	Configures an SDP media profile for a customized offer. Enter into SIP SDP media profile configuration mode.
	Example: Router(config-sbc-sbe)# sip sdp-media-profile profile1	
Step 5	entry sequence-num	Enters the submode for adding a section of media description to the profile. A section, or entry, can contain
	Example: Router(config-sbc-sbe-sip-sdp-media)# entry 1	one or more media description lines.
Step 6	<pre>media-line index "media_description"</pre>	Adds a media description line to the entry. Quotation marks must surround the media description. See
	Example: Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 1 "m=audio 0 RTP/AVP 0"	?\$paranum>Rules for Media Lines in SDP Media Profiles? section on page 49-3.
Step 7	(Optional) Repeat the previous step with a different <i>index</i> to add more media lines to this entry.	Adds additional media descriptions to the entry. The index controls the ordering of the media descriptions.
	Example: Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 2 "a=rtpmap:12 H264/90000"	
Step 8	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-sip-sdp-media-ele)# exit	
Step 9	(Optional) Repeat Steps 5 through 8 with a different <i>entry-num</i> in Step 5.	Adds another entry to this profile.
	Example:	
	Router(config-sbc-sbe-sip-sdp-media)# entry 2 Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 1 "m=audio 0 RTP/AVP 0" Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 2 "a=rtpmap:0 PCMU/8000" Router(config-sbc-sbe-sip-sdp-media-ele)# exit	
Step 10	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-sip-sdp-media)# exit	
Step 11	exit	Exits the current configuration mode.
	Example: Router(config-sbc-sbe-sip)# exit	
Step 12	cac-policy-set policy-set-id	Enters the submode to make a change to a previously configured CAC policy set. Changes are not permitted to the
	Example: Router(config-sbc-sbe)# cac-policy-set 1	active policy set.

	Command or Action	Purpose
Step 13	cac-table cac-table-name	Enters the submode to make a change to a previously configured admission control table.
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table MyCacTable	
Step 14	entry entry-number	Enters the submode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	
Step 15	sip sdp-media-profile profile-name	Associates an SDP media profile with an admission control table entry.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) #sip sdp-media-profile profile1</pre>	
Step 16	Ctrl Z	Returns to user EXEC mode.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)#Ctrl Z	
Step 17	show sbc sbc-name sbe sip sdp-media-profile profile-name	Shows the contents of the profile. It is important to check the contents of the profile to make sure it is syntactically valid SDP as defined in RFC 2327. The command line
	Example: Router# show sbc test sbe sip sdp-media-profile profile1	interface does not check the syntax of the <i>media_description</i> arguments.

Configuration Examples for the Late-to-Early Media Interworking Feature

This section includes the following examples:

- Example: Late-to-Early Media Interworking, page 49-13
- Example: Customized Offer for Late-to-Early Media Interworking, page 49-16

Example: Late-to-Early Media Interworking

The following example shows a configuration of the Late-to-Early Media Interworking feature.



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

Router# configure terminal

Router(config) # **sbc mySbc** Router(config-sbc) # **sbe** Router(config-sbc-sbe) # adjacency sip SIPP-1 Router(config-sbe-adj-sip)# nat force-off Router(config-sbe-adj-sip)# preferred-transport udp Router(config-sbe-adj-sip)# redirect-mode pass-through Router(config-sbe-adj-sip)# authentication nonce timeout 300 Router(config-sbe-adj-sip)# signaling-address ipv4 201.201.201.20 Router(config-sbe-adj-sip)# signaling-port 5060 Router(config-sbe-adj-sip)# remote-address ipv4 202.202.202.11 255.255.255.255 Router(config-sbe-adj-sip)# signaling-peer 202.202.202.11 Router(config-sbe-adj-sip)# signaling-peer-port 5060 Router(config-sbe-adj-sip)# dbe-location-id 4294967295 Router(config-sbe-adj-sip)# account SIPP-1 Router(config-sbe-adj-sip)# reg-min-expiry 3000 Router(config-sbe-adj-sip) # media-late-to-early-iw incoming Router(config-sbe-adj-sip)# attach Router(config-sbe-adj-sip)# exit Router(config-sbe-adj)# exit Router(config-sbc-sbe) # adjacency sip SIPP-2 Router(config-sbe-adj-sip)# nat force-off Router(config-sbe-adj-sip)# preferred-transport udp Router(config-sbe-adj-sip)# redirect-mode pass-through Router(config-sbe-adj-sip)# authentication nonce timeout 300 Router(config-sbe-adj-sip)# signaling-address ipv4 201.201.201.20 Router(config-sbe-adj-sip)# signaling-port 5060 Router(config-sbe-adj-sip)# remote-address ipv4 201.201.201.11 255.255.255.255 Router(config-sbe-adj-sip)# signaling-peer 201.201.201.11 Router(config-sbe-adj-sip)# **signaling-peer-port 5060** Router(config-sbe-adj-sip)# dbe-location-id 4294967295 Router(config-sbe-adj-sip)# account SIPP-2 Router(config-sbe-adj-sip)# reg-min-expiry 3000 Router(config-sbe-adj-sip) # media-late-to-early-iw outgoing Router(config-sbe-adj-sip)# attach Router(config-sbe-adj-sip)# exit Router(config-sbe-adj)# exit Router(config-sbc-sbe) # sip inherit profile preset-core Router(config-sbc-sbe)# cac-policy-set 1 Router(config-sbc-sbe-cacpolicy) # first-cac-table table Router(config-sbc-sbe-cacpolicy)# first-cac-scope call Router(config-sbc-sbe-cacpolicy)# averaging-period 60 Router(config-sbc-sbe-cacpolicy) # cac-table table Router(config-sbc-sbe-cacpolicy-cactable) # table-type limit adjacency Router(config-sbc-sbe-cacpolicy-cactable)# entry 1 Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIPP-1 Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 64009 Gbps Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-updates 4294967295 Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-channels 4294967295 Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-type full-duplex Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-timeout 0 Router(config-sbc-sbe-cacpolicy-cactable-entry)# codec-restrict-to-list allowed_caller Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-codec-list allowed_caller Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-privacy never Router(config-sbc-sbe-cacpolicy-cactable-entry)# caller-privacy never Router(config-sbc-sbe-cacpolicy-cactable) # entry 2 Router(config-sbc-sbe-cacpolicy-cactable-entry)# match-value SIPP-2 Router(config-sbc-sbe-cacpolicy-cactable-entry)# action cac-complete Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-bandwidth 64009 Gbps Router(config-sbc-sbe-cacpolicy-cactable-entry)# max-updates 4294967295 Router(config-sbc-sbe-cacpolicy-cactable-entry) # max-channels 4294967295 Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-type full-duplex Router(config-sbc-sbe-cacpolicy-cactable-entry)# early-media-timeout 0 Router(config-sbc-sbe-cacpolicy-cactable-entry)# codec-restrict-to-list allowed

```
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-codec-list allowed
Router(config-sbc-sbe-cacpolicy-cactable-entry)# callee-privacy never
Router (config-sbc-sbe-cacpolicy-cactable-entry) # caller-privacy never
Router(config-sbc-sbe-cacpolicy-cactable-entry) # exit
Router(config-sbc-sbe-cacpolicy-cactable)# exit
Router(config-sbc-sbe-cacpolicy) # complete
Router(config-sbc-sbe-cacpolicy)# exit
Router (config-sbc-sbe) # active-cac-policy-set 1
Router(config-sbc-sbe) # retry-limit 3
Router(config-sbc-sbe)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy) # first-call-routing-table start-table
Router(config-sbc-sbe-rtgpolicy)# rtg-src-adjacency-table start-table
Router(config-sbc-sbe-rtgpolicy-entry)# entry 1
Router(config-sbc-sbe-rtgpolicy-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-entry)# dst-adjacency SIPP-1
Router(config-sbc-sbe-rtgpolicy-entry)# match-adjacency SIPP-2
Router(config-sbc-sbe-rtgpolicy-entry)# exit
Router(config-sbc-sbe-rtgpolicy) # entry 2
Router(config-sbc-sbe-rtgpolicy-entry) # action complete
Router(config-sbc-sbe-rtgpolicy-entry)# dst-adjacency SIPP-2
Router(config-sbc-sbe-rtgpolicy-entry)# match-adjacency SIPP-1
Router(config-sbc-sbe-rtgpolicy-entry)# exit
Router(config-sbc-sbe-rtgpolicy) # complete
Router(config-sbc-sbe-rtgpolicy)# exit
Router(config-sbc-sbe)# active-call-policy-set 1
Router(config-sbc-sbe) # sip max-connections 2
Router(config-sbc-sbe)# sip timer
Router(config-sbc-sbe-tmr)# tcp-idle-timeout 120000
Router(config-sbc-sbe-tmr)# tls-idle-timeout 3600000
Router(config-sbc-sbe-tmr)# udp-response-linger-period 32000
Router(config-sbc-sbe-tmr)# udp-first-retransmit-interval 500
Router(config-sbc-sbe-tmr)# udp-max-retransmit-interval 4000
Router(config-sbc-sbe-tmr)# invite-timeout 180
Router(config-sbc-sbe-tmr)# exit
Router(config-sbc-sbe)# codec-list allowed
Router(config-sbc-sbe-codec-list)# description allowed codecs
Router(config-sbc-sbe-codec-list)# codec PCMA
Router(config-sbc-sbe-codec-list)# codec PCMU
Router(config-sbc-sbe-codec-list)# exit
Router(config-sbc-sbe) # codec-list allowed_caller
Router(config-sbc-sbe-codec-list)# description caller
Router(config-sbc-sbe-codec-list)# codec PCMA
Router(config-sbc-sbe-codec-list)# exit
Router(config-sbc-sbe)# h323
Router(config-sbc-sbe-h323)# ras timeout arg 5000
Router(config-sbc-sbe-h323)# ras retry arg 2
Router(config-sbc-sbe-h323)# ras timeout brg 3000
Router(config-sbc-sbe-h323)# ras retry brg 2
Router(config-sbc-sbe-h323) # ras timeout drg 3000
Router(config-sbc-sbe-h323)# ras retry drg 2
Router(config-sbc-sbe-h323) # ras timeout grg 5000
Router(config-sbc-sbe-h323) # ras retry grq 2
Router(config-sbc-sbe-h323) # ras timeout rrg 3000
Router(config-sbc-sbe-h323)# ras retry rrg 2
Router(config-sbc-sbe-h323)# ras rrg ttl 60
Router(config-sbc-sbe-h323) # ras timeout urg 3000
Router(config-sbc-sbe-h323)# ras retry urg 1
Router(config-sbc-sbe-h323)# h225 timeout proceeding 10000
Router(config-sbc-sbe-h323) # h225 timeout establishment 180000
Router(config-sbc-sbe-h323) # h225 timeout setup 4000
Router(config-sbc-sbe-h323) # exit
Router(config-sbc-sbe) # h323
Router(config-sbc-sbe-h323) # adjacency timeout 30000
Router(config-sbc-sbe-h323)# exit
```

```
Router(config-sbc-sbe)# redirect-limit 2
Router(config-sbc-sbe)# deact-mode normal
Router(config-sbc-sbe)# activate
Router(config-sbc-sbe)# exit
Router(config-sbc-dbe)# media-address ipv4 201.201.201.201
Router(config-sbc-dbe)# location-id 0
Router(config-sbc-dbe)# media-timeout 9000
Router(config-sbc-dbe)# deact-mode normal
Router(config-sbc-dbe)# activate
```

Example: Customized Offer for Late-to-Early Media Interworking

The following example configures a customized media description and assigns it to a CAC policy.

```
Router(config)# sbc test
Router(config-sbc)# sbe
Router(config-sbc-sbe) # sip sdp-media-profile MediaProfile
Router(config-sbc-sbe-sip-sdp-media)# entry 1
Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 1 "m=audio 0 RTP/AVP 31"
Router(config-sbc-sbe-sip-sdp-media-ele)# media-line 2 "a=aaa:testing"
Router(config-sbc-sbe-sip-sdp-media-ele)# Ctrl Z
Router# show sbc test sbe sip sdp-media-profile MediaProfile
 SDP media profile "MediaProfile"
   Elements:
      Sequence Number : 1
          media-line 1 : m=audio 0 RTP/AVP 31
          media-Line 2 : a=aaa:testing
Not in use by any CAC table entries
Router# configure terminal
Router(config) # sbc test
Router(config-sbc) # sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table testpolicytable
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# sip sdp-media-profile MediaProfile
Router(config-sbc-sbe-cacpolicy-cactable-entry)Ctrl Z
Router# show sbc test sbe sip sdp-media-profile MediaProfile
 SDP media profile "MediaProfile"
    Elements:
      Sequence Number : 1
          media-line 1 : m=audio 0 RTP/AVP 31
          media-line 2 : a=aaa:testing
In use by CAC table testpolicytable, entry 1
```

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Verification

Use the commands listed in Table 49-1 to verify operation.

Table 49-1Commands to Verify Operation

Command	Purpose
show sbc <i>sbc-name</i> sbe cac-policy-set <i>id</i> table <i>name</i> entries	Lists a summary of the CAC policy tables associated with the given policy set.
show sbc sbc-name sbe adjacencies	Lists the adjacencies configured on SBEs.
show sbc sbc-name sbe sdp-profiles	Lists the SIP SDP media profiles defined under a named SBE and indicates whether they are currently associated with a CAC policy.
<pre>show sbc sbc-name sbe sip sdp-media-profile [profile-name]</pre>	Lists the SIP SDP media profiles defined under a named SBE and indicates whether they are currently associated with a CAC policy, or, if you include a profile name, shows the contents of the named profile.

The following example shows adjacencies.

SBC	Service "test"	
Ad	djacency asr1k-1 (SIP)	
	Status:	Attached
	Signaling address:	22.22.22.2:5060, VRF Admin
	Signaling-peer:	33.33.33.3:5060
	Remote address:	33.33.33.3 255.255.255.255
	Force next hop:	No
	Account:	
	Group:	None
	In header profile:	Default
	Out header profile:	Default
	In method profile:	Default
	Out method profile:	Default
	In UA option prof:	Default
	Out UA option prof:	Default
	In proxy opt prof:	Default
	Out proxy opt prof:	Default
	Priority set name:	None
	Local-id:	None
	Rewrite REGISTER:	Off
	Target address:	None
	NAT Status:	Auto Detect
	Reg-min-expiry:	3000 seconds
	Fast-register:	Enabled
	Fast-register-int:	30 seconds
	Authenticated mode:	None
	Authenticated realm:	None
	Auth. nonce life time:	300 seconds
	IMS visited NetID:	None
	Inherit profile:	Default
	Force next hop:	No
	Home network Id:	None
	UnEncrypt key data:	None
	SIPI passthrough:	No

Router# show sbc test sbe adjacencies asr1k-1 de

Rewrite from domain: Yes Rewrite to header: Yes Rewrite to neaser.NoMedia passthrough:NoHunting Triggers:Global TriggersRedirect mode:Pass-through Redirect mode: Security: Untrusted Outbound-flood-rate: None Ping-enabled: No Signaling Peer Status: Not Tested media-late-to-early-iw: incoming Router# show sbc test sbe adjacencies asr1k-2 de SBC Service "test" Adjacency asr1k-2 (SIP) Status: Attached Signaling address: 22.22.22.5061, VRF Admin Signaling-peer: 44.44.44.4:5061 Remote address: 44.44.44.4 255.255.255.255 Force next hop: No Account: None Group: In header profile: Default Out header profile: Default In method profile: Default Out method profile: Default In UA option prof: Default Out UA option prof: Default In proxy opt prof: Default Out proxy opt prof: Default Priority set name: None Local-id: None Off None Rewrite REGISTER: Target address: NAT Status: Auto Detect Reg-min-expiry: 3000 seconds Fast-register: Enabled Fast-register-int: 30 seconds Authenticated mode: None Authenticated realm: None Auth. nonce life time: 300 seconds IMS visited NetID: None Inherit profile: Default No Force next hop: Home network Id: None UnEncrypt key data: None SIPI passthrough: No Rewrite from domain: Yes Rewrite to header: Yes Media passthrough: No Hunting Triggers: Global Triggers Redirect mode: Pass-through Redirect mode: Security: Untrusted Outbound-flood-rate: None Ping-enabled: No Signaling Peer Status: Not Tested media-late-to-early-iw: outgoing

The following command lists a summary of the CAC policy tables associated with the given policy set: Router# show sbc test sbe cac-policy-set 1 table table entry 1

SBC Service "test"

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```
Policy set 1 table table entry 1
 Match value SIPP-1
 Action
                          CAC policy complete
 Max updates
                         Unlimited
 Max bandwidth
                         Unlimited
 Max channels
                         Unlimited
                          Allowed
 Transcoder
 Caller privacy setting Never hide
 Callee privacy setting Never hide
 Early media
                          Allowed
 Early media direction
                          Both
 Early media timeout
                          0
 Caller voice QoS profile default
 Caller video QoS profile default
 Caller sig QoS profile
                          default
 Callee voice QoS profile default
 Callee video QoS profile default
 Callee sig QoS profile
                          default
 Restrict codecs to list
                         allowed_caller
 Restrict caller codecs to list allowed_caller
 Restrict callee codecs to list default
                         Allowed
 Media bypass
 Number of calls rejected by this entry
                                          0
Router# show sbc test sbe cac-policy-set 1 table table entry 2
SBC Service "test"
Policy set 1 table table entry 2
                          SIPP-2
 Match value
 Action
                          CAC policy complete
                         Unlimited
 Max updates
 Max bandwidth
                         Unlimited
 Max channels
                         Unlimited
 Transcoder
                          Allowed
 Caller privacy setting Never hide
 Callee privacy setting Never hide
 Early media
                          Allowed
 Early media direction
                          Both
 Early media timeout
                          0
 Caller voice QoS profile default
 Caller video QoS profile default
 Caller sig QoS profile
                          default
 Callee voice QoS profile default
 Callee video QoS profile default
                         default
 Callee sig QoS profile
 Restrict codecs to list
                         allowed
 Restrict caller codecs to list
                                default
 Restrict callee codecs to list
                                 allowed
 Media bypass
                          Allowed
 Number of calls rejected by this entry
                                          0
Router#
```

The following example shows a list of SDP media profiles configured under an SBC service:

Router# **show sbc test sbe sip sdp-media-profile** SDP Media profiles for SBC service "test"

The following example shows the contents of a named SDP media profile:

Router# show sbc test sbe sip sdp-media-profile MediaProfile

SDP media profile "MediaProfile"
Elements:
 Sequence Number : 1
 media-Line 1 : m=audio 0 RTP/AVP 31
 media-Line 2 : a=aaa:testing

In use by CAC table testpolicytable, entry 1



Early Media

The Early Media feature is supported for Session Initiation Protocol (SIP)calls. Early Media is the ability of two user agents to communicate before a call is actually established. Support for early media is important both for interoperability with the Public Switched Telephone Network (PSTN) and billing purposes.

Early Media is defined when media begins to flow before the call is officially connected. Media channels are set up prior to the call connection. These channels are used to provide the ring tone that the caller hears and are not generated by the caller's endpoint or other queuing services, for example, hold music.

Note

For Cisco IOS XR Software Release and later, this feature is supported in the unified model only.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Feature History for Early Media

Release	Modification
Cisco IOS XR Software Release	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for the Early Media Support, page 50-1
- Information About Early Media, page 50-2

Restrictions for the Early Media Support

The restrictions for Early Media Support are:

- Cisco Unified Border Element (SP Edition) offers support for the gateway model of early media (as defined in RFC 3960).
- Early media does not work with endpoints which send late SDP.
- Cisco Unified Border Element (SP Edition) does not currently support RFC 3312.

Information About Early Media

Current implementations support early media through the 183 response code. When the called party wishes to send early media to the caller, it sends a 183 response to the caller. This response contains the Session Description Protocol (SDP). When the caller receives the response, it suppresses any local alerting of the user (for example, audible ring tones or a pop-up window) and begins playing out the media that it receives. The SDP in the 183 response provides an address, to which the real-time control protocol (RTCP) packets can be sent.

Some implementations take media from the caller, and send it to the callee as well. If the call is ultimately rejected, the called party generates a non-2xx final response. When this response is received by the caller, it ceases playing out, or sending media. However, if the call is accepted, the called party generates a 2xx response (generally, with the same SDP as in the 183 response), and sends it to the caller. The media transmission continues as before.

In addition, Cisco Unified Border Element (SP Edition) supports the following for early media:

- Renegotiation of the media after early media is flowing (before and after the call is connected). Media renegotiation is supported on Cisco Unified Border Element (SP Edition) using the PRACK and UPDATE methods.
- Optional SIP UPDATE support by SIP endpoints (including early media without UPDATE support).
- RFC 3312 preconditions.
- Configurable SIP support of Required, Supported, and Proxy-Require headers.
- A per-adjacency flag to allow interoperability with the Cisco Gateway's non-standard PRACK behavior.



SIP Instant Messaging

Cisco Unified Border Element (SP Edition) supports SIP instant messaging (IM). Two options for SIP instant messaging are configurable—record-route passthrough and privacy for SIP Instant Messaging.

The typical SIP instant messaging implementation uses end-to-end Record-Route passthrough, but privacy is not applied.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for SIP Instant Messaging

Release	Modification
Cisco IOS XE Release	The SIP Instant Messaging feature was introduced on the Cisco IOS XR.
2.5	

Contents

This chapter contains the following sections:

- SIP Instant Messaging, page 51-1
- Configurable Options for SIP Instant Messaging, page 51-2
- SDP Handling for SIP Instant Messaging, page 51-5

SIP Instant Messaging

Cisco Unified Border Element (SP Edition) supports SIP instant messaging (IM). For SIP instant messaging, SBC handles the calls as follows:

- Calls with the Media Announcement m=message or m=x-ms-message are allowed.
- Messages containing multiple m=message lines are permitted.

- SBC does not allocate any bandwidth to support IM dialogs because there is no media stream for such dialogs.
- The IM message is rejected if the SDP also contains any other media types.
- SBC forwards the SDP body unchanged unless privacy is configured.

SBC can be configured to pass through Record-Route headers from one side of the IM dialog to the other so that the end-to-end Route header set is available to caller and callee. In this case, SBC remains in the flow of messages by appending Record-Route headers and not rewriting the Contact.

Configurable Options for SIP Instant Messaging

Two options for SIP instant messaging are configurable and explained in the following sections:

- Record-Route Passthrough for SIP Instant Messaging, page 51-2
- Privacy for SIP Instant Messaging, page 51-4

The typical SIP instant messaging implementation will use end-to-end Record-Route passthrough but will not apply privacy.

Record-Route Passthrough for SIP Instant Messaging

This section explains Record-Route passthrough and contains the following subsections:

- Record-Route Passthrough Overview, page 51-2
- Registered Subscribers and Record-Route Passthrough, page 51-3
- Record-Route Passthrough Configuration, page 51-3

Record-Route Passthrough Overview

A Record-Route set represents the hop-by-hop route SIP messages must traverse between two endpoints as part of a SIP dialog. The final hop to the endpoint is represented by the Contact header. SBC's normal behavior is to rewrite the Contact and maintain two independent Record-Route sets, one for each side of the call. Each of these Record-Route sets represents the route between the endpoint and the adjacency on that side of the call.

Passing through the end-to-end Record-Route set means that the adjacencies on a call do not represent the last hop and therefore must not be identified by the Contact header. When passing through the end-to-end Record Route set, SBC also passes through the end-to-end Contacts without rewriting them. In this case, SBC remains in the flow by appending Record-Route headers representing the inbound and outbound adjacencies.

If inbound and outbound adjacencies have conflicting Record-Route passthrough configurations, the setting of the inbound adjacency is used. For example, if the inbound adjacency enables Record-Route passthrough but the outbound adjacency does not, the outbound adjacency will forward the supplied Record-Route set, append a Record-Route header corresponding to the outbound adjacency, and leave the Contact header unaltered.

Registered Subscribers and Record-Route Passthrough

Record-Route passthrough behavior does not apply to SIP messages received from or going to Registered endpoints regardless of the inbound adjacency's Record-Route passthrough configuration.

The SIP specification (RFC 326) mandates that registrars ignore Record-Route headers present on a REGISTER message. Therefore to ensure that SBC remains in subsequent dialogs created to or from registered subscribers, SBC must rewrite the Contact in REGISTER messages. SBC updates the Contacts in later dialogs created by a registered endpoint so that they match the Contact previously published to network. At the point that SBC rewrites the Contact, SBC terminates any existing Record-Route set and creates a new one.

Record-Route Passthrough Configuration

To pass through Record-Route sets, SBC provides configuration on a per adjacency basis using the **passthrough header record-route** command.

- If turned off (**no passthrough header record-route**), the Record-Route set is cached and returned to the endpoint. Thereafter, the Record-Route set is used to build Route headers on subsequent outgoing messages. Neither end will see the entire end-to-end Record-Route set. This is the default behavior.
- If turned on, the request is not from or to a registered subscriber, the following occurs:
 - For dialog-creating requests, any Record-Route set present on the request is passed through SBC and forwarded to the receiving endpoint.
 - The Contact present on the request is passed through SBC and forwarded to the receiving endpoint.
 - Record-Route headers representing the inbound and outbound adjacency are appended to the request.
 - The end-to-end Record-Route set is passed back through SBC on the response and forwarded to the calling endpoint. Both endpoints see the entire end-to-end Record-Route set.
 - The preceding will be the behavior regardless of outbound adjacency configuration.
 - Subsequent in-dialog requests do not have their Request URI updated to match the Contact received on the dialog-creating request.

If a request is from or to a registered subscriber, it is processed as though **record-route passthrough** was turned off.

If topology hiding or privacy is applied to a call, the Record-Route set is stripped from the request regardless of the **record-route passthrough** configuration.

Configuring Record-Route passthrough does not result in the Route headers being passed through on subsequent messages.

With Record-Route passthrough enabled, messages in an IM dialog are adjusted as described earlier in this section. As an example of these Record-Route passthrough adjustments, the following requests show how enabling Record-Route passthrough affects an INVITE request that is used for an IM dialog. In the first example, Record-Route passthrough is not used. In the second example, Record-Route passthrough is enabled, and SBC adds the Record-Route information in bold font to the INVITE request. In both examples, privacy is not used.

In these examples, the INVITE requests are outbound from the SBC to a SIP proxy server.

Outbound INVITE (without Record-Route passthrough):

INVITE sip:callee@callee.com SIP/2.0

```
Via: SIP/2.0/UDP sbc.com;branch=z9hG4bK-sbc-1
From: Caller <sip:caller@dcsbc.com>;tag=dcsbc
To: Callee <sip:callee@callee.com>
Call-ID: sbc-call-1
Contact: Caller <sip:abcd@sbc.com>
Content-Type: application/sdp
Content-Length: 135
v=0
o=- 0 0 IN IP4 192.168.1.121
s=session
c=IN IP4 192.168.1.121
t=0 0
m=message 5060 sip sip:caller@caller.com
c=IN IP4 192.168.1.122
```

Outbound INVITE (with Record-Route passthrough):

```
INVITE sip:callee@callee.com SIP/2.0
From: Caller <sip:caller@caller.com>;tag=caller-tag-1
To: Callee <sip:callee@callee.com>
Call-ID: call-1
Contact: Caller <sip:caller@caller.com>
Record-Route: <sip:proxy1@example.com;lr>
Record-Route: <sip:caller_adj@sbc.com;lr>
Record-Route: <sip:callee_adj@sbc.com;lr>
```

Privacy for SIP Instant Messaging

For IM privacy, the CAC configuration that is used for normal calls is also used for IM dialogs. The CAC table entry command option **caller-privacy** configures privacy. For information on using this command, see the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

If privacy is enabled, SBC adjusts three fields in the SIP IM SDP to make them anonymous.

- The Connection lines (c=) may contain sensitive IP address information in the connection address field. This information is hidden by inserting a Session Description level Connection line with the local address of the corresponding adjacency. Any other Connection lines are removed.
- The o= line is replaced with SBC's own information. The address field in this line is set to match the c= line.
- For SDP indicating a SIP IM stream, the Media Descriptions lines (m=) will be of the form:

m=message port[/number_of_ports] sip format_list

In the preceding, *format_list* is a SIP URL and may contain user sensitive information. The user-sensitive information is hidden by stripping all messages indicating Media Description lines and forwarding an Offer containing the following Media Description line:

m=message 5060 sip sip:anonymous@192.168.10.10

The following examples show how enabling **caller-privacy** affects the SDP messages that are used for an IM dialog. In the examples, 192.168.10.10 is the local address of the outbound adjacency, and 192.168.2.20 is the address of the inbound adjacency. In the examples, the information in bold font has been adjusted to make the caller anonymous. The following example is outbound from the SBC to a SIP proxy server.

Outbound INVITE (without privacy):

v=0

```
o=- 0 0 IN IP4 192.168.1.121
s=session
c=IN IP4 192.168.1.121
t=0 0
m=message 5060 sip sip:caller@caller.com
c=IN IP4 192.168.1.122
```

Outbound INVITE (with privacy):

```
v=0
o=- 0 0 IN IP4 192.168.10.10
s=session
c=IN IP4 192.168.10.10
t=0 0
m=message 5060 sip sip:anonymous@192.168.10.10
c=IN IP4 192.168.10.10
```

The following example is outbound from the SBC to a SIP gateway.

Outbound 200 OK (without privacy):

```
v=0
o=- 0 0 IN IP4 192.168.2.200
s=session
c=IN IP4 192.168.2.200
t=0 0
m=x-ms-message 5060 sip null
```

Outbound 200 OK (with privacy)

```
v=0
o=- 0 0 IN IP4 192.168.2.20
s=session
c=IN IP4 192.168.2.20
t=0 0
m=x-ms-message 5060 sip sip:anonymous@192.168.2.20
```

SDP Handling for SIP Instant Messaging

In the SDP, SBC identifies IM dialogs by the presence of m=message or m=x-ms-message. For an offer that includes both IM and audio/video media lines, SBC rejects the offer with error code 488, Not Acceptable Here. After the initial media negotiation, any subsequent reoffer that attempts to change a call from an IM to a non-IM or vice versa is also rejected with error code 488.

SBC passes through SDP for IM dialogs unchanged with the exception of modifications due to privacy (see ?\$paranum>Privacy for SIP Instant Messaging? section on page 51-4). SBC does not allocate any gates or media pinholes.

The following sections provide more information on how SBC handles SDP for SIP instant messaging:

- ?\$paranum>SIP URLs in the SDP? section on page 51-6
- ?\$paranum>Miscellaneous SDP Handling? section on page 51-6

SIP URLs in the SDP

SDP attached to SIP IM dialogs can contain SIP URLs in the Media Description lines. For example:

m=message 5060 sip sip:example@home.net

If an endpoint sends an in-dialog message to the URL in the SDP (by placing the URL in the Request URI), the behavior is as follows:

- If SBC has not been configured to pass through the Record-Route set, the request may fail to route.
- If SBC has been configured to pass through the Record-Route set, the request will be forwarded by SBC without altering the Request URI.

For information on Record-Route passthrough, see the ?\$paranum>Record-Route Passthrough for SIP Instant Messaging? section on page 51-2.

Miscellaneous SDP Handling

This section describes some miscellaneous SDP handling for SIP instant messaging.

- Transcoding—Since there is no media stream, SBC never attempts to bring in a transcoder. If the callee endpoint does not support the media type, the call fails.
- Special Media Descriptions—In some cases, SDP may include a proprietary media type of m=x-ms-message. SBC treats m=x-ms-message exactly the same as m=message. No support is added for any other proprietary media types.
- Interworking—SIP instant messaging does not interwork for SIP/H.323 calls, or for SIP-SIP late to early media interworking calls. If an IM dialogs is used in these scenarios, call setup fails with the response code 488, Not Acceptable Here.
- Invalid Connection Address—SBC does not edit the SDP of SIP IM dialogs except when privacy is configured. Therefore, the connection address passed through SBC may not be valid as far as the receiving endpoint is concerned. This is acceptable because no media is flowing between the endpoints.



Integration of Resource Management and SIP

As per IETF RFC 3312, call endpoints can determine whether resources are fully reserved for a media stream before using it. This feature is useful when separate quality of service (QoS) signaling, such as Resource ReSerVation Protocol (RSVP), is used. To accomplish this, RFC 3312 defines three new a=lines at media stream granularity. Endpoints use these lines to signal reservation information and their preconditions for adopting the new Session Description Protocol (SDP).

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

Note

For Cisco IOS XE Release 2.4 and later, this feature is supported in the unified model only.

Feature History for Integration of Resource Management and SIP Support

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco IOS XR along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for Integration of Resource Management, page 52-1
- Information about Integration of Resource Management, page 52-2

Restrictions for Integration of Resource Management

The restrictions for integration of resource management are:

- When this feature is implemented, Cisco Unified Border Element (SP Edition) does not report the media state or generate preconditions. It only detects whether preconditions are present, and whether all the mandatory preconditions have been met if preconditions exist.
- This feature is a SIP-only feature and is not supported by H.323 or SIP-H.323 interworking.
- With RFC 3312 signaling procedures, media renegotiation is completed only when the mandatory preconditions have been met.

Information about Integration of Resource Management

When the precondition tag appears in the Require or Supported header fields of SIP messages, Cisco Unified Border Element (SP Edition) allows them to pass through. Cisco Unified Border Element (SP Edition) also allows the unmodified SDP to pass through, which represents the state and the preconditions.

When processing an offer results in failure, the underlying SIP message is either rejected or the call is torn down. When processing an answer results in failure, the call is torn down, regardless of the reason for the failure.

Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model



ENUM Client

Cisco Unified Border Element (SP Edition) supports E.164 Number Mapping (ENUM).

Feature History for Implementing SNMP

Release	Modification
Cisco IOS XE Release	ENUM Client Feature was introduced.
3.1S	

Contents

- Information about ENUM Client Configuration, page 53-1
- Configuring ENUM Client, page 53-3
- Configuration Examples of ENUM Client Configuration, page 53-11

Information about ENUM Client Configuration

E.164 Number Mapping (ENUM) is an IETF standard protocol for converting telephone numbers into IP addresses (and vice versa), so that the telephone numbers can be maintained by a DNS server.

The SBC ENUM client is configurable and accepts the ITU standard format for international telephone numbers, E.164: country code, area code, phone number.

The ENUM client translates telephone numbers into standard sip/sips URIs that are resolved by a DNS server and then stored in an SBC routing table. Currently, only IPv4 is supported.

When a telephone number is called, the ENUM client queries the DNS server for a sip/sips URI. The DNS server returns the URI to the ENUM client, and the ENUM client stores the URI in an SBC routing table.

Destination Address

The destination address of a called number is typically derived from the Request URI. However, the destination address may also be derived from other headers in the routing table, such as the *To:* header or the *P*-*Called*-*Party-ID:* header.

The ENUM Client feature provides the user with the ability to configure a prioritized list of headers. This list may consist of any non-essential SIP headers, including the *To:* header and the Request URI. Once the list is configured, SBC can derive destination addresses for called numbers from this list of headers.

Destination address headers are stored in the header filter profile MIB table. Destination addresses must conform to the address syntax specification defined in RFC 3261. An address header list may contain a maximum to 10 entries.

The ENUM Client first searches the Request URI. If it does not find a match for the called number, it then searches the header list.

Source Address

The source address of a calling party number is typically derived from the *From:* header. The source addresses can be modified using the following configuration.

```
header-profile <name> src-address
```

You can also configure a prioritized list of headers from which the source address for a calling number is derived. This list may consist of any non-essential SIP headers.

Source address headers are stored in the header filter profile MIB table. Source addresses must conform to the address syntax specification defined in RFC 3261. An address header list may contain a maximum to 10 entries.

Diverted-by Address

The ENUM Client feature also provides support for deriving the source number from a prioritized list of headers for calls which have been diverted by another number. If a call has been diverted by another number, the source address must be derived from the diverted-by list of headers. Users can also configure a header action to reject these types of calls.

Header Profiles

The user can configure actions to be performed on a target address by configuring a header profile.

The following actions can be configured in a header profile for a target address:

- goto-table-name
- complete
- reject

For the SBC ENUM client configuration steps, see the ?\$paranum>Configuring ENUM Client? section on page 53-3.

For an example of SBC ENUM client configuration see the ?\$paranum>Configuration Examples of ENUM Client Configuration? section on page 53-11.

Additionally, you can also configure the SIP DNS cache, using the following commands:

- cache lifetime—configures the lifetime of a cached DNS entry.
- cache limit—configures the maximum number of entries that are permitted in the cache

Configuring ENUM Client

The sections presents two configurations:

- Configuring ENUM Client, page 53-3
- Configuring a Call Policy for Multiple ENUM Entries, page 53-4

Configuring ENUM Client

Use the following procedure to configure and ENUM client:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. enum enum-id
- 5. req-timeout timeout
- 6. max-recursive-depth number
- 7. entry entry-name
- 8. server ipv4 *ip_address* [vrf *vrf_name*]
- 9. dial-plan-suffix suffix
- 10. max-responses number
- 11. activate
- 12. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example:	
	Router(config)# sbc MySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	enum enum-id	Assigns the ENUM CUSTOMER ID number and enters ENUM configuration mode. Currently, only the number 1 is allowed
	Example: Router(config-sbc-sbe)# enum 1	
Step 5	req-timeout timeout	Configures the ENUM request timeout period.
	Example: Router(config-sbc-sbe-enum)# req-timeout 10000	
Step 6	max-recursive-depth number	Configures the maximum number of recursive ENUM look-ups for non-terminal Resource Records (RR).
	Example: Router(config-sbc-sbe-enum)# max-recursive-depth 100	
Step 7	entry entry-name	Configures the ENUM Client entry name and enter the ENUM entry configuration mode.
	Example: Router(config-sbc-sbe-enum)# entry ENUM_1	
Step 8	<pre>server ipv4 ip_address [vrf vrf_name]</pre>	Configures the IPv4 address of a DNS server for ENUM Client and optionally associates the DNS server to a VRF.
	Example: Router(config-sbc-sbe-enum-entry)# server ipv4 10.10.10.10 vrf VRF1	
Step 9	dial-plan-suffix suffix	Configures the dial plan suffix used for the ENUM query.
	Example: Router(config-sbc-sbe-enum-entry)# dial-plan-suffix Example.Suffix	
Step 10	max-responses number	Configures the maximum number of ENUM records returned to the routing module.
	Example: Router(config-sbc-sbe-enum)# max-responses 100	
Step 11	activate	Activates ENUM Client.
	Example: Router(config-sbc-sbe-enum)# activate	
Step 12	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	

Configuring a Call Policy for Multiple ENUM Entries

Use the following procedure to configure a call policy for multiple ENUM entries:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. enum enum-id
- 5. entry (enum) entry-name
- 6. server ipv4 *ip_address* [vrf *vrf_name*]
- 7. dial-plan-suffix suffix
- 8. entry (enum) entry-name
- 9. server ipv4 *ip_address* [vrf *vrf_name*]
- 10. dial-plan-suffix suffix
- 11. activate
- 12. exit
- 13. sip header-profile profile-name
- 14. dst-address
 or
 src-address
 or
 div-address
- 15. header-prio priority-level header-name header-name
- 16. exit
- 17. call-policy-set policy-set-id
- 18. first-call-routing-table table-name
- 19. rtg-src-adjacency-table table-id
- 20. entry entry-id
- 21. enum enum-id entry (enum) entry-name
- 22. action next-table goto-table-name
- 23. entry entry-id
- 24. match-adjacency key
- 25. enum enum-id entry (enum) entry-name
- 26. dst-adjacency target-adjacency
- 27. action complete
- 28. rtg-dst-address-table table-id
- 29. entry entry-id
- **30.** match-address key
- 31. dst-adjacency target-adjacency
- 32. action complete
- 33. entry entry-id

- **34.** match-address *key*
- **35.** dst-adjacency target-adjacency
- 36. action complete
- 37. entry entry-id
- **38.** match-address *key*
- 39. prefix
- **40. dst-adjacency** *target-adjacency*
- 41. action complete
- 42. complete
- 43. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc MySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	enum enum-id	Assigns the ENUM ID number and enters ENUM configuration mode. Currently, only the number 1 is
	Example: Router(config-sbc-sbe)# enum 1	allowed.
Step 5	entry (enum) entry-name	Configures the default ENUM entry and enters ENUM entry configuration mode.
	Example: Router(config-sbc-sbe-enum)# entry default-enum	
Step 6	<pre>server ipv4 ip_address [vrf vrf_name]</pre>	Configures the IPv4 address of a DNS server for the ENUM Client.
	Example: Router(config-sbc-sbe-enum-entry)# server ipv4 10.10.10.10	

	Command or Action	Purpose
Step 7	dial-plan-suffix suffix	Configures the dial plan suffix used for this ENUM query.
	Example: Router(config-sbc-sbe-enum-entry)# dial-plan-suffix e164.arpa	
Step 8	entry (enum) entry-name	Configures another ENUM entry and enters ENUM entry configuration mode.
	Example: Router(config-sbc-sbe-enum-entry)# entry cisco-enum	
Step 9	<pre>server ipv4 ip_address [vrf vrf_name]</pre>	Configures the IPv4 address of a DNS server for ENUM Client and associates the DNS server to a VRF.
	Example: Router(config-sbc-sbe-enum-entry)# server ipv4 10.0.0.22 vrf cisco-vrf	
Step 10	dial-plan-suffix suffix	Configures the dial plan suffix used for this ENUM query.
	Example: Router(config-sbc-sbe-enum-entry)# dial-plan-suffix cisco.com	
Step 11	activate	Activates the ENUM client.
	Example: Router(config-sbc-sbe-enum-entry)# activate	
Step 12	exit	Exits to the previous mode.
	Example: Router(config-sbc-sbe-enum)# exit	
Step 13	sip header-profile profile-name	Configures a header profile in the mode of an SBE entity.
	Example: Router(config-sbc-sbe)# sip header-profile enum	
Step 14	dst-address	Enters destination address submode.
	src-address	or Enters source address submode.
	or div-address	or Enters diverted-by address submode.
	Example:	
	Router(config-sbc-sbe-sip-hdr)# dst-address or	
	Router(config-sbc-sbe-sip-hdr)# src-address or	
	Router(config-sbc-sbe-sip-hdr)# div-address	

	Command or Action	Purpose
Step 15	header-prio priority-level header-name header-name	Configures the priority of the header from which the destination address is derived.
	<pre>Example: Router(config-sbc-sbe-sip-hdr-dst)# header-prio 1 header-name Dst_Add_Hdr_1 or</pre>	Configures the priority of the header from which the source address is derived. or
	Router(config-sbc-sbe-sip-hdr-src)# header-prio 1 header-name Src_Add_Hdr_1 or	diverted-by address is derived.
	Router(config-sbc-sbe-sip-hdr-div)# header-prio 1 header-name Div_Add_Hdr_1	
Step 16	exit	Exits to the previous mode.
	Example: Router(config-sbc-sbe-sip-hdr-src)# exit	
Step 17	call-policy-set policy-set-id	Creates a new call policy set and enters SBE routing policy configuration mode.
	<pre>Example: Router(config-sbc-sbe-sip-hdr)# call-policy-set 1</pre>	
Step 18	first-call-routing-table table-name	Configures the name of the first policy table to process when performing the routing stage of policy for new-call
	<pre>Example: Router(config-sbc-sbe-rtgpolicy)# first-call-routing-table rt1</pre>	events.
Step 19	rtg-src-adjacency-table table-id	Enters the configuration mode of the existing routing table, in this case, rt1.
	Example: Router(config-sbc-sbe-rtgpolicy)# rtg-src-adjacency-table rt1	
Step 20	entry entry-id	Creates an entry in the routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 2	
Step 21	enum enum-id entry (enum) entry-name	Configures the default ENUM entry for the routing table entry.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # enum 1 entry default-enum</pre>	
Step 22	action next-table goto-table-name	Configures the action to take on routing table entry 1.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action next-table da1</pre>	

	Command or Action	Purpose
Step 23	entry entry-id	Creates an entry in the routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # entry 2</pre>	
Step 24	match-adjacency key	Configures the match value for entry 1 against a source adjacency. In this case, the source adjacency is sip2.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-adjacency sip2</pre>	
Step 25	enum enum-id entry (enum) entry-name	Configures an ENUM entry for the routing table entry.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # enum 1 entry cisco-enum</pre>	
Step 26	dst-adjacency target-adjacency	Configures the destination adjacency for entry 2 in table routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip-proxy1</pre>	
Step 27	action complete	Configures the action to take on routing table entry 2. In this case, the action is complete.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action complete</pre>	
Step 28	rtg-dst-address-table table-id	Specifies the routing table (da1) that is searched for destination addresses to match called numbers.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # rtg-dst-address-table da1</pre>	
Step 29	entry entry-id	Creates an entry in the routing table.
	Example: Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1	
Step 30	match-address key	Configures the match value for entry 1 in the routing table, to match against a destination called number.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address bob</pre>	
Step 31	dst-adjacency target-adjacency	Configures the destination adjacency for entry 1 in table routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip-proxy2</pre>	

	Command or Action	Purpose
Step 32	action complete	Configures the action to take on routing table entry 1. In this case, the action is complete.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action complete</pre>	
Step 33	entry entry-id	Creates an entry in the routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # entry 2</pre>	
Step 34	match-address key	Configures the match value for entry 2 in the routing table, to match against a destination called number.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address kate</pre>	
Step 35	dst-adjacency target-adjacency	Configures the destination adjacency for entry 2 in table routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip-proxy3</pre>	
Step 36	action complete	Configures the action to take on routing table entry 2. In this case, the action is complete.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action complete</pre>	
Step 37	entry entry-id	Creates an entry in the routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # entry 3</pre>	
Step 38	match-address key	Configures the match value for entry 3 in the routing table, to match against a destination called number.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # match-address 44</pre>	
Step 39	prefix	Configures whether the match-address of this entry matches the start of the address.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # prefix</pre>	
Step 40	dst-adjacency target-adjacency	Configures the destination adjacency for entry 3 in table routing table.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # dst-adjacency sip-proxy4</pre>	

	Command or Action	Purpose
Step 41	action complete	Configures the action to take on routing table entry 3. In this case, the action is complete.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # action complete</pre>	
Step 42	complete	Completes the call-policy set after committing the full set.
	<pre>Example: Router(config-sbc-sbe-rtgpolicy-rtgtable-entry) # complete</pre>	
Step 43	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-rtgpolicy)# end	

Configuration Examples of ENUM Client Configuration

Example 1: ENUM Client

Use the following procedure to configure an ENUM Client:

```
Router# configure terminal
Router(config)# sbc MySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# enum 1
Router(config-sbc-sbe-enum)# req-timeout 10000
Router(config-sbc-sbe-enum)# max-recursive-depth 100
Router(config-sbc-sbe-enum)# entry ENUM_1
Router(config-sbc-sbe-enum-entry)# server ipv4 10.10.10.10 vrf VRF1
Router(config-sbc-sbe-enum-entry)# dial-plan-suffix Example.Suffix
Router(config-sbc-sbe-enum)# max-responses 100
Router(config-sbc-sbe-enum)# activate
Router(config-sbc-sbe-enum-entry)# end
```

Example 2: Call Policy for Multiple ENUM Entries

Use the following procedure to configure a call policy for multiple ENUM entries:

```
Router# configure terminal
Router(config)# sbc mysbc
Router(config-sbc)# sbe
Router(config-sbc-sbe)# enum 1
Router(config-sbc-sbe-enum)# entry default-enum
Router(config-sbc-sbe-enum-entry)# server ipv4 192.168.10.1
Router(config-sbc-sbe-enum-entry)# dial-plan-suffix e164.arpa
Router(config-sbc-sbe-enum-entry)# entry cisco-enum
Router(config-sbc-sbe-enum-entry)# server ipv4 10.0.0.22 vrf cisco-vrf
Router(config-sbc-sbe-enum-entry)# dial-plan-suffix cisco.com
Router(config-sbc-sbe-enum-entry)# dial-plan-suffix cisco.com
Router(config-sbc-sbe-enum-entry)# activate
Router(config-sbc-sbe-enum)# exit
```

```
Router(config-sbc-sbe)# sip header-profile enum
Router(config-sbc-sbe-sip-hdr) dst-address
Router(config-sbc-sbe-sip-hdr-dst)# header-prio 1 header-name Dst_Add_Hdr_1
```

```
Router(config-sbc-sbe-sip-hdr-dst)# header-prio 2 header-name Dst_Add_Hdr_2
Router(config-sbc-sbe-sip-hdr-dst)# exit
or
Router(config-sbc-sbe) # sip header-profile enum
Router(config-sbc-sbe-sip-hdr) src-address
Router (config-sbc-sip-hdr-src) # header-prio 1 header-name Src_Add_Hdr_1
Router(config-sbc-sbe-sip-hdr-src)# header-prio 2 header-name Src_Add_Hdr_2
Router(config-sbc-sbe-sip-hdr-src)# exit
or
Router(config-sbc-sbe)# sip header-profile enum
Router(config-sbc-sbe-sip-hdr) div-address
Router(config-sbc-sbe-sip-hdr-div)# header-prio 1 header-name Div_Add_Hdr_1
Router(config-sbc-sbe-sip-hdr-div)# header-prio 2 header-name Div_Add_Hdr_2
Router(config-sbc-sbe-sip-hdr-div)# exit
Router(config-sbc-sbe-sip-hdr)# call-policy-set 1
Router(config-sbc-sbe-rtgpolicy)# first-call-routing-table rt1
Router(config-sbc-sbe-rtgpolicy) # rtg-src-adjacency-table rt1
Router(config-sbc-sbe-rtgpolicy-rtgtable)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-adjacency sip2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# enum 1 entry default-enum
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action next-table da1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-adjacency sip2
Router (config-sbc-sbe-rtgpolicy-rtgtable-entry) # enum 1 entry cisco-enum
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency sip-proxy1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# rtg-dst-address-table da1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# entry 1
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address bob
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency sip-proxy2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# entry 2
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address kate
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency sip-proxy3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# entry 3
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# match-address 44
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# prefix
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# dst-adjacency sip-proxy4
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# action complete
Router(config-sbc-sbe-rtgpolicy-rtgtable-entry)# complete
Router(config-sbc-sbe-rtgpolicy) # end
Router#
```



IPv6 Support

Cisco Unified Border Element (SP Edition) supports IPv6 addressing on the unified model for SIP signaling and media. Cisco Unified Border Element (SP Edition) has the ability to handle IPv4 to IPv6 SIP signaling and media interworking, as well as IPv6 to IPv6 SIP signaling and media (RTP) interworking.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for IPv6 Support

Release	Modification
Cisco IOS XE Release 2.6	IPv6 Support features were introduced on the Cisco ASR 1000 Series Router.
Cisco IOS XE Release 3.1S	IPv6 Support for VRF was added on the Cisco ASR 1000 Series Router.

Contents

This module contains the following sections:

- Prerequisites, page 54-2
- Restrictions, page 54-2
- Information About IPv6 Support, page 54-2
- Configuring IPv6, page 54-5

Prerequisites

The following prerequisite is required to implement IPv6 Support:

Before implementing IPv6 Support, Cisco Unified Border Element (SP Edition) must already be configured.

Restrictions

The following are restrictions for IPv6 support on the Cisco Unified Border Element (SP Edition):

- H.323 over IPv6 is not supported.
- H.248 over IPv6 is not supported.
- The SBC does not support receiving and sending multiple IP addresses per media stream.

For more information, refer to RFC 4091, The Alternative Network Address Types (ANAT) Semantics for the Session Description Protocol (SDP) Grouping Framework and ICE (Interactive Connectivity Establishment).

- DNS look up over IPv6 is not supported.
- RADIUS (accounting and authentication) over IPv6 is not supported.

Information About IPv6 Support

In Cisco IOS XE Release 2.6, Cisco Unified Border Element (SP Edition) supports IPv6 addressing on the unified model for SIP signaling and media in the following ways:

- IPv4 to IPv6 SIP signaling interworking
- IPv4 to IPv6 media interworking
- IPv6 to IPv6 SIP signaling
- IPv6 to IPv6 media RTP interworking
- AAAA DNS query support

IPv6 to IPv6 RTP interworking on the media plane have been supported on the distributed model. The unified model now can enable IPv6 to IPv6 SIP signaling calls and IPv4 to IPv6 SIP signaling and media interworking calls.

The default behavior is that SBC assumes that the media address type to be used must match the signaling address type configured on the adjacency. Thus the default behavior which can be overridden by configuring a Call Admission Control (CAC) policy is for the media (RTP) to use the same version as used by signaling (SIP). The IP version used by SIP is dictated by the IP addresses configured on the adjacency. For example, if the incoming SIP INVITE comes in on an IPv4 adjacency and is routed out via an IPv6 adjacency, the incoming RTP will come over IPv4 and will be sent out over IPv4

IPv6 support for SIP calls affects the following SBC functions and existing unified SBC features:

- SIP URIs—IPv6 addresses are parsed in SIP URIs.
- Interworking IPv4 and IPv6 Adjacencies:
 - IPv6 addresses are passed through without modification in the Contact Username Passthrough feature.

- SBC supports IP/fully-qualified domain name (FQDN) entries for IPv6 adjacencies

Note

An adjacency can be configured for either IPv4 or IPv6 addresses only. Combinations of IPv4 and IPv6 addresses on the same adjacency are not supported.

- **Note** An adjacency configured for IPv4 cannot be changed to IPv6 and vice-versa, without deleting and recreating the adjacency. If the adjacency is referred to in the routing or CAC tables, these references must be removed before unconfiguring the adjacency.
- TLS over IPv6—Handles IPv6 addresses for adjacencies configured for SIP over Transport Layer Security (TLS) encryption.
- Access Authentication for SIP over IPv6

Supports IPv6 adjacencies for SIP inbound authentication to challenge inbound SIP requests.

For an incoming call over an IPv6 adjacency if the adjacency is configured for access authentication or inbound authentication, the call is challenged with a nonce (similar to what occurs in IPv4 addressing). The subsequent REGISTER message must have authentication parameters that should result in a RADIUS Access Request or an Access Accept (or reject) message. Note that the communication with the RADIUS server occurs over IPv4.

• Billing for IPv4 and IPv6 Calls

Packetcable billing records do not have IP addresses embedded in them. Therefore, billing for IPv6 calls work in the same manner as for IPv4 calls and no additional configuration is needed in billing and RADIUS configuration.

However only IPv4 addressing is supported for communicating with the RADIUS server. Both authentication and accounting requests go over IPv4, even when the requests are coming over an IPv6 adjacency. The control address used as source address in RADIUS requests is IPv4 only.

The billing manager local address goes in the NAS IP address field of RADIUS requests. This address is also an IPv4 address.

- SRTP Passthrough mode—No additional configuration changes for IPv6 addressing.
- Media bypass in Call Admission Control—The SBC must not attempt to perform media bypass between endpoints with different IP versions, even if media bypass CAC policy permits it.
- Blacklist Support—Supports the configuration of blacklist entries with IPv6 addresses or prefixes in the same way as IPv4 addresses.
- Logging—Displays IPv6 and IPv4 addresses.
- Late-to-Early Media Interworking—Supports calls terminating and originating from an IPv6 adjacency.
- Softswitch Shielding—Supports IPv6 endpoints registering with a softswitch.
- Call Hold—Supports IPv4 to IPv6 call interworking.

Using "c=0.0.0.0" as specified in RFC 2543 to indicate call hold is not valid with IPv6 addresses, and you must use "a=sendonly/inactive" to indicate a call hold.

 ToS/DSCP Marking for Signaling Messages—Supports DSCP marking for outgoing IPv4 and IPv6 signaling packets. • SIP Header Manipulation—Supports the passthrough header TO and FROM functionality for IPv6 to IPv4 interworked calls by passing the headername unchanged for incoming calls.

SBC rewrites the CONTACT header for outgoing calls.

- DTMF Interworking—Supports IPv6 adjacencies.
- IP Realms—Supports IPv6 adjacencies. IP addresses are assigned based on the realm configured on the IPv6 or IPv4 adjacency.
- SIP Instant Messaging—Supports IPv4 to IPv6 interworked calls.
- SIP IP-FQDN URI Translation—Supports IP-FQDN entries for IPv6 adjacencies.
- Domain Name Lookup (DNS)—Supports name lookup for IPv6 addresses and supports both A and AAAA DNS queries. DNS lookup happens over IPv4.
- Fast Registration—Supports IPv6 addresses.
- High Availability—IPv4 to IPv6 and IPv6 to IPv6 calls behave the same as IPv4 to IPv4 calls during failover.

Calls over UDP are replicated. For calls made over TCP, the signaling state is not replicated and will generate a TCP reset on receiving any SIP message after switchover.

• Media Hair-pinning—Supports IPv6 to IPv6 call hair pinning in the same manner as IPv4 to IPv4 calls. With media hair-pinning, calls come in and go back out on the same adjacency.



IPv4 to IPv6 hair-pinning is not supported because an adjacency can only be an IPv4 adjacency or an IPv6 adjacency.

3xx Redirect Messages—Supports redirection from IPv4 to IPv6 and from IPv6 to IPv4.

3xx represents a class of SIP response codes used in SIP to indicate that the sender of the request should try the request to an alternate URI or URIs that are presented in the 3xx response. Some of the widely-used response code examples are 301 "Moved Temporarily" or 302 "Moved Permanently."

Performing ISSU for IPv6 Calls

When performing ISSU to upgrade to a higher version Cisco IOS XE release, IPv4 to IPv4 calls migrate successfully to the higher version.

Before performing ISSU to migrate to a lower version release, you must first unconfigure all IPv6 adjacencies and remove all active IPv6 call states. You can clear calls through IPv6 adjacencies with the **no attach force abort** command. This command executes a forced detach, tearing down calls without signaling their end.

When performing ISSU to downgrade to a lower version Cisco IOS XE release, for example from Cisco IOS XE Release 2.6 to 2.5, if there is any IPv6 configuration or any active calls through an IPv6 adjacency, an error message is reported. If the user continues with the ISSU, the system will reach stateful switchover (SSO) without the SBC configuration being available on the standby processor. Before performing a downgrade, unconfigure all IPv6 configuration and dynamic state (for example, IPv6 to IPv6 and IPv6 to IPv4 calls, as well as IPv6 blacklists).

Configuring IPv6

To configure Cisco Unified Border Element (SP Edition) for IPv6 to IPv6 calls or IPv4 to IPv6 interworked calls, configure the local and remote addresses on the adjacency with IPv6 addresses.

If you have a peer or another SBC in your network that supports both IPv4 and IPv6 addresses, then you should define two adjacencies on the local SBC, one adjacency with IPv4 addresses and a second adjacency with IPv6 addresses.

Configuration Examples

The following example shows the asr1 SBC configured with IPv6 and IPv4 signaling and remote addresses on several SIP adjacencies and the 1 call policy set using a round-robin routing rule to implement call routing:



The **caller** and **callee** commands have been used in this procedure. In some scenarios, the **branch** command can be used as an alternative to the **caller** and **callee** command pair. The **branch** command has been introduced in Release 3.5.0. See the ?\$paranum>Configuring Directed Nonlimiting CAC Policies? section on page 7-37 for information about this command.

```
T
sbc asr1
sbe
  control address aaa ipv4 33.33.36.1
  radius authentication
  radius accounting server1
    server server1
    address ipv4 10.0.120.19
    kev cisco
   activate
   sip header-profile ccmpf1
   header Allow entry 1
    action pass
   header Call-Info entry 1
    action pass
   sip method-profile 1
   pass-body
   method MESSAGE
    action pass
   sip method-profile method1
   pass-body
   method INFO
    action pass
  sip method-profile ccmmethod1
   pass-body
   method SUBSCRIBER
    action pass
   sip method-profile ccmmethod2
   pass-body
   method INFO
    action pass
   method NOTIFY
    action pass
   method SUBSCRIBER
    action pass
   adjacency sip UEV6
```

L

```
group IPv6
 inherit profile preset-p-cscf-access
 visited network identifier open-ims.test
local-id host pcscf.open-ims.test
 signaling-address ipv6 2001:A401::33:33:36:1
 statistics method summary
 signaling-port 4060
 remote-address ipv6 2001::/64
 signaling-peer 2001::10:0:120:19
dbe-location-id 0
 attach
adjacency sip CCM134
 force-signaling-peer
group v4
nat force-on
header-profile inbound ccmpf1
header-profile outbound ccmpf1
method-profile inbound ccmmethod2
method-profile outbound ccmmethod2
 preferred-transport udp
 signaling-address ipv4 33.33.36.1
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.50.134 255.255.255.255
 signaling-peer 10.0.50.134
 dbe-location-id 0
 account CCM134
media-late-to-early-iw incoming
media-late-to-early-iw outgoing
 dtmf disable sip notify
 dtmf prefer sip info
 attach
adjacency sip CCM135
group v4
nat force-on
header-profile inbound ccmpf1
header-profile outbound ccmpf1
 preferred-transport udp
 signaling-address ipv4 33.33.36.1
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.50.135 255.255.255.255
 signaling-peer 10.0.50.135
 dbe-location-id 0
 attach
adjacency sip CCM136
 force-signaling-peer
 redirect-mode recurse
 signaling-address ipv4 33.33.36.1
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.50.136 255.255.255.255
 signaling-peer 10.0.50.136
 dbe-location-id 0
ping-enable
  ping-interval 60
  ping-lifetime 2
 attach
adjacency sip CSPS23
nat force-off
preferred-transport udp
 signaling-address ipv4 33.33.36.1
 statistics method summary
 remote-address ipv4 10.0.7.23 255.255.255.255
```

```
signaling-peer 10.0.7.23
 dbe-location-id 0
attach
adjacency sip OpensipsV6
group IPv6
nat force-off
 inherit profile preset-core
 signaling-address ipv6 2001:A401::33:33:36:1
 statistics method summary
 signaling-port 7060
 remote-address ipv6 2001::216:ECFF:FE3B:40DD/128
 signaling-peer opensips.cisco.com
 dbe-location-id 0
 registration target address opensips.cisco.com
header-name From passthrough
dtmf prefer sip info
 attach
adjacency sip CCM135-IPV6
 force-signaling-peer
 group v6
nat force-off
header-profile inbound ccmpf1
header-profile outbound ccmpf1
method-profile inbound ccmmethod2
method-profile outbound ccmmethod2
preferred-transport udp
 signaling-address ipv6 2001:A401::33:33:36:1
 statistics method summary
 signaling-port 5060
 remote-address ipv6 2001::10:0:50:135/128
 signaling-peer 2001::10:0:50:135
dbe-location-id 0
attach
adjacency sip CCM135-vrfb
vrf h323-vrf-b
nat force-off
preferred-transport udp
 signaling-address ipv4 10.190.7.97
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.50.135 255.255.255.255
 signaling-peer 10.0.50.135
 dbe-location-id 0
attach
adjacency sip CCM136-IPv6
 group v6
nat force-off
header-profile inbound ccmpf1
header-profile outbound ccmpf1
method-profile inbound ccmmethod2
method-profile outbound ccmmethod2
 signaling-address ipv6 2001:A401::33:33:36:1
 statistics method summary
 signaling-port 5060
 remote-address ipv6 2001::10:0:50:136/128
 signaling-peer 2001::10:0:50:136
 ping-enable
 ping-interval 60
 ping-lifetime 2
 dtmf prefer sip info
attach
adjacency sip SIPP81-IPv6
 group v6
nat force-off
```

```
preferred-transport udp
 signaling-address ipv6 2001:A401::33:33:36:1
 statistics method summary
signaling-port 5060
remote-address ipv6 2001::/64
 signaling-peer 2001::10:0:244:81
 dbe-location-id 0
 dtmf disable sip notify
dtmf prefer sip info
attach
call-policy-set 1
first-call-routing-table ROUTE-ON-DEST-NUM
first-reg-routing-table REG-ROUTE-ON-SRC-ADJ
rtg-src-adjacency-table REG-ROUTE-ON-SRC-ADJ
 entry 1
  action complete
  dst-adjacency OpensipsV6
  match-adjacency UEV6
 rtg-round-robin-table ROUND-ROBIN
  entry 1
  action complete
  dst-adjacency CCM136
 entry 2
  action complete
  dst-adjacency CCM136-IPv6
 rtg-dst-address-table ROUTE-ON-DEST-NUM
  entry 1
  action next-table ROUND-ROBIN
   edit del-prefix 3
  match-address 536X digits
  prefix
  entry 2
  action next-table ROUND-ROBIN
  edit del-prefix 4
  match-address 7898X digits
  prefix
  entry 3
  action next-table ROUND-ROBIN
  edit del-prefix 3
  match-address 491X digits
  prefix
  entry 4
  action next-table ROUND-ROBIN
  edit del-prefix 3
  match-address 526X digits
  prefix
  entry 5
   action next-table ROUND-ROBIN
  edit del-prefix 3
  match-address 496X digits
  prefix
  entry 6
  action complete
  edit del-prefix 3
  dst-adjacency CCM135
  match-address 4553X digits
  prefix
  entry 7
  action complete
  edit del-prefix 3
  dst-adjacency CCM135
  match-address 789X digits
  prefix
  entry 8
```

action complete

```
edit del-prefix 4
     dst-adjacency CCM135
     match-address 5678X digits
     prefix
    entry 9
     action complete
     edit del-prefix 4
     dst-adjacency CCM135
     match-address 5677X digits
     prefix
    entry 10
     action complete
     edit del-prefix 3
     dst-adjacency CCM135
     match-address 516X digits
     prefix
   complete
  active-call-policy-set 1
   sip dns
   support-type sip-dns-naptr
  sip ip-fqdn-mapping 1 ipv6 2001::10:0:50:137 ccm137.cisco.com ip-to-fqdn
1
  billing
   local-address ipv4 33.33.36.1
   ldr-check 0 0
   method packetcable-em
   cache path harddisk:/cdr/
   retry interval 20
   cdr media-info
   packetcable-em 1 transport radius server1
    local-address ipv4 33.33.36.1
   activate
  blacklist global ipv6 2002::10:0:0:1
   reason corrupt-message
    trigger-size 65535
    trigger-period 1 minutes
  blacklist critical global ipv6 2003::10:0:0:1
   reason authentication-failure
    trigger-size 65535
    trigger-period 1 minutes
   subscriber sip:bob@isp.example.com
    sip-contact 2001::10:1:1:2
     adjacency UEV6
    delegate-registration sip:reg@isp.example.com
     adjacency OpensipsV6
     header-name supported add path
    activate
1
media-address ipv4 33.33.36.2
media-address ipv6 2001:A401::33:33:36:2
media-timeout 360
activate
T
```

The following example shows different signaling and media addresses configured on the SBC asr1, where the signaling address configured is ipv6 and the media address configured is ipv4:

```
sbc asr1
sbe
   adjacency sip CCM1-IPV6
   group media-v4
   nat force-off
   preferred-transport udp
    signaling-address ipv6 2001:A401::33:33:36:1
    statistics method summary
    signaling-port 5060
    remote-address ipv6 2001::10:0:56:186/128
    signaling-peer 2001::10:0:56:186
   dbe-location-id 0
   attach
   adjacency sip CCM2-IPV6
    group media-v4
   nat force-off
   preferred-transport udp
    signaling-address ipv6 2001:A401::33:33:36:1
    statistics method summary
    signaling-port 5060
    remote-address ipv6 2009::100:0:0:4/128
    signaling-peer 2009::100:0:0:4
    dbe-location-id 0
   attach
   cac-policy-set 1
    first-cac-table table1
    cac-table table1
    table-type limit account
     entry 1
     match-value media-v4
     action cac-complete
      caller media-type ipv4
      callee media-type ipv4
    complete
   active-cac-policy-set 1
   call-policy-set 1
    first-call-routing-table table1
    rtg-dst-address-table table1
     entry 1
      action complete
      edit del-prefix 3
      dst-adjacency CCM2-IPV6
      match-address 123X digits
      prefix
   complete
   active-call-policy-set 1
 !
 !
 1
media-address ipv4 33.33.36.10
media-timeout 360
activate
!
!
```

IPv6 Configuration Commands

This section describes the configuration commands used to configure various types of IPv6 addressing or show output listing IPv6 addresses.

For details of the following commands, see the Cisco Unified Border Element (SP Edition) Command Reference: Unified Model at

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html

Table 54-1 lists the new commands introduced in Cisco IOS XE Release 2.6.

Table 54-1 New Commands Introduced in Cisco IOS XE Release 2.6

Command	Description
callee media-type {ipv4 ipv6 inherit both}	To configure the media address type settings for a callee on the Cisco Unified Border Element (SP Edition).
	Use the no form of this command to disable the media address type settings for a callee.
caller media-type {ipv4 ipv6 inherit both}	To configure the media address type settings for a caller on the Cisco Unified Border Element (SP Edition).
	Use the no form of this command to disable the media address type settings for a caller.

Table 54-2 lists the commands modified for IPv6 addressing in Cisco IOS XE Release 2.6.

Table 54-2	Commands Modified for IPv6 Addressing in Cisco IOS XE Release 2.6
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Command	Description
blacklist [critical] global [address-default	The ipv6 keyword was added.
{ $ipv4$ { $addr$ } $ipv6$ { $addr$ } } [tcp { $tcp-port$ } udp { $udp-port$ } default-port-limit]]	Use this command to enter the mode for configuring the default event limits for the IPv6 address.
	For IPv6, only global option can be used.
clear sbc <i>sbc-name</i> sbe blacklist [critical]	The ipv6 keyword was added.
{ $ipv4 addr ipv6 addr$ } [{ $udp tcp$ } port]	To clear the blacklist for the specified Session Border Controller (SBC) service.
remote-address {ipv4 ip-address ip-mask ipv6	The ipv6 keyword was added.
ip-address / prefix-length}	To configure either an H.323 or SIP adjacency to restrict the set of remote signaling peers that can be contacted over the adjacency to those with a given IP address prefix.
	To remove this configuration, use the no form of this command.
show sbc sbc-name sbe addresses	The output of this command was modified.
<pre>show sbc sbc-name sbe adjacencies {adjacency-name} [detail]</pre>	The output of this command was modified.

Command	Description
<pre>show sbc sbc-name sbe blacklist critical { ipv4</pre>	The ipv6 keyword was added.
addr ipv6 addr] [tcp tcp-port udp udp-port]	The command was updated to show all configured critical blacklists for IPv6 addresses.
<pre>show sbc sbc-name sbe blacklist [source] { ipv4 IP address ipv6 IP address }</pre>	The ipv6 keyword was added.
<pre>show sbc name sbe cac-policy-set [id [table name [entry id]] active [table name [entry id]]] [detail]</pre>	The output of this command was modified.
show sbc sbc-name sbe calls	To provide details of IPv6 calls.
<pre>show sbc sbc-name sbe call-stats {all global src-sccount name dst-account name src-adjacency name dst-adjacency name} period</pre>	To list the number of active IPv6 calls.
show sbc sbc-name sbe addresses	The output of this command was modified.
show sbc <i>sbc-name</i> sbe sip ip-fqdn-mapping	Displays the IP-FQDN mapping table.
	The output of this command was modified to include IPv6 details.
<pre>signaling-address {ipv4 ipv4_IP_address ipv6 ipv6_IP_address}</pre>	To define the local signaling address of an H.323 (IPv4 only) or SIP adjacency.
	The ipv6 keyword was added.
	To return to the default behavior, use the no form of this command.
<pre>sip ip-fqdn-mapping index { ipv4 ipv6 } ip-address fqdn {both-ways ip-to-fqdn}</pre>	To configure SIP IP-to-FQDN mapping on signaling border elements (SBEs).
	The ipv6 keyword was added.

Table 54-2 Commands Modified for IPv6 Addressing in Cisco IOS XE Release 2.6

Table 54-3 lists the command modified for IPv6 addressing in Cisco IOS XE Release 3.1.0S.

Table 54-3Command Modified for IPv6 Addressing in Cisco IOS XE Release 3.1.0S

Description
The ipv6 keyword was added. Use this command to enter the mode for configuring the default event limits for the IPv6 address in a VPN.
Table 54-4 lists the command modified for IPv6 addressing in Cisco IOS XE Release 3.5.0S.

Command	Description
branch media-type {ipv4 ipv6 inherit both}	Configures the media address type settings for a caller or callee on the Cisco Unified Border Element (SP Edition). Use the no form of this command to disable the media address type settings for the caller or callee.

 Table 54-4
 Command Modified for IPv6 Addressing in Cisco IOS XE Release 3.5.0S

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Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model



P-CSCF Support

The Proxy-Call Session Control Function (P-CSCF) is the first contact point for the users of the IP Multimedia Subsystem (IMS). The P-CSCF functions as a proxy server for the user equipment; all Session Initiation Protocol (SIP) signaling traffic to and from the user equipment must go through the P-CSCF. The P-CSCF validates and then forwards requests from the user equipment and then processes and forwards the responses to the user equipment.

The P-CSCF can also function as a user agent in the context of the SIP operating procedures. If an abnormal condition arises in a session, the P-CSCF can unilaterally release the session for the user equipment. The user agent role can also be used to generate independent SIP messages required during the registration, such as sending the user's public and private identities. There may be more than one P-CSCF in the operator's network based on survivability, number of users, expected traffic, and network topology. The P-CSCF can be also referred to as the SIP server.

To implement the P-CSCF support on Cisco Unified Border Element (SP Edition), users must select an Inherit Profile for a SIP adjacency. The three available Inherit Profiles are:

- Standard Non-IMS Profile
- P-CSCF Access Profile
- P-CSCF Core Profile

Each of these profiles groups a set of IMS-related configuration fields that can be applied across multiple adjacencies.

If a valid profile is configured, this profile is applied to an adjacency that does not have a profile configured. If a profile is already selected for a SIP adjacency, that profile is used instead of the entity's profile.

In Cisco IOS XE Release 2.5 and later, Cisco Unified Border Element (SP Edition) supports Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA) for SIP calls. This type of authentication is used for access authentication in mobile IMS deployments and typically may reside on a mobile subscriber's card inside a phone. No special configuration is needed. The only requirement is that a UNI SIP profile is configured on the access side of the network.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XR Software Release and later, this feature is supported in the unified model only.

Feature History for P-CSCF Support

Release	Modification
Cisco IOS XR Software Release	This support was introduced on the Cisco IOS XR along with support for the unified model.
Cisco IOS XE Release 2.5	The HTTP Digest Authentication Using AKA feature was introduced on the Cisco ASR 1000 Series Routers.

Contents

This module contains the following sections:

- Restrictions for Implementing P-CSCF Support, page 55-2
- Information About P-CSCF Support, page 55-2
- Implementing P-CSCF Support, page 55-6
- Information About HTTP Digest Authentication Using AKA, page 55-7

Restrictions for Implementing P-CSCF Support

The following restrictions and limitations apply to implementing P-CSCF support:

- Since the Visited Network Identifier is not part of an Inherit Profile, you need to configure it independently on a per-adjacency basis.
- This feature does not offer support for securing access links through IPsec or Network Attachment Subsystem (NASS) bundled authentication.
- This feature does not support emergency calls.

Information About P-CSCF Support

This section contains the following subsections:

- Standard Non-IMS Profile, page 55-3
- P-CSCF Access Profile, page 55-3
- P-CSCF Core Profile, page 55-3
- Effect of P-CSCF Inherit Profiles on Method Profiles, Header Profiles, and Option Profiles, page 55-4

Standard Non-IMS Profile

This profile provides compatibility with existing Cisco Unified Border Element (SP Edition) functionality and is used for adjacencies that do not operate in an IMS network. When this profile is applied to an adjacency, Cisco Unified Border Element (SP Edition) exhibits the following properties:

- Contact headers are rewritten to ensure that the SBC remains on the signaling path.
- Unknown headers, methods, and options are, by default, not allowed to pass through.
- Cisco Unified Border Element (SP Edition) does not attach Path headers to outbound signals.
- Cisco Unified Border Element (SP Edition) does not attach Record-Route headers to outbound signals.
- The endpoints on this adjacency do not need to be registered to receive or send Non-REGISTER requests.
- The endpoints do not need to attach a Route header to outbound signals.
- The adjacencies do not generate P-Charging Vector headers for outbound signals.

P-CSCF Access Profile

This profile provides the configurations required to perform the functions of a P-CSCF Access adjacency. When this profile is applied to an adjacency, Cisco Unified Border Element (SP Edition) exhibits the following properties:

- Contact headers are not rewritten.
- The endpoints on this adjacency need to be registered to receive or send Non-REGISTER requests.
- The endpoints need to attach a Route header to outbound signals, which in turn, matches a Service-Route set from the Registrar.
- The SBC appends Record-Route headers to outbound signals for adjacencies with P-CSCF profiles.
- The SBC does not attach Path headers to outbound signals.
- The adjacencies do not generate P-Charging Vector headers for outbound signals.
- The SBC, by default, allows all inbound non-essential headers to pass through, except P-Asserted Identity, Security-Client, Security-Verify, P-Charging-Function Addresses, P-Charging-Vector, and P-Media-Authorization.
- The SBC, by default, allows all outbound non-essential headers, except P-Charging-Function-Addresses, P-Charging-Vector, and P-Media-Authorization.
- The SBC allows all inbound non-essential methods to pass through.
- The SBC allows all outbound non-essential methods to pass through; UEs are not permitted to act as Registrars.
- The Option tags in Supported, Require, or Proxy-Require headers are allowed to pass through in both directions.

P-CSCF Core Profile

This profile provides the configurations required to perform the functions of a P-CSCF Core adjacency. When this profile is applied to an adjacency, Cisco Unified Border Element (SP Edition) exhibits the following properties:

- Contact headers are not rewritten.
- The SBC, by default, allows all inbound unknown headers, except the P-Charging-Function-Addresses and P-Media-Authorization.
- The SBC appends Record-Route headers to outbound signals for adjacencies with P-CSCF profiles.
- The SBC attaches Path headers to outbound REGISTER signals from P-CSCF.
- The adjacencies generate P-Charging Vector headers for outbound signals.
- The endpoints on this adjacency do not need to be registered to receive or send Non-REGISTER requests.
- The SBC, by default, allows all outbound non-essential headers, except P-Charging-Function-Addresses and P-Media-Authorization.
- The SBC allows all unknown methods to pass through.
- The Option tags in Supported, Require, or Proxy-Require headers are allowed to pass through in both directions.

Effect of P-CSCF Inherit Profiles on Method Profiles, Header Profiles, and Option Profiles

Use of a P-CSCF inherit profile dynamically assigns the following sets of profiles (method profile, header profile, and option profile) to a call based on the P-CSCF inherit profile selected. Table 55-1 shows which P-CSCF inherit profile has an effect on which specific method profile, header profile, and option profile.

The effect is not visible in the adjacency configuration for header-profile, method-profile or option profiles, and can be overridden by explicit configuration of header, method, option profiles as needed.

P-CSCF Inherit Profile	Method Profile	Header Profile	Option Profile
preset-p-cscf-access	preset-acc-in-mth	preset-acc-in-hdr	preset-acc-in-opt
	Type: Blacklist	Type: Blacklist	preset-acc-out-opt
	Actions: No methods rejected preset-acc-out-mth	Actions: Removes Security-Client Removes Security-Verify Removes P-Charging-Vector Removes P-Asserted-Identity	Type: Blacklist Actions: No options (Passes on all)
	Actions: Rejects REGISTER	Removes P-Visited-Network-ID Removes P-Media-Authorization Removes P-Charging-Function-Address	
		preset-acc-out-hdr	
		Type: Blacklist	
		Actions: Removes P-Charging-Vector Removes P-Media-Authorization	
preset-p-cscf-core	preset-core-in-mth	preset-core-in-hdr	preset-core-in-opt
	Type: Blacklist	preset-core-out-hdr	preset-core-out-opt
	Actions: No methods removed preset-core-out-mth	Type: Blacklist Actions: Removes no headers (passes all)	Type: Blacklist Actions: No options (Passes
	Type: Blacklist		on all)
	Actions: No methods rejected		
preset-standard-non-	preset-std-in-mth	preset-std-in-hdr	preset-std-in-opt
ims	preset-std-out-mth	preset-std-out-hdr	preset-std-out-opt
	Type: Whitelist	Type: Whitelist	Type: Whitelist
	Actions: Passes INFO Passes UPDATE	Actions: Passes Server Passes Diversion Passes Resource-Priority	Actions: Passes Replaces (only)

	Table 55-1	Effect of P-CSCF Inherit Profiles on Method, Header and Option Prof	files
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Implementing P-CSCF Support

This section explains how to configure intrinsic profiles and profile inheritance.

Configuring Profile Inheritance

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. sip inherit profile preset-p-cscf-access
- 5. adjacency sip adjacency-name
- 6. inherit profile preset-p-cscf-access
- 7. visited network identifier network-name
- 8. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of a SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	sip inherit profile preset-p-cscf-access	Configures the P-CSCF Access Inherit Profile as the global profile. For a list of other configurable parameters, see the sip inherit profile command.
	Example: Router(config-sbc-sbe)# sip inherit profile preset-p-cscf-access	
Step 5	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipadj	• Use the <i>adjacency-name</i> argument to define the name of the SIP adjacency.

	Command or Action	Purpose
Step 6	inherit profile preset-p-cscf-access	Configures the SIP adjacency to use the P-CSCF-Access profile.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-p-cscf-access</pre>	
Step 7	visited network identifier network-name	Configures the specified visited network identifier on the SIP adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# visited network identifier mynetwork.com	
Step 8	exit	Exits the SIP adjacency mode to the SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	

Information About HTTP Digest Authentication Using AKA

This section contains the following subsections:

- Configuring HTTP Digest Authentication Using AKA, page 55-8
- Configuration Example—HTTP Digest Authentication Using AKA, page 55-10

Cisco Unified Border Element (SP Edition) supports Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA) for SIP calls. This type of authentication is used for access authentication in mobile IMS deployments and typically resides on a mobile subscriber's card inside a phone. Cisco Unified Border Element (SP Edition) supports the HTTP Digest Authentication Using AKA feature with no special configuration needed, as long as a User-to-Network Interconnections (UNI) SIP profile is configured on the access side (that is, with a P-CSCF access side profile).

The AKA function carries out user authentication and session key distribution in Universal Mobile Telecommunications System (UMTS) networks. AKA is challenge- response based. The response to the challenge is computed by the application running on the mobile subscriber's card inside the phone.

HTTP Digest Authentication is common with IP-PBXs. The HTTP Digest Authentication procedure is used to ensure that only valid devices can register (at a SIP level) to a network. The SBC supports the typical registration call flow, that is, passing through authentication challenges and their responses. A typical call flow consists of a SIP REGISTER message from an endpoint that is routed by the SBC to the SIP registrar. The registrar replies with a 401 Unauthorized response and a "challenge."

The challenge contains a random number that the endpoint uses to compute a response, which is sent in another REGISTER message. Finally the registrar replies with a 200 OK message if the response was valid. In the case of HTTP Digest Authentication Using AKA, the response to the challenge is computed by the application running on the mobile subscriber's card inside the phone. The SBC supports this typical call flow by means of enabling a SIP profile that allows SIP registrations.

Another usage of HTTP Digest Authentication Using AKA concerns the ability of using the procedure to establish an IPsec connection (actually two IPsec connections) for ensuring signaling security. Cisco Unified Border Element (SP Edition) supports IPsec, however the ability to extract the port security association identifiers and key information from SIP messages is not supported in Cisco IOS XE Release 2.5.

Configuring HTTP Digest Authentication Using AKA

This task configures HTTP Digest Authentication Using AKA on two related adjacencies where preset-access and preset-core profiles must be configured.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency {sip | h323} adjacency-name
- 5. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}
- 6. exit
- 7. adjacency {sip | h323} adjacency-name
- 8. inherit profile {preset-access | preset-core | preset-ibcf-ext-untrusted | preset-ibcf-external | preset-ibcf-internal | preset-p-cscf-access | preset-p-cscf-core | preset-peering | preset-standard-non-ims}
- 9. exit
- 10. end
- 11. show sbc sbc-name sbe adjacencies adjacency-name detail

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the SBC and enters into SBC configuration mode.
	Example: Router(config)# sbc mySbc	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the SIP adjacency facing the endpoint, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip sipEndpoint	

	Command or Action	Purpose
Step 5	<pre>inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal </pre>	Required. Configures a preset P-CSCF access profile for the SIP adjacency facing the endpoint.
	preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}	P-CSCF is Proxy-Call Session Control Function—part of its function is to authenticate the user and establish an IPsec security association with the IMS terminal.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-p-cscf-access</pre>	
Step 6	exit	Exits adjacency sip configuration mode and enters into sbe configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 7	<pre>adjacency {sip h323} adjacency-name</pre>	Configures the SIP adjacency facing the registrar/softswitch, and enters into adjacency sip configuration mode.
	Example: Router(config-sbc-sbe)# adjacency sip SoftSwitch	
Step 8	inherit profile {preset-access preset-core preset-ibcf-ext-untrusted preset-ibcf-external preset-ibcf-internal	Required. Configures a preset P-CSCF core profile for the SIP adjacency facing the registrar/softswitch.
	<pre>preset-p-cscf-access preset-p-cscf-core preset-peering preset-standard-non-ims}</pre>	An adjacency facing the registrar typically has a preset-core profile. The default is preset-core.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# inherit profile preset-p-cscf-core</pre>	
Step 9	exit	Exits adjacency sip configuration mode and enters into SBE configuration mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 10	end	Exits SBE configuration mode and returns to EXEC mode.
	Example: Router(config-sbc-sbe)# end	
Step 11	<pre>show sbc sbc-name sbe adjacencies adjacency-name detail</pre>	Displays all the detailed field output for the specified SIP adjacency.
	Example: Router# show sbc sbe mySBC sbe adjacencies SoftSwitch detail	

Configuration Example—HTTP Digest Authentication Using AKA

The following is a configuration example used to verify HTTP Digest Authentication Using AKA:

```
sbc asr
sbe
   adjacency sip UE
   inherit profile preset-p-cscf-access
   visited network identifier open-ims.test
   local-id host pcscf.open-ims.test
    signaling-address ipv4 10.190.5.129
   signaling-port 4060
   remote-address ipv4 10.0.0.0 255.255.0.0
   signaling-peer 10.0.120.19
   dbe-location-id 100
    fast-register disable
   attach
   adjacency sip OpenIMSCore
    inherit profile preset-p-cscf-core
    visited network identifier open-ims.test
   local-id host pcscf.open-ims.test
   signaling-address ipv4 10.190.5.129
   signaling-port 4060
   remote-address ipv4 10.0.48.236 255.255.255.255
    signaling-peer 10.0.48.236
    dbe-location-id 100
    registration rewrite-register
    registration target address open-ims.test
    attach
```



IBCF Processing Support

Users can configure Cisco Unified Border Element (SP Edition) to perform the role of an Interconnection Border Control Function (IBCF) Session Initiation Protocol (SIP) border gateway, both managing requests across a network border between IP Multimedia Subsystem (IMS) core networks and interworking with non-IMS core networks.

When functioning as an IBCF, Cisco Unified Border Element (SP Edition) supports the following IBCF functions:

- Adding to Path header on REGISTER
- Modifying Service Route header
- Routing based on SIP Route headers
- Topology hiding
- Screening of SIP signaling
- IBCF inherit profiles
- Passthrough of From, To, and Contact headers
- Passthrough of request Uniform Resource Identifier (URI) on REGISTER
- Interworking with Proxy Call Session Control Function (P-CSCF), Interrogating Call Session Control Function (I-CSCF, and Serving Call Session Control Function (S-CSCF)
- · Handling messages from untrusted domains
- Adding Record-Route headers on outbound messages for adjacencies with IBCF profiles.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the session border controller (SBC).

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.



For Cisco IOS XE Release 2.4, this feature is supported in the unified model only.

Feature History for IBCF Support

Release	Modification
Cisco IOS XE Release 2.4	This feature was introduced on the Cisco CRS-1 along with support for the unified model.

Contents

This module contains the following sections:

- Restrictions for Implementing IBCF Support, page 56-2
- Information About IBCF Support, page 56-2
- Implementing IBCF Support, page 56-5

Restrictions for Implementing IBCF Support

The following features are not included in the IBCF support:

- Blacklist or whitelist header-values-content-type, content-disposition, and content-language headers
- Blacklist or whitelist MIME bodies
- Session timer
- Co-location with I-CSCF
- Cisco Unified Border Element (SP Edition) does not reject long message bodies.
- Cisco Unified Border Element (SP Edition) does not check the length of SIP bodies.
- Cisco Unified Border Element (SP Edition) does not provide a default implementation of the Encryption User Exit.
- Cisco Unified Border Element (SP Edition) does not hide network devices that are identified by IP addresses.
- Cisco Unified Border Element (SP Edition) does not support the full IBCF handling of failed REGISTERs.
- Cisco Unified Border Element (SP Edition) does not provide interoperability between IMS and other SIP domains.
- The IBCF selection of a new entry point for forwarding REGISTER requests is limited to SIP Server Location procedures (as per IETF RFC 3263) and is applicable only if the initial server selected does not respond.

Information About IBCF Support

This section contains the following subsections:

- Adding to Path Header on REGISTER, page 56-3
- Modifying Service-Route Header on REGISTER, page 56-3
- Routing Based on SIP Route Headers, page 56-3

- Topology Hiding, page 56-3
- Screening of SIP Signaling, page 56-3
- IBCF Inherit Profiles, page 56-3
- Passthrough of From, To, and Contact Headers, page 56-5
- Passthrough of Request URI on REGISTER, page 56-5
- Interworking with P-CSCF, I-CSCF, and S-CSCF, page 56-5
- Handling Messages from Untrusted Domains, page 56-5

Adding to Path Header on REGISTER

When Cisco Unified Border Element (SP Edition) is configured to perform the role of an IBCF gateway, the IBCF adds itself to the Path header to ensure that all INVITE requests to the subscriber are routed via the IBCF.

Modifying Service-Route Header on REGISTER

The Service-Route header is analogous to the Path header, but it is used to specify the list of devices a call should traverse for calls originating from a subscriber. By default, the IBCF does not modify the Service-Route header sent on REGISTER responses. However, if topology hiding is required, then the IBCF encrypts the header elements that match its configured HomeNetworkId.

Routing Based on SIP Route Headers

You can configure Cisco Unified Border Element (SP Edition) to route Dialog-creating requests, such as INVITE, to the next hop-IP address based on the Route header, which ensures that the SIP messages go through the specified border gateways between networks and the S-CSCF that handled the User Agent (UA) REGISTER.

Topology Hiding

Cisco Unified Border Element (SP Edition) hides those parts of the routing-related headers that reveal the internal topology of the SBC network. But this feature also ensures that the headers are usable for INVITE requests and other methods.

Screening of SIP Signaling

When configured to perform the role of an IBCF gateway, Cisco Unified Border Element (SP Edition) does not screen certain SIP headers using profile whitelists and blacklists.

IBCF Inherit Profiles

IBCF inherit profiles comprise a collection of related configuration appropriate to a particular network role. IBCF Inherit profiles may be configured for an application on a per-adjacency basis.

Cisco Unified Border Element (SP Edition) supports the following IBCF inherit profiles:

- preset-ibcf-ext-untrusted
- preset-ibcf-external
- preset-ibcf-internal

Use of an IBCF inherit profile dynamically assigns a method profile, header profile, and/or option profile to a call based on the inherit-profile selected. Table 56-1 shows which IBCF inherit profile has an effect on which specific method profile, header profile, and option profile.

The effect is not visible in the adjacency configuration for header-profile, method-profile or option profiles, and can be overridden by explicit configuration of header, method, option profiles as needed.

IBCF Inherit Profile	Method Profile	Header Profile	Option Profile
preset-ibcf-ext-untrust	preset-ibcf-utr-in-mth	preset-ibcf-utr-in-hdr	preset-ibcf-utr-in-opt
ed	preset-ibcf-utr-out-mth	Type: Blacklist	preset-ibcf-utr-out-opt
	Type: Blacklist	Actions:	Type: Blacklist
	Actions: No methods rejected	Removes P-Charging-Vector. Removes P-Asserted-Identity. Removes P-Access-Network-Info. Removes P-Charging-Function-Addresses	Actions: No options (passes on all)
		preset-ibcf-utr-out-hdr	
		Type: Blacklist	
		Actions: Removes P-Charging-Function-Addresses	
preset-ibcf-external	preset-ibcf-ext-in-mth	preset-ibcf-ext-in-hdr	preset-ibcf-ext-in-opt
	preset-ibcf-ext-out-mth	Type: Blacklist	preset-ibcf-ext-out-opt
	Type: Blacklist	Actions:	Type: Blacklist
	Actions: No methods rejected	Removes no headers (passes all) preset-ibcf-ext-out-hdr	Actions: No options (passes on all)
		Type: Blacklist	
		Actions: Removes P-Charging-Vector. Removes P-Charging-Function-Addresses	
preset-ibcf-internal	preset-ibcf-int-in-mth	preset-ibcf-int-in-hdr	preset-ibcf-int-in-opt
	preset-ibcf-int-out-mth	preset-ibcf-int-out-hdr	preset-ibcf-int-out-opt
	Type: Blacklist	Type: Blacklist	Type: Blacklist
	Actions: No methods rejected	Actions: Removes no headers (passes all)	Actions: No options (passes on all)

Table 56-1 Effect of IBCF Inherit Profiles on Method, Header and Option Profiles

Passthrough of From, To, and Contact Headers

For Dialog-creating and Out-of-dialog requests, Cisco Unified Border Element (SP Edition) allows the From, To, and Contact header URIs to pass through without modifying them. For dialog headers, Cisco Unified Border Element (SP Edition) uses the values corresponding to those on the Out-of-dialog requests.

Passthrough of Request URI on REGISTER

Cisco Unified Border Element (SP Edition) allows the Request URI on a REGISTER message to pass through without modifying it.

Interworking with P-CSCF, I-CSCF, and S-CSCF

When performing the role of an IBCF gateway, Cisco Unified Border Element (SP Edition) allows the CSCF-specific headers on SIP messages to pass through.

Handling Messages from Untrusted Domains

When Cisco Unified Border Element (SP Edition) is acting as an IBCF entry point, it handles out-of-dialog requests from untrusted domains as follows:

- Cisco Unified Border Element (SP Edition) rejects all REGISTER requests with a 403 response.
- Cisco Unified Border Element (SP Edition) removes all P-Asserted-Identity headers, P-Access-Network-Info headers, P-Charging-Vector headers, and P-Charging-Function-Address headers from other requests.
- Cisco Unified Border Element (SP Edition) rejects requests if the router contains the Orig parameter.

Implementing IBCF Support

Configuring the Domain Names to Use for IBCF Adjacencies

SUMMARY STEPS

- 1. configure terminal
- 2. sbc service-name
- 3. sbe
- 4. sip home network identifier domain-name
- 5. sip encryption key string
- 6. adjacency sip adjacency-name
- 7. inherit profile preset-ibcf-internal
- 8. home network identifier *domain-name*

9. encryption key string

10. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc service-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>service-name</i> argument to define the name of the service.
Step 3	sbe	Enters the mode of a SBE entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	sip home network identifier domain-name	Configures the specified domain name as the global home network identifier for use in all SIP IBCF adjacencies.
	Example: Router(config-sbc-sbe)# sip home network identifier mydomain.com	Use the <i>domain-name</i> argument to specify the domain name of the SIP adjacency.
Step 5	sip encryption key string	Configures a global encryption key for all SIP IBCF adjacencies.
	Example: Router(config-sbc-sbe)# encryption key code1	• Use the <i>string</i> value to specify the encryption key to use for all SIP IBCF adjacencies.
Step 6	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe)# adjacency sip sipadj	• Use the <i>adjacency-name</i> argument to define the name of the SIP adjacency.
Step 7	inherit profile preset-ibcf-internal	Configures a global inherit profile and specifies a preset IBCF internal profile.
	Example: Router(config-sbe-adj-sip)# inherit profile preset-ibcf-internal	
Step 8	home network identifier network-name	Configures a home network identifier on an IBCF adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# home network identifier Cisco.com	Use the <i>network-name</i> argument to specify the name of the home network identifier.

	Command or Action	Purpose
Step 9	encryption key string	Configures an encryption key on the SIP IBCF adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# encryption key code2	• Use the <i>string</i> argument to specify the encryption key for the SIP IBCF adjacency.
Step 10	exit	Exits the SIP adjacency mode to the SBE mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	





IMS Rx, Diameter, and IMS Rf

The Cisco Unified Border Element (SP Edition) supports IP Multimedia Subsystem (IMS) Rx interfaces, Diameter protocol, and IMS Rf interfaces.

An IMS Rx is a Third Generation Partnership Project (3GPP) interface that runs between an application function and a Policy Charging and Rules Function (PCRF) in a 3GPP architecture.

The Diameter is an Authentication Authorization Accounting (AAA) protocol and is an enhanced version of the RADIUS (Remote Authentication Dial-In User Service) protocol.

An IMS Rf is an interface that runs between Charging Trigger Function (CTF) and Charging Data Function (CDF) in a 3GPP architecture.

Feature History for IMS Rx, Diameter, and IMS Rf

Release	Modification
Cisco IOS XE Release 3.1S	• The IMS Rx Interfaces feature was introduced.
	• The Diameter feature was introduced.
Cisco IOS XE Release 3.7S	The IMS Rf Billing Interface feature was introduced.

Contents

- Information About IMS Rx Interfaces, page 57-2
- Configuring IMS Rx, page 57-3
- Configuration Examples for IMS Rx, page 57-7
- Information About the Diameter Protocol in the SBC, page 57-8
- Configuring SBC Diameter Routing, page 57-9
- Configuration Examples for Diameter Routing, page 57-14
- Information About IMS Rf Billing Interfaces, page 57-16
- Configuring an IMS Rf Billing Interface, page 57-17
- Configuration Example for IMS Rf Billing Interface, page 57-20

Information About IMS Rx Interfaces

An IMS Rx interface is a 3GPP interface that runs between an application function and a Policy Charging and Rules Function (PCRF) in a 3GPP architecture. In this case, SBC is the application function.

SBC uses the Rx interface to communicate with the PCRF during call initiation and renegotiation to ensure that a call conforms to policy. SBC uses the Rx interface during registration to learn access network information.

The PCRF performs the following functions for SBC via an IMS Rx interface:

- Confirms that call media requests conform to the appropriate policy.
- Opens gates or pinholes in the media route, and specifies the appropriate QoS.
- Requests per-flow charging information when needed.
- Informs SBC of media-plane events.

An IMS Rx interface can be configured as a pure Rx environment or as a mixed Rx and media resource environment in unified SBC.

Features Supported

SBC can be deployed as the application function connecting to a PCRF via an Rx interface, in a mobile network, under an IMS or non-IMS environment. SBC supports the following requirements for these environments:

- Support for precondition call flows with Rx
- Support for late-INVITE and PRACK with Rx
- SIP late and early interworking in combination with Rx
- SIP PRACK and non-PRACK interworking in combination with Rx
- Support for session binding on registration
- SBC does not add any IMS-specific SIP headers to requests or responses in non-IMS environment, and does not add P-Charging-Vector or P-Access-Network-Info information
- SBC can also use an Rx interface to query a policy server to perform admission control for requests from subscribers on an access network in non-IMS environments.

Restrictions

- SBC does not provide preferred or alternate routes for SIP or DNS interfaces.
- SBC does not support use of Rx in combination with local call transfers.
- Lawful Intercept of media for calls using Rx is not possible.
- SBC does not support Packet Cable billing on Rx interfaces.

Call Failures

If the PCRF fails to respond to a request from SBC, SBC treats only the individual request as failed.

Only fully established calls are maintained during redundant switchovers. Calls in the process of being set up are dropped.

Configuration

• See the ?\$paranum>Configuring IMS Rx? section on page 57-3 for the procedure for configuring an IMS Rx Interface.

• See the ?\$paranum>Configuration Examples for Diameter Routing? section on page 57-14 for configuration examples of IMS Rx.

Configuring IMS Rx

This section describes the following procedures:

- Configuring an IMS Rx Interface, page 57-3
- Configuring Media Service for IMS Rx, page 57-4
- Disabling Preliminary AAR Messages, page 57-6

Configuring an IMS Rx Interface

Use the following procedure to configure an IMS Rx interface.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ims realm realm-name
- 6. ims rx
- 7. ims pani
- 8. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc MySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	

	Command or Action	Purpose
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe) adjacency sip A_1	
Step 5	ims realm realm-name	Configures an IMS realm for use by an IMS Rx interface.
	Example: Router(config-sbc-sbe-adj-sip)# ims realm Realm_1	
Step 6	ims rx	Configures an IMS Rx interface for access adjacency
	Example: Router(config-sbc-sbe-adj-sip)# ims rx	
Step 7	ims pani	(Optional) Configures the P-Access-Network-Info (PANI) header process preference for the adjacency.
	Example:	
	Router(config-sbc-sbe-adj-sip)# ims pani	
Step 8	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	

Configuring Media Service for IMS Rx

Use the following procedure to configure media service for IMS Rx.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type policy-set
- 7. entry entry-id
- 8. ims media-service
- 9. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an SBE policy set
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.
Step 6	table-type policy-set	Configures a CAC table to allow the use of media resources and 3rd party transcoding resources as well as Rx resources the table type of a CAC table within the context of an SBE
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set	policy set.
Step 7	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 8	ims media-service	(Optional) Configures a CAC table to allow the use of media resources and third party transcoding resources as
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # ims media-service</pre>	well as Kx resources.
Step 9	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	

Disabling Preliminary AAR Messages

Use the following procedure optionally to prevent preliminary AAR messages from being sent during an IMS Rx session.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. cac-policy-set policy-set-id
- 5. cac-table table-name
- 6. table-type policy-set
- 7. entry entry-id
- 8. ims rx preliminary-aar-forbid
- 9. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc SBC1	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	cac-policy-set policy-set-id	Enters the mode of CAC policy set configuration within an SBE entity, creating a new policy set if necessary.
	Example: Router(config-sbc-sbe)# cac-policy-set 1	<i>policy-set-id</i> —Integer chosen by the user to identify the policy set. The range is 1 to 2147483647.
Step 5	cac-table table-name	Enters the mode for configuration of an admission control table (creating one if necessary) within the context of an
	Example: Router(config-sbc-sbe-cacpolicy)# cac-table testSecure	<i>table-name</i> —Name of the admission control table.

	Command or Action	Purpose
Step 6	<pre>table-type policy-set Example: Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set</pre>	Configures a CAC table to allow the use of media resources and third party transcoding resources as well as Rx resources the table type of a CAC table within the context of an SBE policy set.
Step 7	entry entry-id	Enters the mode to modify an entry in an admission control table.
	Example: Router(config-sbc-sbe-cacpolicy-cactable)# entry 1	<i>entry-id</i> —Specifies the table entry.
Step 8	ims rx preliminary-aar-forbid	Prevents preliminary AAR messages from being sent during an IMS Rx session.
	<pre>Example: Router(config-sbc-sbe-cacpolicy-cactable-entry) # ims rx preliminary-aar-forbid</pre>	
Step 9	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	

Configuration Examples for IMS Rx

This section provides the following examples:

- Configuration Example for IMS Rx Interface, page 57-7
- Configuration Example for IMS Rx Media Service, page 57-7
- Configuration Example for Disabling Preliminary AAR Messages, page 57-8

Configuration Example for IMS Rx Interface

The following example shows how to configure an IMS Rx interface:

```
Router# configure terminal
Router(config)# sbc mySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# adjacency sip A_1
Router(config-sbc-sbe-adj-sip)# ims realm Realm_1
Router(config-sbc-sbe-adj-sip)# ims rx
Router(config-sbc-sbe-adj-sip)# ims pani
Router(config-sbc-sbe-adj-sip)# end
```

Configuration Example for IMS Rx Media Service

The following example shows how to configure media service for IMS Rx:

```
Router# configure terminal
Router(config)# sbc MySBC
Router(config-sbc)# sbe
```

```
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table my_table
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# ims media-service
Router(config-sbc-sbe-cacpolicy-cactable-entry)# end
```

Configuration Example for Disabling Preliminary AAR Messages

The following example shows how to prevent preliminary AAR messages from being sent during an IMS Rx session (optional):

```
Router# configure terminal
Router(config)# sbc MySBC
Router(config-sbc)# sbe
Router(config-sbc-sbe)# cac-policy-set 1
Router(config-sbc-sbe-cacpolicy)# cac-table my_table
Router(config-sbc-sbe-cacpolicy-cactable)# table-type policy-set
Router(config-sbc-sbe-cacpolicy-cactable)# entry 1
Router(config-sbc-sbe-cacpolicy-cactable-entry)# ims rx preliminary-aar-forbid
Router(config-sbc-sbe-cacpolicy-cactable-entry)# end
```

Information About the Diameter Protocol in the SBC

Diameter is an Authentication Authorization Accounting (AAA) protocol and is an enhanced version of the RADIUS (Remote Authentication Dial-In User Service) protocol. Diameter is the protocol of choice for the next generation IMS network developed by 3GPP.

When the Diameter protocol is implemented on a network, the Policy Charging and Rules Function (PCRF) acts as the Diameter server and the Application Function (AF), in our case SBC, acts as the Diameter client. SBC performs the functions of an IMS Rx Diameter client application and handles policy information and media reservations at the border of an access network.

SBC Diameter provides users with the option of configuring of either of two types of routing:

- Host-based routing
- Realm-based routing where multiple peers can be configured

Interfaces are referred as reference points in IMS. Reference points are named using unique acronyms, such as Rx (receiving reference point).

Features Supported

The following features are supported by SBC Diameter:

- SBC Diameter runs over TCP.
- SBC Diameter uses IPv4 addressing only.
- SBC Diameter supports IP Security Protocol (IPSEC).
- SBC Diameter supports multiple peers per realm.
- SBC Diameter supports redundancy switchover of Diameter peers as follows:
 - All Diameter messages are sent to the primary peer of the realm by default.
 - If the primary peer fails, Diameter switches to a secondary peer and retransmits all pending messages

Restrictions

SBC Diameter has the following restrictions:

- SBC Diameter does not replicate states or outstanding requests during redundancy switchovers. All states and outstanding requests are lost after a switchover from a failed active connection to a backup connection.
- SBC Diameter does not support IPv6 addressing.
- IPv6 is not supported.

Configuration

See the ?\$paranum>Configuring SBC Diameter Routing? section on page 57-9 for the procedure for configuring the Diameter protocol in SBC.

See the ?\$paranum>Configuration Examples for Diameter Routing? section on page 57-14 for configuration examples of the Diameter protocol in SBC.

Configuring SBC Diameter Routing

This section provides two routing configurations:

- Configuring Diameter Host-Based Routing, page 57-9
- Configuring Diameter Realm-Based Routing, page 57-11

Configuring Diameter Host-Based Routing

Use the following procedure to configure Diameter host-based routing. This procedure sets up an Rx adjacency first and then the Diameter host-based routing.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ims realm realm-name
- 6. ims rx pcrf pcrf-name
- 7. ims pani [received | rx | received rx | rx received]
- 8. exit
- 9. diameter
- 10. origin-realm realm-name
- 11. origin-host host-name
- 12. activate
- 13. end
- 14. show sbc sbc-name sbe diameter

- **15. show sbc** *sbc-name* **sbe diameter peers** *peer-name*
- **16**. **show sbc** *sbc-name* **sbe diameter stats**

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc MySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe) adjacency sip Adj_1	
Step 5	ims realm realm-name	Creates an IMS realm for the Rx.
	Example: Router(config-sbc-sbe-adj-sip)#ims realm Rx_Realm_1	
Step 6	ims rx pcrf pcrf-name	Configures an IMS Rx reference point on this SIP adjacency and specifies the PCRF host where the Rx messages are
	<pre>Example: Router(config-sbc-sbe-adj-sip)# ims rx pcrf cisco.com</pre>	fouted.
Step 7	ims pani [received rx received rx rx received]	(Optional) Configures the P-Access-Network-Info (PANI) header process preference for the adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# ims pani rx received	
Step 8	exit	Exits to the previous mode.
	Example: Router(config-sbc-sbe-enum)# exit	
Step 9	diameter	Enters the Diameter configuration mode.
	Example: Router(config-sbc-sbe)# diameter	

	Command or Action	Purpose
Step 10	origin-realm realm-name	Configures the name of SBC's local realm for diameter messages.
	Example: Router(config-sbc-sbe-diameter)# origin-realm cisco.com	
Step 11	origin-host host-name	Configures the name of SBC's local host for diameter messages.
	<pre>Example: Router(config-sbc-sbe-diameter)# origin-host sbc.cisco.com</pre>	
Step 12	activate	Activates Diameter host-based routing.
	Example: Router(config-sbc-sbe-enum)# activate	
Step 13	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	
Step 14	show sbc sbc-name sbe diameter	Displays the local configuration information for Diameter.
	Example: Router# show sbc MySBC sbe diameter	
Step 15	show sbc sbc-name sbe diameter peers peer-name	Displays the configuration information for IMS peers.
	Example: Router# show sbc MySBC sbe diameter peers Peer1	
Step 16	show sbc sbc-name sbe diameter stats	Displays the transport statistics for an IMS peer.
	Example: Router# show sbc MySBC sbe diameter stats	

Configuring Diameter Realm-Based Routing

Use the following procedure to configure Diameter realm-based routing.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ims realm realm-name
- 6. ims rx
- 7. exit

- 8. diameter
- 9. origin-realm realm-name
- **10.** origin-host host-name
- 11. peer peer-name ipv4 ipv4-address
- 12. peer peer-name ipv4 ipv4-address
- **13.** realm realm-name [app rx] peer peer-name [priority priority]
- 14. realm realm-name [app rx] peer peer-name [priority priority]
- 15. activate
- 16. end
- 17. show sbc *sbc-name* sbe diameter peers
- 18. show sbc *sbc-name* sbe diameter peers *peer-name*
- 19. show sbc sbc-name sbe diameter peers peer-name

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on Cisco Unified Border Element (SP Edition) and enters into SBC configuration mode.
	Example: Router(config)# sbc MySBC	
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the mode of an SBE SIP adjacency.
	Example: Router(config-sbc-sbe) adjacency sip Adj_1	
Step 5	ims realm realm-name	Creates an IMS realm for the Rx.
	Example: Router(config-sbc-sbe-adj-sip)#ims realm Rx_Realm_1	
Step 6	ims rx	Configures an IMS Rx reference point on this SIP adjacency.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# ims rx pcrf cisco.com</pre>	

	Command or Action	Purpose
Step 7	exit	Exits to the previous mode.
	Example: Router(config-sbc-sbe-enum)# exit	
Step 8	diameter	Enters the Diameter configuration mode.
	Example: Router(config-sbc-sbe)# diameter	
Step 9	origin-realm realm-name	Configures the domain name of an IMS local realm.
	Example: Router(config-sbc-sbe-diameter)# origin-realm cisco.com	
Step 10	origin-host host-name	Configures the domain name of the local IMS host.
	Example: Router(config-sbc-sbe-diameter)# origin-host sbc.cisco.com	
Step 11	peer peer-name ipv4 ipv4-address	Configures the name and IPv4 address of peerA.
	Example: Router(config-sbc-sbe-diameter)# peer peerA address ipv4 1.2.3.4	
Step 12	<pre>peer peer-name ipv4 ipv4-address</pre>	Configures the name and IPv4 address of peerB.
	Example: Router(config-sbc-sbe-diameter)# peer peerB address ipv4 1.2.3.5	
Step 13	<pre>realm realm-name [app rx] peer peer-name [priority priority]</pre>	Configures a peer and assign the peer to the realm.
	Example: Router(config-sbc-sbe-diameter)# realm test.com app rx peer peerA	
Step 14	<pre>realm realm-name [app rx] peer peer-name [priority priority]</pre>	Configures another peer and assign the peer to the realm.
	Example: Router(config-sbc-sbe-diameter)# realm test.com app rx peer peerB priority 10	
Step 15	activate	Activates Diameter realm-based routing.
	Example: Router(config-sbc-sbe-enum)# activate	

	Command or Action	Purpose
Step 16	end	Exits configuration mode and returns to privileged EXEC mode.
	Example: Router(config-sbc-sbe-enum-entry)# end	
Step 17	show sbc sbc-name sbe diameter peers	Displays the configuration information for all IMS peers.
	Example: Router# show sbc MySBC sbe diameter peers	
Step 18	show sbc sbc-name sbe diameter peers peer-name	Displays the configuration information for peerA.
	Example: Router# show sbc MySBC sbe diameter peers peerA	
Step 19	show sbc sbc-name sbe diameter peers peer-name	Displays the configuration information for peerB.
	Example: Router# show sbc MySBC sbe diameter peers peerB	

Configuration Examples for Diameter Routing

This section provides the following examples:

- Configuration Example for Diameter Host-Based Routing, page 57-14
- Configuration Example for Diameter Realm-Based Routing, page 57-15

Configuration Example for Diameter Host-Based Routing

The following example shows how to configure Diameter host-based routing:

```
Router# configure terminal
Router(config) # sbc MySBC
Router(config-sbc) # sbe
Router(config-sbc-sbe) adjacency sip Adj_1
Router(config-sbc-sbe-adj-sip)# ims realm Rx_Realm_1
Router(config-sbc-sbe-adj-sip) # ims rx pcrf cisco.com
Router(config-sbc-sbe-adj-sip)# ims pani
Router(config-sbc-sbe-enum)# exit
Router(config-sbc-sbe)# diameter
Router(config-sbc-sbe-diameter) # origin-realm cisco.com
Router(config-sbc-sbe-diameter) # origin-host sbc.cisco.com
Router(config-sbc-sbe-enum)# activate
Router(config-sbc-sbe-enum-entry)# end
Router# show sbc MySBC sbe diameter
Router# show sbc MySBC sbe diameter peers Peer1
Router# show sbc MySBC sbe diameter stats
```

Configuration Example for Diameter Realm-Based Routing

The following example shows how to configure Diameter realm-based routing:

```
Router# configure terminal
Router(config) # sbc MySBC
Router(config-sbc) # sbe
Router(config-sbc-sbe) adjacency sip Adj_1
Router(config-sbc-sbe-adj-sip)# ims realm Rx_Realm_1
Router(config-sbc-sbe-adj-sip)# ims rx
Router(config-sbc-sbe-enum)# exit
Router(config-sbc-sbe) # diameter
Router(config-sbc-sbe-diameter) # origin-realm cisco.com
Router(config-sbc-sbe-diameter) # origin-host sbc.cisco.com
Router(config-sbc-sbe-diameter)# peer peerA address ipv4 1.2.3.4
Router(config-sbc-sbe-diameter)# peer peerB address ipv4 1.2.3.5
Router(config-sbc-sbe-diameter)# realm test.com app rx peer peerA
Router(config-sbc-sbe-diameter)# realm test.com app rx peer peerB priority 10
Router(config-sbc-sbe-enum) # activate
Router(config-sbc-sbe-enum-entry)# end
Router# show sbc MySBC sbe diameter peers
Router# show sbc MySBC sbe diameter peers peerA
Router# show sbc MySBC sbe diameter peers peerB
```

```
Note
```

You can use the following, existing ASR1000 IPSEC functionality to provide secure Diameter protocol transport:

```
crypto isakmp policy 1
     encr aes
     authentication pre-share
    group 2
crypto isakmp key cisco123 address 0.0.0.0 0.0.0.0
crypto ipsec transform-set testcpoc esp-des esp-md5-hmac
crypto map diamap 10 ipsec-isakmp
     set peer 192.68.9.1
     set security-association lifetime kilobytes 536870912
     set transform-set testcpoc
    match address 199
access-list 199 permit ip 192.169.0.0 0.0.255.255 193.169.0.0 0.0.255.255
interface SBC01
ip address 192.68.9.2 255.255.255.0
crypto map diamap
```

Information About IMS Rf Billing Interfaces

The SBC supports Rf billing interfaces for SIP-to-SIP calls when operating as a Proxy Call Session Control Function (P-CSCF) and as an Interconnection Border Control Function (IBCF). The Charging Trigger Function (CTF) in the SBC uses an Rf billing interface to provide offline charging information to the billing domain in an IMS network. The Rf billing interface uses the Diameter protocol for sending billing information to the Charging Data Function (CDF). Offline charging is used for network services that are paid periodically, for example, a user may have a subscription for voice calls that is paid for on a monthly basis.

In IMS, billing information originates from the CTF. The CTF sends Accounting Request (ACR) messages containing billing information to the CDF, which collates this information into event-based and session-based Call Detail Record (CDR) files. The CDF then passes the files to the Charging Gateway Function (CGF), which is responsible for nonvolatile storage of the CDRs and for other functions such as, duplicate detection, error correction, and filtering. The CGF transfers the files to the billing domain for eventual account reconciliation. This final transfer is not time sensitive and can occur in batch mode. The billing domain uses the CDR to charge for the services used.

Offline Charging Events

For both event-based charging and session-based charging, the CTF supports the accounting state machine. The task of reporting offline charging events to the CDF is managed through a Diameter Accounting Request (ACR) message. The IMS Rf interface supports the ACR event types described in Table 57-1.

Event Type	Description
START	Starts an accounting session.
INTERIM	Updates an accounting session.
STOP	Stops an accounting session.
EVENT	Indicates a one-time accounting event.

Table 57-1IMS Rf ACR Event Types

The START, INTERIM, and STOP event types are used for session-based charging. The EVENT type is used either for event-based charging or to indicate a failed attempt at establishing a session.

Rf Billing Error Handling

This section describes how the SBC handles the various types of Rf billing errors.

CDF Connection Failure

If the connection to the primary CDF is broken, the SBC sends the corresponding charging information to the secondary CDF (if present). If statically configured CDFs are used, the secondary CDF is the redundant peer of the next highest priority. If the dynamic CDF discovery task is performed, the secondary CDF is the address in the next ccf parameter in the P-Charging-Function-Address header. This process continues until a CDF responds, or there are no more CDFs. In the latter scenario, if an appropriate file system is available, the charging messages are stored in the nonvolatile memory until the CDF connection is restored. The connection to any of the available CDFs has no impact on the call setup.

57-16
No Reply from CDF

Because DIAMETER messages are transmitted over TCP or Stream Control Transmission Protocol (SCTP), a missing Accounting Answer response to an ACR must indicate that a connection is going down. In such a scenario, the procedure described in CDF Connection Failure section is followed.

Failure Response from CDF

The CDF can return any failure encountered while collecting billing information from the SBC, in the ACA message, even though the connection to the peer is active.

If the failure return code is DIAMETER_UNABLE_TO_DELIVER, this message is cached in nonvolatile memory and follows the procedure described in CDF Connection Failure section.

If the failure return code is any other value, a PD log is created to convey this information to the user, but no other action is taken.

Duplicate Detection

The SBC does not retransmit DIAMETER requests because the underlying TCP transport handles such requests. The CDF does not handle duplicate requests from the SBC.

CDF Detected Failure

If the SBC fails over, some Rf sessions may not be closed correctly, for example, when a call is set up during failover. The CDF must close CDRs pertaining to a particular session if it detects that ACRs are not received within a certain period.

Restrictions for IMS Rf Billing Interfaces

The IMS Rf Billing Interfaces feature has the following restrictions:

- The SBC does not support Rf billing for SIP-to-H.323 calls and H.323-to-H.323 calls.
- The SBC does not support Rf billing in a non-IMS network.
- The SBC does not supply the PS-Information attribute-value pairs (AVP) on its messages. Therefore, the SBC does not send the Cisco Gateway GPRS Support Node (GGSN)-Address AVP.
- The SBC does not supply the Third Generation Partnership Project (3GPP)-Charging-ID AVP.

Configuring an IMS Rf Billing Interface

Use the following procedure to configure an IMS Rf billing interface.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip adjacency-name
- 5. ims rf
- 6. ims realm realm-name
- 7. exit

- 8. billing
- 9. method 3gpp-rf
- **10. rf** *index*
- 11. origin-host host-name
- 12. origin-realm realm-name
- **13. realm** *realm*-*name* [**usePCFAHeader** | **cdf** *cdf*-*name* {**FQDN** *FQDN*-*name* | **ipv4** *ipv4*-*address* | **vpn** *vpn*-*name*} [**port** *port*-*number*] [**priority** *priority*-*number*]]
- 14. attach
- 15. activate
- 16. end
- 17. show sbc sbc-name sbe adjacencies adjacency-name [authentication-realms | detail | peers]
- **18.** show sbc *sbc-name* sbe billing instance [*instance-index*] [rf {realms [*realm-name* current5mins]} | {cdfs [*cdf-name*]}]

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example:	
Step 2	sbc sbc-name	Creates the SBC service on the Cisco Unified Border Element (SP Edition) and enters the SBC configuration
	Example: Router(config)# sbc <i>MvSBC</i>	mode.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip adjacency-name	Enters the SBE SIP adjacency mode.
	Example: Router(config-sbc-sbe) adjacency <i>sip</i> A_1	
Step 5	ims rf	Configures an IMS Rf interface for access adjacency.
	Example: Router(config-sbc-sbe-adj-sip)# ims rf	
Step 6	ims realm realm-name	Configures an IMS realm for use by an IMS Rf interface.
	Example: Router(config-sbc-sbe-adj-sip)# ims realm <i>Realm_1</i>	

	Command or Action	Purpose
Step 7	exit	Exits the SBE SIP adjacency mode.
	Example: Router(config-sbc-sbe-adj-sip)# exit	
Step 8	billing	Configures the IMS Rf billing method.
	Example: Router(config-sbc-sbe)# billing	
Step 9	method 3gpp-rf	Enables the 3GPP Rf billing method on the SBC.
	Example: Router(config-sbc-sbe-billing)# method 3gpp-rf	
Step 10	rf index	Creates a new Rf billing instance.
	Example: Router(config-sbc-sbe-billing)# rf 0	
Step 11	origin-host host-name	Configures the domain name of an IMS local host. This value is displayed in the diameter Origin-Host AVP.
	<pre>Example: Router(config-sbc-sbe-billing-rf)# origin-host sbc.com</pre>	
Step 12	origin-realm realm-name	Configures the domain name of an IMS local realm. This value is displayed in the diameter Origin-Realm AVP.
	Example: Router(config-sbc-sbe-billing-rf)# origin-realm <i>cisco.com</i>	
Step 13	<pre>realm realm-name [usePCFAHeader cdf cdf-name {FQDN FQDN-name ipv4 ipv4-address vpn vpn-name} [port port-number] [priority priority-number]]</pre>	Enables dynamic CDF detection.
	Example: Router(config-sbc-sbe-billing-rf)# realm cisco.com cdf cdf1 ipv4 192.0.2.1 port 3688	
Step 14	attach	Attaches an adjacency to an account on the SBE.
	Example: Router(config-sbc-sbe-billing-rf)# attach	
Step 15	activate	Activates billing after it is configured.
	Example: Router(config-sbc-sbe-billing-rf)# activate	
Step 16	end	Exits the configuration mode and returns to the privileged EXEC mode.
	Example: Router(config-sbc-sbe-billing-rf)# end	

	Command or Action	Purpose
Step 17	<pre>show sbc sbc-name sbe adjacencies adjacency-name [authentication-realms detail peers]</pre>	Displays whether IMS Rf is enabled or not.
	Example: Router# show sbc asr sbe adjacencies adj1 detail	
Step 18	<pre>show sbc sbc-name sbe billing instance [instance-index] [rf {realms [realm-name current5mins]} {cdfs [cdf-name]}]</pre>	Displays the configuration of the Rf billing interface.
	Example: Router# show sbc asr sbe billing instance 6 rf realms realm1 current5mins	

Configuration Example for IMS Rf Billing Interface

The following example shows how to configure the IMS Rf Billing Interface feature:

```
configure terminal
sbc MySBC
sbe
adjacency sip test
ims rf
ims realm cisco.com
billing
   method 3GPP-RF
   rf 0
       orig-host sbc.com
       orig-realm cisco.com
       rf 0 realm cisco.com cdf cdf1 ipv4 1.2.3.4 port 3688
       rf 0 realm cisco.com cdf cdf2 cdf.cisco.com priority 2
       attach
       activate
       end
```

The following is a sample output of the show sbc sbe billing instance command:

Router# show sbc asr sbe billing instance 1

Billing Manager Information:	
Local IP address:	3.3.3.3
LDR check time:	0:0
Method	rf
Admin Status:	UP
Operation Status:	UP
Billing Methods	
Instance:	1
Type:	3GPP-RF
Transport Mechanism Status:	FAILED
Active Calls Billed:	0
Deact-mode:	abort
Admin Status:	UP
Operation Status:	UP
LDR check time:	24:0
Origin Host:	yfasr.open-ims.test

Origin Realm:

open-ims.test





CALEA IRI Interface Support

The Communications Assistance for Law Enforcement Act (CALEA) intercept-related information (IRI) Interface Support feature enables service providers to define a legal warrant on VoIP endpoints to gather both signaling and media content information. The CALEA IRI Interface Support feature is based on PacketCable 1.5 standard specifications.

The CALEA IRI Interface Support feature is applicable to both Session Initiation Protocol (SIP) and H.323 calls in a unified Session Border Controller (SBC) configuration. It is not, however, applicable to distributed SBC.

Cisco Unified Border Element (SP Edition) was formerly known as Integrated Session Border Controller and may be commonly referred to in this document as the SBC.

For a complete description of the commands used in this chapter, refer to the *Cisco Unified Border Element (SP Edition) Command Reference: Unified Model* at:

http://www.cisco.com/en/US/docs/ios/sbc/command/reference/sbcu_book.html.

For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or a Cisco IOS master commands list.

Feature History for CALEA IRI Interface Support

Release	Modification
Cisco IOS XE Release	The CALEA IRI Interface Support feature was introduced on Cisco ASR
3.18	1000 Series Routers.

Contents

This module contains the following sections:

- Information About CALEA IRI Interface Support, page 58-2
- Restrictions for Implementing CALEA IRI Interface Support, page 58-12
- Implementing CALEA IRI Interface Support, page 58-13

Information About CALEA IRI Interface Support

The SBC can be used for the dual functions of Intercepting Control Element (ICE) and Intercepting Network Element (INE). You can place a request for a warrant using the Simple Network Management Protocol (SNMP) interface. The Cisco ASR 1000 series router responds with PacketCable1.5 messages and with replicated IP/UDP/RTP media packets, as required by the warrant.

You can also define the endpoint match using username, phone number, or SIP-Uniform Resource Identifier (URI). In addition, you can set up pen, trace, pen-and-trace, or intercept type of warrant.

You can define the VoIP endpoint information along with mediation device information using Simple Network Management Protocol Version 3 (SNMPv3) MIBs. The VoIP signaling information is sent from a router to a mediation device. In addition, the media content is tapped, replicated, encapsulated, and sent to the mediation device in real time.

Define the warrant by providing only the VoIP endpoint information. A Cisco ASR 1000 Series Router determines the local pinhole being used for a particular call, and replicates the call content to the mediation device. In addition, you can define the warrant by requesting only the call signaling-related information using PacketCable1.5 Event messages (IRI).

In the context of calls coming in on an adjacency, with the inherit profile set to preset-access, the source information from the SIP header will be used to match the configured warrants. In the context of the calls coming in on an adjacency, with the inherit profile set to preset-core, the destination information from the SIP header will be used to match the configured warrants. However, the provider can override these rules by configuring the **warrant match-order** command on the adjacencies.

For a registered SIP endpoint, we recommend setting cvoiptapStreamMatchType to URI.

When the VoIP call gets tapped, the Cisco ASR 1000 series router sends the locally generated unique Call Content Connection ID (CCCID) information using the RADIUS message. The same CCCID information is then used to encapsulate the media IP packet. An mediation device can use the CCCID information to correlate the signaling and media information. The VoIP LI warrant information can be retrieved using a secure SNMPv3 interface.

For each INTERCEPT, a unique IRI stream with CCCID information is present.

In a network setup of multiple Cisco ASR 1000 series routers, the CALEA IRI Interface Support feature is designed to tap the information on the router that is closest to the endpoint under surveillance.

This section contains the following information pertaining to the CALEA IRI Interface Support feature:

- CALEA IRI Interface Support Flow, page 58-3
- SNMP Row Indices, page 58-4
- Tap Interfaces, page 58-4

CALEA IRI Interface Support Flow

Figure 58-1 Flow of the CALEA IRI Interface Support Feature

Figure 58-1 shows the flow of the CALEA IRI Interface Support feature.

The steps pertaining to the flow of the CALEA IRI Interface Support feature are as follows:

- 1. Provisioning of mediation device information and VoIP warrant is done as a combination of SNMPv3 and IOS CLI commands on the Cisco ASR 1000 series router.
- 2. The calling party originates the call.
- **3.** If a warrant matches the signaling parameters, RADIUS messages are sent to the mediation device. The message contains the unique CCCID generated by the Cisco ASR 1000 series router.
- 4. The party that was called answers, and the media information starts flowing through the Cisco ASR 1000 series router.
- **5.** The Cisco ASR 1000 series router replicates the media information, and sends it to the mediation device.

SNMP Row Indices

Figure 58-2 represents the SNMP table and rows. There are two independent mediation device rows. The GenericStream and VoIP TAP MIB rows are the children of the IRI MD row. There is a CCIndex field in the VoIP TAP MIB row that captures the relationship with CC MD MIB row. A one-to-one relationship also exists between GenericStream and VoIP TAP MIB rows.





Tap Interfaces

This section describes the following Tap interfaces:

- IRI Interface, page 58-5
- CC Interface, page 58-11

IRI Interface

The PacketCable 1.5 standard specifications for Electronic Surveillance contains the packet definition for all IRI-related messages. Table 58-1 details the supported call event messages that are sent for each Tapped Call.

Event Message	Notes
Signaling_Start	Sent when signalling has commenced (inbound), and when it is about to commence (outbound), for example, received INVITE on inbound, and about to send INVITE on outbound for a SIP endpoint.
QoS_Reserve	Sent for the inbound leg when the inbound QoS is reserved, and for the outbound leg when the outbound QoS is reserved.
Call_Answer	Indicates that the terminating party has answered, and that media has started. This message is sent for both the legs simultaneously.
QoS_Commit	Sent when QoS is committed by the SBC. This message is sent for both the legs at the same time.
Call_Disconnect	Sent when a call has been terminated, and media has ceased flowing. The message is sent for both the legs at the same time.
QoS_Release	Sent when the QoS is released by the SBC. Sent for both the legs at the same time.
Signaling_Stop	Sent when signaling is complete for each party in the call. The event is generated once for each party after the last signaling message is sent.
Media_Report	Sent by the SBC whenever a flow is created, modified, and released.
Surveillance_Stop	Sent by the SBC to indicate the end of the IRI or CC tapping or both. Generally, this means the end of a call.
Redirection	Sent by the SBC when a call has been transferred, either due to a 3XX redirect response, or a SIP REFER request.

 Table 58-1
 Supported Call Event Messages

Call Event Messages

Table 58-2 details the Signaling_Start message attributes that are supported and sent when the SBC has information that the destination is routable and the originating endpoint is allowed to make the call.

 Table 58-2
 Signaling_Start Message Attributes

Attribute Name	Comment
EM_Header	Common header attribute.
Direction_Indicator	Specifies if the device represents an originating or terminating part of a call.
	1—originating
	2—terminating

MTA_Endpoint_Name	The SBC has no direct contact with the MTA. By default, the value is set to MTA Endpoint. Alternatively, the attribute could be configured to report adjacency or signalling address information.
Calling_Party_Number	The number of the calling party (if available). In the SBC, this is the canonical format of the number after inbound number translations, if any, and before the routing.
Called_Party_Number	The number of the called party (always present). In the SBC, this is the canonical format of the number after any inbound number translation and before routing.
Routing_Number	Indicates a routable number (always present).
User_Input	The number of the called party prior to any translation performed during inbound number analysis.
Translation_Input	The number of the called party after inbound number analysis and before routing, if different from the value supplied in User_Input.
Redirected_From_Info	If originating an INVITE in response to a 3XX or a REFER, the attribute is set to the previous destination of the call (the sender of the 3XX or REFER), the initial destination of the call (if there are multiple redirections), and the number of redirections so far on the call.
Carrier_Identification_Code	The Carrier Identification Code associated with this call.
Trunk_Group_ID	Trunk_Type set to 9. Signaling type is not specified.
	Trunk_Group_ID set to the Trunk Group ID associated with the side of the call that is being tapped.

Table 58-2	Signaling	Start Message	Attributes	(continued)

The following Signaling_Start message attributes are not included in the message:

- Attribute Name
- Location_Routing_Number
- Intl_Code
- Dial_Around_Code
- Jurisdiction_Information_Parameter
- Ported_In_Calling_Number
- Ported_In_Called_Number
- Called_Party_NP_source

- Calling_Party_NP_source
- Billing_Type
- Electronic_Surveillance_Indication

Table 58-3 details the QoS_Reserve message attributes. This message is generated when the SBC has reserved bandwidth (QoS) on the network. If the reserved bandwidth changes, QoS_Reserve and QoS_Commit messages are generated anew.

Table 58-3 QoS_Reserve Message Attributes

Attribute Name	Comments
EM_Header	Common header attribute.
QoS_Descriptor	Similar to the description of the QoS_Reserve message.
MTA_UDP_Portnum	The UDP port number on the network element endpoint. Because the SBC has no direct contact with the MTA, the attribute is set to 0.
Flow_Direction	1—upstream
	2—downstream
SF_ID	A Data-over-Cable Service Interface Specifications-specific attribute that is required, and generated by the CMTS in a PacketCable architecture. Because the SBC does not support DOCSIS, this attribute is always 0.
CCC_ID	The local CCC ID for this call. It is included if CC tapping is being done on the call.

Table 58-4 details the Call_Answer message attributes. This message indicates the earliest point at which two-way media is established. The SBC sends the message to the billing servers when the SBC is notified that the called party has answered the call.

Table 58-4 Call_Answer Message Attributes

Attribute Name	Comment
EM_Header	Common header attribute.
Charge_Number	The charge number during collect call, calling-card call, call billed to a third party, and so on.
	For the SBC, this is the calling number, unless the call has been diverted. The diverted call has a Diverted-By number.
Related_Call_Billing_Correlation_ID	The billing correlation ID (BCID) assigned to the leg from the terminating network element. The SBC does not share the BCID and financial entity ID (FEID) information with other network elements.



The FEID attribute is not sent in a Call_Answer message.

Table 58-5 details the QoS_Commit message attributes. This message is sent by the SBC when the gate bandwidth is committed. This message is sent after a QoS_Reserve message that has been sent previously.

Attribute Name	Comments
EM_Header	Common header attribute containing timestamp and BCID.
MTA_UDP_Portnum	The UDP port number on the network element endpoint. Because the SBC has no direct contact with the MTA, so the attribute is set to 0.
Flow_Direction	1—upstream
	2—downstream
SF_ID	This is always 0 because the SBC does not support DOCSIS.
Total_Bandwidth (attribute ID 253)	The total bandwidth being used by the streams described in a QoS_Commit message.
CCC_ID	The local CCC ID for a call. The attribute is included if CC tapping is being done on the call.

 Table 58-5
 QoS_Commit Message Attributes

The following attributes are not included in the QoS_Commit message:

- QoS_Descriptor
- Media_Session_Desc (attribute ID 254)

Table 58-6 details the Call_Disconnect message attributes. This message is generated by the SBC when a two-way media flow is terminated. This message immediately precedes the QoS_Release and Signaling_Stop messages, and is sent only after the Call_Answer message that has been sent previously.

Table 58-6 Call_Disconnect Message Attributes

Attribute Name	Comments
EM_Header	Common header attribute.
Call_Termination_Cause	Reason for termination of call.

Table 58-7 details the QoS_Release message attributes. This message is generated by the SBC when the reserved bandwidth is released.

Attribute Name	Comments	
EM_Header	Common header attribute containing timestamp and BCID.	
Flow_Direction	1—upstream	
	2—downstream	
SF_ID	A DOCSIS-specific attribute, Service Flow ID, generated by the CMTS in a PacketCable architecture. Because the SBC does not support DOCSIS, this attribute is always set to 0.	
CCC_ID	The local CCC ID for a call. The attribute is included if CC tapping is being done on the call.	

Table 58-7	QoS_	Release	Message	Attributes
------------	------	---------	---------	------------



The Media_Session_Desc (attribute ID 254) attribute is not sent with the QoS_Release message.

Table 58-8 details the Signaling_Stop message attributes. This message is sent during the following events:

- A terminating signalling request, for example, a SIP BYE, from the party terminating the call is acknowledged by the SBC.
- When the terminating signalling request for the party not terminating the call is sent by the SBC, and acknowledged by that party.



The Signaling_Stop message is not sent if the Signaling_Start message for this call is not sent.

Table 58-8 Signaling_Stop Message Attributes

Attribute Name	Comments
EM_Header	The header attribute that must be first in the message.
Related_Call_Billing_Correlation_ID	The BCID of the other leg. For example, if BCID is the caller, the attribute is for the callee.
Call_Termination_Cause	The reason the call was terminated.



The FEID attribute of the Signaling_Stop message is not included.

Table 58-9 details the Surveillance_Stop message attributes. This message is sent by SIG to indicate the end of IRI or CC tapping or both. This message means the call has ended.

Attribute Name	Comment
EM_Header	Common header attribute containing the timestamp and BCID.
Surveillance_Stop_Type	Always included.
	1—End of all surveillance.
	2—End of only CC tapping.
Surveillance_Stop_Destination	Always included.
	1—Surveillance_Stop applies to local surveillance only. The value 1 is not used by the SBC.
	2—Surveillance_Stop is applicable to both local and remote surveillance.
	3—Surveillance_Stop is applicable only to remote surveillance.

Table 58-9	Surveillance	Stop Message	Attributes
	_		



The Electronic_Surveillance_Indication attribute is not included in the Surveillance_Stop message.

Table 58-10 details the Media_Report message attributes. The message is specific to a flow. Therefore, if more than one flow is created at the same time, multiple event messages are sent, one per flow.

A Media_Report message is sent during the following events, when a flow is created, modified, and released:

- A flow is considered Created when the gate bandwidth for the flow is committed. A QoS_Commit message is also sent at the same time.
- A flow is considered Modified when the flow is renegotiated.
- A flow is considered Released when the gate bandwidth for the flow is released. A Qos_Release message is also sent at the same time.

Attribute Name	Comment	
EM_Header	Common header attribute containing the timestamp and BCID.	
CCC_ID	The local CCCID for a call. Included if CC tapping is being done on the call.	
SDP_Upstream	The upstream SDP for the flow, SDP corresponding to flow in direction of caller, on side indicated by Flow_Direction, is always included.	
SDP_Downstream	The downstream SDP for the flow. SDP corresponding to flow in direction of callee, on side indicated by Flow_Direction, is always included.	

Table 58-10 Media_Report Message Attributes

L

Channel_State	Always included.
	1—Open (Flow created)
	2—Change (Flow modified)
	3—Close (Flow released)
Flow_Direction	Always included. Specifies if the device is acting on behalf of an originating part or terminating part of a call at the time the message is generated.
	1—upstream (Caller side)
	2—downstream (Callee side)

Table 58-10 Media_Report Message Attributes (continued)

Table 58-11 details the Redirection message attributes. This message is sent by the SBC when a call has been transferred either due to a 3XX redirect response or a SIP REFER request.

Table 58-11	Redirection	Message	Attributes
-------------	-------------	---------	------------

Attribute Name	Comment
EM_Header	Common header attribute containing the timestamp and BCID.
Related_Call_Billing_Correlator	Always included. The BCID used previously for the old branch of a call.
Redirected_From_Party_Number	Always included. The number of the party a call is being transferred from or forwarded from.
Redirected_To_Party_Number	Always included. The number of the party a call is being transferred to or forwarded to.
Carrier_Identification_Code	The Carrier Identification Code associated with a call.

CC Interface

The PacketCable 1.5 standard specifications contain the packet header format for replicated voice content packets.

Figure 58-3 shows a replicated packet. The first three rows of the packet are the outer Layer2, Layer3, and Layer4 information. This information consists of destination IP and UDP port of the Mediation Device, and the source IP and UDP port of the Cisco ASR 1000 series router. The fourth row of the packet is the CCC ID that is used to correlate the signaling and media information. The last four rows of the packet are the original media packet that is being TAPed. It starts from Layer 3 IP, and is followed by UDP, RTP, and media payload.

Outer L2 Header	
Mediation Device Destination IP Address Cisco ASR 1000 Series Routers Local Source IP Address	
Mediation Device Destination UDP Port Cisco ASR 1000 Series Routers Local Source UDP Port	
CCC Identifier (4 bytes)	
Original IP Header	
Original UDP Header	
Original RTP Header	
Encoded Voice	079500

Figure 58-3 Packet Format

Restrictions for Implementing CALEA IRI Interface Support

The following restrictions and limitations are applicable to CALEA IRI Interface Support feature implementing:

- Only one mediation device IP address is supported.
- The IPv6 address pertaining to the mediation device is not supported. Only IPv4 address in the global routing space is supported for mediation device. The IPv4 address should not be associated to any virtual routing and forwarding (VRF).
- The mediation device's IP address must be accessible from the Cisco ASR 1000 series router global routing space. CISCO-TAP2-MIB does not allow mediation device IP address to be in a VRF.
- The Cisco ASR 1000 series router does not support the CLIs of the Cisco BTS 10200 Softswitch and the Cisco PGW 2200 Softswitch for warrant configuration.
- The PacketCable 2.0 standard specification for Electronic Surveillance is not supported.
- LI using the SIP P-DCS-LAES header is not supported.
- Tap is not applied to the existing calls.
- The IPv6 Media Addresses cannot be intercepted in a VRF, but can be intercepted in a global routing space. However, IPv4 Media Addresses can be intercepted both in the global routing space and the VRF.

Implementing CALEA IRI Interface Support

The following sections explain how to configure the CALEA IRI Interface Support feature:

- Configuring the SBC for CALEA IRI Interface Support, page 58-16
- Configuring VoIP LI SNMP, page 58-13
- Configuring the SBC for CALEA IRI Interface Support, page 58-16

Configuring LI

To see the SNMPv3 and SNMP View configuration information pertaining to the LI TAP definitions, see the How to Configure Lawful Intercept section in the *Cisco IOS and NX-OS Software Lawful Intercept Architecture* feature guide at:

http://www.cisco.com/en/US/docs/ios/sec_user_services/configuration/guide/sec_lawful_intercept.htm l#wp1077988

Use the following commands provided in the *Cisco IOS and NX-OS Software Lawful Intercept Architecture* feature guide to configure LI:

- snmp-server view view-name MIB-name included—Defines an SNMPv2 MIB view, and includes a MIB family in the view.
- **snmp-server group group-name v3 auth read view-name write view-name**—Defines a read and write view for a group using the User Security Model (SNMPv3) and the authNoPriv Security Level.
- snmp-server user user-name group-name v3 auth md5 auth-password—Defines an authentication password for a user by using the HMAC MD5 algorithm for authentication and V3 security model.

The following example shows how to enable the mediation device to access the lawful intercept MIBs. It creates an SNMP view (tapV) that includes three LI MIBs (CISCO-VoIp-Tap-MIB, CISCO-TAP2-MIB, and CISCO-IP-TAP-MIB). It also creates a user group that has read, write, and notify access to MIBs in the tapV view.

snmp-server view tapV ciscoVoIpTapMIB included snmp-server view tapV ciscoIpTapMIB included snmp-server view tapV ciscoTap2MIB included snmp-server group tapGrp v3 auth read tapV write tapV notify tapV snmp-server user li tapGrp v3 auth md5 cisco snmp-server community public

Configuring VoIP LI SNMP

SNMP provisioning is done using the SNMP research tools available for Sun workstations. However, you can use any tool that uses the SNMPv3 protocol.

The **setany** commands listed here are executed using the SNMP application. Note that these commands are not Cisco IOS CLI commands. It is assumed that SNMP has been configured on your routing device. A secure K9 image is required for the MIBs to work.

There are four parts to the following example:

- Adding the Mediation Device Information
- Adding the VoIP User Warrant
- Retrieving the Mediation Device and VoIP User Warrant Information
- Removing the VoIP User Warrant and Mediation Device Information

Adding the Mediation Device Information

Perform the following steps to add the mediation device information:

Step 1 Configure the mediation device IP, RADIUS receiving port, transport type, and shared RADIUS Key to receive Voice signaling information from the SBC through the PacketCable1.5 Event Messages.

The following example shows how to create the TAP2 MD Row for IRI, with an IP address of 101.10.7.61, UDP port of 1813, and RADIUS key of "cisco":

```
setany -v3 172.18.37.151 li cTap2MediationStatus.1 -i 5
setany -v3 172.18.37.151 li cTap2MediationTimeout.1 -o "07 da 05 08 0e 3b 37 06"
setany -v3 172.18.37.151 li cTap2MediationDestAddressType.1 -i 1
setany -v3 172.18.37.151 li cTap2MediationTransport.1 -i 6
setany -v3 172.18.37.151 li cTap2MediationSrcInterface.1 -i 0
setany -v3 172.18.37.151 li cTap2MediationDestAddress.1 -o "65 0a 07 3d"
setany -v3 172.18.37.151 li cTap2MediationDestAddress.1 -o "65 0a 07 3d"
setany -v3 172.18.37.151 li cTap2MediationDestAddress.1 -o "65 0a 07 3d"
setany -v3 172.18.37.151 li cTap2MediationDestPort.1 -g 1813
setany -v3 172.18.37.151 li cTap2MediationStatus.1 -i 1
```

Step 2 Configure the mediation device IP, Call Content (CC) receiving port, and transport type to receive Voice CC from the SBC.

The following example shows how to create the TAP2 Mediation Device Row for a CC, with an IP address of 101.10.7.61, and UDP port of 45000:

```
setany -v3 172.18.37.151 li cTap2MediationStatus.2 -i 5
setany -v3 172.18.37.151 li cTap2MediationDestAddressType.2 -i 1
setany -v3 172.18.37.151 li cTap2MediationTimeout.2 -o "07 da 05 08 0e 3b 37 06"
setany -v3 172.18.37.151 li cTap2MediationTransport.2 -i 1
setany -v3 172.18.37.151 li cTap2MediationSrcInterface.2 -i 0
setany -v3 172.18.37.151 li cTap2MediationDestAddress.2 -o "65 0a 07 3d"
setany -v3 172.18.37.151 li cTap2MediationDestPort.2 -g 45000
setany -v3 172.18.37.151 li cTap2MediationStatus.2 -i 1
```

Adding the VoIP User Warrant

Perform the following steps to add the VoIP user warrant:

Step 1 Configure the VoIP user warrant.

The following example shows how to create the VoIP TAP SNMP Row with a matching username for "712020" and type "Intercept":

```
setany -v3 172.18.37.151 li cvoiptapStreamRowStatus.1.1 -i 5
setany -v3 172.18.37.151 li cvoiptapStreamId.1.1 -o "72 72 2d 31"
setany -v3 172.18.37.151 li cvoiptapStreamType.1.1 -i 4
setany -v3 172.18.37.151 li cvoiptapStreamMatch.1.1 -o "37 31 32 30 32 30"
```

```
setany -v3 172.18.37.151 li cvoiptapStreamMatchType.1.1 -i 1
setany -v3 172.18.37.151 li cvoiptapStreamCCMediationDevice.1.1 -i 2
setany -v3 172.18.37.151 li cvoiptapStreamRowStatus.1.1 -i 1
```

Step 2 The following example shows how to configure an associated generic stream for VoIP, and enable generic stream:

```
setany -v3 172.18.37.151 li cTap2StreamStatus.1.1 -i 5
setany -v3 172.18.37.151 li cTap2StreamType.1.1 -i 6
setany -v3 172.18.37.151 li cTap2StreamInterceptEnable.1.1 -i 1
setany -v3 172.18.37.151 li cTap2StreamStatus.1.1 -i 1
```

Retrieving the Mediation Device and VoIP User Warrant Information

Perform the following steps to retrieve the mediation device and VoIP user warrant information:

```
Step 1
        The following example shows how to retrieve the MD TAP2 SNMP row:
        getmany -v3 172.18.37.151 li ciscoTap2MIB
        SNMP GETMANY for the configured values
        cTap2MediationCapabilities.0 = ipV4SrcInterface(0), udp(2), radius(7)
        cTap2MediationDestAddressType.1 = ipv4(1)
        cTap2MediationDestAddressType.2 = ipv4(1)
        cTap2MediationDestAddress.1 = 65 0a 07 3d
        cTap2MediationDestAddress.2 = 65 0a 07 3d
        cTap2MediationDestPort.1 = 1813
        cTap2MediationDestPort.2 = 45000
        cTap2MediationSrcInterface.1 = 0
        cTap2MediationSrcInterface.2 = 0
        cTap2MediationRtcpPort.1 = 0
        cTap2MediationRtcpPort.2 = 0
        cTap2MediationDscp.1 = 0
        cTap2MediationDscp.2 = 0
        cTap2MediationDataType.1 = 0
        cTap2MediationDataType.2 = 0
        cTap2MediationRetransmitType.1 = 0
        cTap2MediationRetransmitType.2 = 0
                                                0e 3b 37 06
        cTap2MediationTimeout.1 = 07 da 05 08
        cTap2MediationTimeout.2 = 07 da 05 08
                                                 0e 3b 37 06
        cTap2MediationTransport.1 = radius(6)
        cTap2MediationTransport.2 = udp(1)
        cTap2MediationNotificationEnable.1 = true(1)
        cTap2MediationNotificationEnable.2 = true(1)
```

```
cTap2StreamType.1.1 = voip(6)
cTap2StreamInterceptEnable.1.1 = true(1)
cTap2StreamInterceptedPackets.1.1 = 0
cTap2StreamInterceptDrops.1.1 = 0
cTap2StreamStatus.1.1 = active(1)
cTap2StreamInterceptedHCPackets.1.1 = 0x000000000
cTap2StreamInterceptHCDrops.1.1 = 0x00000000
```

cTap2MediationStatus.1 = active(1) cTap2MediationStatus.2 = active(1) cTap2MediationRadiusKey.1 = cisco cTap2MediationRadiusKey.2 =

Step 2 The following example shows how to retrieve the VoIP TAP SNMP row:

```
getmany -v3 172.18.37.151 li ciscoVoIpTapMIB
```

```
cvoiptapStreamCapabilities.0 = tapEnable(0), usernameOrNumber(1), uri(2)
cvoiptapStreamId.1.1 = rr-1
cvoiptapStreamType.1.1 = intercept(4)
cvoiptapStreamMatch.1.1 = 712020
cvoiptapStreamMatchType.1.1 = usernameOrNumber(1)
cvoiptapStreamCCMediationDevice.1.1 = 2
cvoiptapStreamRowStatus.1.1 = active(1)
```

Removing the VoIP User Warrant and Mediation Device Information

Perform the following steps to remove the VoIP user warrant and mediation device information:

```
Step 1 Disable and delete the generic stream, and delete the VoIP User TAP row:
setany -v3 172.18.37.151 li cTap2StreamInterceptEnable.1.1 -i 2
setany -v3 172.18.37.151 li cvoiptapStreamRowStatus.1.1 -i 6
setany -v3 172.18.37.151 li cvoiptapStreamRowStatus.1.1 -i 6
Step 2 Remove the mediation device RADIUS receiving port:
setany -v3 172.18.37.151 li cTap2MediationStatus.1 -i 6
Step 3 Remove the MD CC receiving Port:
setany -v3 172.18.37.151 li cTap2MediationStatus.2 -i 6
```

Configuring the SBC for CALEA IRI Interface Support

This section details the steps involved in overriding the default match-order.

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. adjacency sip | h323 adjacency-name
- 5. warrant match-order [source | destination | diverted-by]
- 6. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enables global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Enters the mode of an SBC service.
	Example: Router(config)# sbc mysbc	• Use the <i>sbc-name</i> argument to define the name of the service.

	Command or Action	Purpose
Step 3	sbe	Enters the mode of a signaling border element (SBE) entity within an SBC service.
	Example: Router(config-sbc)# sbe	
Step 4	adjacency sip h323 adjacency-name	Enters the mode of an SBE SIP or H.323 adjacency.
	Example: Router(config-sbc-sbe) # adjacency sip sipadj	• Use the <i>adjacency-name</i> argument to define the name of the SIP or H.323 adjacency.
Step 5	warrant match-order [source destination diverted-by]	Configures the lawful inforcement warrant information in an SIP or H.323 adjacency, and specifies the order of fields used to match the warrant.
	Example: Router(config-sbc-sbe-adj-sip)# warrant match-order source destination diverted-by	By default, the incoming Access adjacency matches the source information, and the Core adjacency matches the destination information.
		Note The H.323 adjacency does not support the diverted-by keyword.
Step 6	exit	Exits the adjacency mode to the SBE mode.
	<pre>Example: Router(config-sbc-sbe-adj-sip)# exit</pre>	

The following example shows how to configure the SBC to override the default match-order:

configure terminal
sbc mySBC
sbe
adjacency sip adj1
warrant match-order source destination diverted-by





H.248 Border Access Controller Support

H.248 is a media gateway control protocol that enables Switched Circuit Network (SCN) to transmit voice traffic over IP. The H.248 protocol specifies master-slave architecture for decomposed gateways. In master-slave architecture, the Media Gateway Controller (MGC) is the master server and media gateways are the slave clients that behave as simple switches. One MGC can serve multiple media gateways. The H.248 protocol enables the creation, modification, and deletion of media streams across a media gateway, including the capability to negotiate the media formats to be used.

Feature History for H.248 Border Access Controller Support

Release	Modification
Cisco IOS XE Release 3.7	This feature was introduced on the Cisco ASR 1000 Series Routers.

Contents

This chapter contains the following sections:

- Support for the H.248 Border Access Controller, page 59-1
- Restrictions for H.248 BAC Support, page 59-3
- Prerequisites for Configuring H.248 BAC Support, page 59-4
- Configuring H.248 BAC Support, page 59-4
- Configuration Example for H.248 BAC Support, page 59-7

Support for the H.248 Border Access Controller

The session border controller (SBC) supports the H.248 Border Access Controller (BAC) feature.

This feature protects the core network (with Integrated Access Devices [IADs]) from heartbeat flooding and register flooding. The BAC can terminate heartbeat from the H.248 IADs, initiate heartbeat towards IADs, and limit the register rate from IADs to the core network. The BAC hides the core MGC network topology from the IAD access adjacency, and supports media forwarding. The BAC is placed at the edge of the core network. Figure 59-1 illustrates the H.248 BAC network topology.



Cisco Unified Border Element (SP Edition) Configuration Guide: Unified Model

The H.248 BAC supports the following functionalities:

• Termination of heartbeats from an access adjacency

The BAC has two adjacencies: access adjacency and core adjacency. Only one-to-one mapping is allowed between an access adjacency and a core adjacency. The IADs and the H.248 terminal devices reside on the access adjacency. The Access Gateway Control Function (AGCF) and Media Gateway Control Function (MGCF) reside on the core adjacency. The H.248 terminal devices on the access adjacency periodically send heartbeats to the AGCF through the BAC. To decrease the impact of heartbeats on the performance of core adjacency devices such as AGCF or MGCF, the BAC sends its response to the heartbeats from the access adjacency and does not transit to the core adjacency. Therefore, the BAC can terminate heartbeats from the access adjacency.

• Topology hiding

The BAC can modify the signaling address and the media address of the IADs and the AGCF. If it modifies these addresses, the peer will not know the original IP address of the corresponding IAD.

• Attack detection and protection

The BAC can detect whether a signal message is from a valid or invalid H.248 terminal device. If the signal message is from an invalid H.248 terminal device, it is discarded.

• Media anchoring and forwarding

The BAC can translate a media address according to the required configuration. When H.248 terminal devices reside in the same network, media will not flow through the core network. When media bypass is enabled, media does not anchor on the BAC.

• Signaling trace and debug

The BAC can supply different debug levels for H.248 signaling.

Restrictions for H.248 BAC Support

Following are the restrictions pertaining to the H.248 BAC Support feature:

- Multiple H.248 transactions in one H.248 packet are not supported.
- H.248 signaling interworking with SIP calls or H.323 calls is not supported.
- Multiple streams in local descriptors and remote descriptors are not supported.
- Memory and CPU throttle are not supported.
- Auto media bypass is not supported.
- IPv6 is not supported.
- The BAC cannot operate in the DBE mode.
- The BAC supports only the H.248 text format (long and short) message type, and not the binary format.

Prerequisites for Configuring H.248 BAC Support

The SBC must be activated before configuring the H.248 BAC Support feature. Perform the following procedure to activate the SBC:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc sbc-name
- 3. sbe
- 4. activate

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc sbc-name	Creates the SBC service on the Cisco Unified Border Element (SP Edition) and enters the SBC configuration
	Example: Router(config)# sbc <i>mySbc</i>	mode.
Step 3	sbe	Enters the mode of the signaling border element (SBE) function of the SBC.
	Example: Router(config-sbc)# sbe	
Step 4	activate	Activates the SBC service.
	Example: Router(config-sbc-sbe)# activate	

Configuring H.248 BAC Support

Perform the following procedure to configure the H.248 BAC Support feature:

SUMMARY STEPS

- 1. configure terminal
- 2. sbc h248 bac
- 3. media-address ipv4 ipv4-address realm realm-number vrf vrf-name
- 4. port-range port-range
- 5. adjacency h248 {core core-adjacency name}

- 6. control-address ipv4 ipv4-address {port port number | port-range minimum-port number maximum-port number}
- 7. remote-address ipv4 ipv4-address port port number
- **8**. **realm** *realm-number*
- 9. attach
- 10. exit
- 11. adjacency h248 {access access-adjacency name}
- 12. control-address ipv4 ipv4-address {port port number}
- **13**. **audit interval** *idle time*
- 14. heart-beat terminate terminate-interval
- **15.** domain-name domain-name
- **16. core-adj** *core adjacency-name*
- **17**. **realm** *realm-number*
- 18. attach

DETAILED STEPS

	Command or Action	Purpose
Step 1	configure terminal	Enters the global configuration mode.
	Example: Router# configure terminal	
Step 2	sbc h248 bac	Configures the SBC H.248 BAC.
	Example: Router(config-h248-bac)# sbc h248 bac	
Step 3	<pre>media-address ipv4 ipv4-address realm realm-number vrf vrf-name</pre>	Adds an IPv4 address to the set of addresses that the BAC can use as a local media address.
	Example: Router(config-h248-bac)# media-address ipv4 8.8.8.8 realm 1	
Step 4	port-range port range	Configures the port range of the media address.
	Example: Router(config-h248-bac-media-addr)# port-range 20000 30000	If you do not specify the port range, the default port range values of 40000 to 65535 is applied.
Step 5	<pre>adjacency h248 {core core-adjacency name}</pre>	Configures the H.248 core adjacency and enters into the core adjacency submode.
	Example: Router(config-h248-bac)# adjacency h248 core <i>core1</i>	Note Multiple core adjacencies and access adjacencies can be configured on the BAC. Always configure the core adjacency before configuring its corresponding access adjacency.

	Command or Action	Purpose
Step 6	<pre>control-address ipv4 ipv4-address {port port number} {port-range minimum-port number maximum-port number}</pre>	Configures a local IPv4 H.248 signaling address of the BAC.
	Example: Router(config-h248-bac-adj)# control-address ipv4 192.168.102.222 port-range 2944 4000	Note The BAC handles two types of Message Identifiers (MIDs): domain name and IP address. If the MID of an IAD is IP address, only the port-range is configured and not the port .
Step 7	<pre>remote-address ipv4 ipv4-address port port number</pre>	Configures a remote IPv4 H.248 signaling address of the MGCF and the AGCF.
	Example: Router(config-h248-bac-adj)# remote-address ipv4 192.168.102.14 port 2944	
Step 8	<pre>realm realm-number</pre>	Configures an adjacency with the IP realm that belongs to the BAC.
	Example: Router(config-h248-bac-adj)# realm 1	A realm group can contain multiple media addresses. When you configure a realm group under an adjacency, the IP address and port for the media stream of this adjacency is allocated from the media addresses in this realm group.
Step 9	attach	Sets the BAC adjacency state to Attached.
	Example: Router(config-h248-bac-adj)# attach	
Step 10	Exit	Exits from the core adjacency submode.
Step 11	adjacency h248 {access access-adjacency name}	Configures the H.248 access adjacency and enters the access adjacency submode.
	Example: Router(config-h248-bac)# adjacency h248 access <i>acc1</i>	Note Always configure the access adjacency after configuring its corresponding core adjacency.
Step 12	<pre>control-address ipv4 ipv4-address {port port number}</pre>	Configures a local IPv4 H.248 signaling address of the BAC.
	Example: Router(config-h248-bac-adj)# control-address ipv4 172.16.104.14 port 2940	
Step 13	<pre>audit {force interval idle time}</pre>	Changes the audit interval in the BAC. The default value is 1 minute.
	<pre>Example: Router(config-h248-bac-adj)# audit interval 300</pre>	
Step 14	heart-beat terminate terminate-interval	Configures the time interval during which only one heartbeat request from the H.248 terminal device can pass
	Example: Router(config-h248-bac-adj)# heart-beat terminate 0	during this interval are terminated.

	Command or Action	Purpose
Step 15	domain-name domain-name	Specifies the domain name of the BAC adjacency that replaces the domain name of the AGCF and the MGCF.
	Example:	
	Router(config-h248-bac-adj)# domain-name cisco	
Step 16	core-adj core adjacency-name	Binds the BAC core adjacency with its corresponding BAC access adjacency.
	Example:	
	Router(config-h248-bac-adj)# core-adj	
Step 17	realm realm-number	Configures an adjacency with the IP realm that belongs to the BAC.
	Example:	A realm group can contain multiple media addresses. When
	Router(config-h248-bac-adj)# realm 1	you configure a realm group under an adjacency, the IP address and port for media stream of this adjacency is allocated from the media addresses in this realm group.
Step 18	attach	Sets the BAC adjacency state to Attached.
	Example:	
	Router(config-h248-bac-adj)# attach	

Configuration Example for H.248 BAC Support

The following example shows how to configure the H.248 BAC Support feature:

```
sbc h248 bac
media-address ipv4 8.8.8.8 realm 1
port-range 20000 30000
media-address ipv4 9.9.9.9 realm 2
port-range 40000 50000
adjacency h248 core core1
   control-address ipv4 192.168.102.222 port-range 2944 4000
   remote-address ipv4 192.168.102.14 port 2944
   realm 1
   attach
adjacency h248 access acc1
   control-address ipv4 172.16.104.14 port 2940
   audit-interval 300
   heart-beat terminate 0
   domain-name cisco
   core-adj core1
   realm 2
   attach
sbc sbc
sbe
activate
```







End-to-End Cisco Unified Border Element (SP Edition) Configuration Example

This section contains a complete Cisco Unified Border Element (SP Edition) configuration on the Cisco ASR 1000 Series Routers.

```
Router# show run
Building configuration...
```

```
Current configuration : 17580 bytes
1
! Last configuration change at 11:12:56 SGT Sun Nov 21 2010
1
version 15.1
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
service internal
no platform punt-keepalive disable-kernel-core
platform shell
hostname ASR1002-2
1
boot-start-marker
boot system
bootflash:asr1000rp1-adventerprisek9.BLD_V151_1_S_XE32_THROTTLE_LATEST_20101109_090050.bin
boot system bootflash:asr1000rp1-adventerprisek9.BLD_MCP_DEV_LATEST_20101109_222533.bin
boot-end-marker
Т
T
vrf definition Mgmt-intf
 1
 address-family ipv4
 exit-address-family
 1
address-family ipv6
 exit-address-family
1
vrf definition h323-vrf-a
 description h323-vrf-a
 T
address-family ipv4
 exit-address-family
1
vrf definition h323-vrf-b
 description h323-vrf-b
 !
 address-family ipv4
```

```
exit-address-family
1
vrf definition l2e-vrf-a
description VRF a-side for late-to-early
 1
address-family ipv4
exit-address-family
1
vrf definition l2e-vrf-b
 description VFR b-side for late-to-early
 !
address-family ipv4
exit-address-family
!
vrf definition sigpinhole_customer_a
description SigPinhole-VRF-Customer-A
 1
address-family ipv4
exit-address-family
!
vrf definition sigpinhole_customer_b
description SigPinhole-VRF-Customer-B
 !
address-family ipv4
exit-address-family
1
vrf definition vrf_a
 !
address-family ipv4
 exit-address-family
 1
address-family ipv6
exit-address-family
1
vrf definition vrf_b
 1
address-family ipv4
exit-address-family
 1
address-family ipv6
 exit-address-family
!
logging buffered 1000000
enable secret 5 $1$wVYL$r.SbA2ka.619g7baSdHJx/
!
no aaa new-model
!
!
1
no process cpu extended history
no process cpu autoprofile hog
clock timezone SGT 8 0
ip source-route
1
1
1
ip domain name cisco.com
ip host t-mobile.com 10.0.48.236
ip host ibcf.t-mobile.com 10.0.48.236
ip host scscf.t-mobile.com 10.0.48.236
ip name-server 20.21.28.125
ip name-server vrf vrf_a 20.21.28.125
```

```
ip name-server vrf vrf_b 20.21.28.125
!
1
ipv6 host opensips.cisco.com 2001:20:21:28:20:21:28:93
ipv6 unicast-routing
1
1
!
multilink bundle-name authenticated
1
1
T
I
I.
1
!
!
redundancy
mode none
 application redundancy
  group 1
  name CUBE-SP
   shutdown
   priority 255 failover threshold 100
   control GigabitEthernet0/0/1.726 protocol 1
   data GigabitEthernet0/0/2
  protocol 1
   authentication md5 key-string cisco
!
!
1
ip ftp username fw
ip ftp password cisco
1
class-map type inspect match-any sip-traffic-class
match protocol sip
match protocol icmp
1
policy-map type inspect private-public-policy
class type inspect sip-traffic-class
 inspect
class class-default
1
zone security private
zone security public
zone-pair security private-public source private destination public
service-policy type inspect private-public-policy
!
1
I.
I.
1
!
interface SBC1
ip address 10.160.90.4 255.255.255.0 secondary
ip address 10.160.90.11 255.255.255.0 secondary
ip address 10.160.90.12 255.255.255.0 secondary
 ip address 10.160.90.13 255.255.255.0 secondary
 ip address 10.160.90.14 255.255.255.0 secondary
 ip address 10.160.90.15 255.255.255.0 secondary
```

```
ip address 10.160.90.16 255.255.255.0 secondary
 ip address 10.160.90.17 255.255.255.0 secondary
 ip address 10.160.90.18 255.255.255.0 secondary
ip address 10.160.90.19 255.255.255.0 secondary
 ip address 10.160.90.3 255.255.255.0 secondary
 ip address 20.24.34.1 255.255.255.0
ipv6 address 2001:A401::10:160:90:1/64
ipv6 address 2001:A401::10:160:90:2/64
ipv6 address 2001:A405::20:24:34:1/64
!
interface SBC2
ip address 10.190.6.2 255.255.255.224 secondary
ip address 10.190.6.1 255.255.255.224
!
interface SBC3
ip address 10.190.6.34 255.255.255.224 secondary
ip address 10.190.6.33 255.255.254
L.
interface SBC4
 ip address 10.190.7.66 255.255.255.224 secondary
ip address 10.190.7.65 255.255.254
T.
interface SBC5
ip address 10.190.7.98 255.255.255.224 secondary
ip address 10.190.7.97 255.255.255.224
1
interface SBC9
ip address 9.1.1.1 255.255.255.0
!
interface SBC200
ip address 20.24.31.1 255.255.255.0
ipv6 address 2001:20:24:31:20:24:31:1/64
!
interface SBC749
ip address 20.24.49.1 255.255.255.0
1
interface GigabitEthernet0/0/0
ip address 1.1.1.2 255.255.255.0
 zone-member security private
negotiation auto
cdp enable
redundancy rii 10
!
interface GigabitEthernet0/0/1
no ip address
shutdown
negotiation auto
cdp enable
1
interface GigabitEthernet0/0/1.726
encapsulation dot1Q 726
ip address 20.21.26.120 255.255.255.0
L.
interface GigabitEthernet0/0/2
ip address 1.1.2.2 255.255.255.0
zone-member security public
negotiation auto
I.
interface GigabitEthernet0/0/3
no ip address
shutdown
negotiation auto
1
interface FastEthernet0/1/0
```
```
ip address 20.21.47.16 255.255.255.0 secondary
 ip address 20.21.47.13 255.255.255.0
 speed 100
negotiation auto
!
interface FastEthernet0/1/1
no ip address
 shutdown
 speed 100
negotiation auto
1
interface FastEthernet0/1/2
no ip address
shutdown
 speed 100
negotiation auto
1
interface FastEthernet0/1/3
no ip address
 shutdown
 speed 100
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
 ip address 10.74.48.165 255.255.254
negotiation auto
!
!
no ip http server
no ip http secure-server
ip route 10.74.48.151 255.255.255.255 20.21.26.1
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.74.48.161
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.74.28.65
ip route vrf vrf_a 0.0.0.0 0.0.0.0 20.21.27.1
ip route vrf vrf_b 0.0.0.0 0.0.0.0 20.21.26.1
1
logging esm config
cdp run
ipv6 route ::/0 2001:20:21:28:20:21:28:1
ipv6 route vrf vrf_b ::/0 2001:20:21:26:20:21:26:1
ipv6 route vrf vrf_a ::/0 2001:20:21:27:20:21:27:1
1
!
1
control-plane
1
!
sbc diagnostics sparse
1
1
sbc rls8
sbe
   control address aaa ipv4 20.24.34.1
   radius authentication
    server freeRadius
     address ipv4 10.0.48.236
     mode local
     key cisco
   radius accounting Codenomicon
    concurrent-requests 4000
    retry-interval 5000
    retry-limit 9
```

server Codenomicon

```
address ipv4 10.0.48.236
    port 1812
    key cisco
  sip body-profile PASSALL
  sip parameter-profile test
   parameter aaa
    action strip
  sip parameter-profile testb
  sip parameter-profile proxy-param
   parameter firewall
    action strip
  sip parameter-profile access-param
   parameter firewall
    action add-or-replace value public-ip-address
  sip header-profile h1
   src-address
    header-prio 1 header-name P-Called-Party-ID
    header-prio 2 header-name P-Preferred-Identity
   header Allow entry 1
    action pass
   header Call-Info entry 1
    action pass
   header P-Asserted-Identity entry 1
    action pass
  sip header-profile 111
   header Allow entry 1
    action replace-value value "ddd"
  sip header-profile p-kt
   header P-KT-UE-IP entry 1
    action strip
   header P-KT-UE-IP entry 2
    action add-header value "${msg.rmt_ip_addr}"
  sip header-profile proxy
   header contact entry 1
    parameter-profile proxy-param
    action as-profile
  sip header-profile access
   header contact entry 1
    parameter-profile access-param
    action as-profile
  sip header-profile default
   blacklist
  sip header-profile IMS_Access
   blacklist
   header P-Called-Party-ID entry 1
    action strip
  sip header-profile P-Charging-Fucntion-Address
   blacklist
   header P-Charging-Function-Addresses entry 1
    action add-first-header value "1.1.1.1"
  sip method-profile PASS
   blacklist
  sip method-profile default
   blacklist
  sip option-profile default
   blacklist
  sip error-profile default
   cause rtg-no-route-found sub-cause rtg-src-adjacency status-code 604 reason "Q.850
;cause=16 ;text=\"SBC: No route found based on src adjacency\""
  adjacency h323 H323CCM134-GK
   signaling-address ipv4 20.24.34.1
   signaling-port 1719
   remote-address ipv4 10.0.50.134 255.255.255.255
   signaling-peer gk 10.0.48.93
```

```
tech-prefix 567
 dbe-location-id 0
 allow private info
 trunk trusted
inbound secure
attach
adjacency h323 H323CCM134-vrfa
vrf h323-vrf-a
 signaling-address ipv4 10.190.7.65
 remote-address ipv4 10.0.50.134 255.255.255.255
 signaling-peer 10.0.50.134
 dbe-location-id 0
trunk trusted
inbound secure
attach
adjacency sip SIPP1
signaling-address ipv4 20.24.34.1
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.244.81 255.255.255.255
 signaling-peer 10.0.244.81
dbe-location-id 0
attach
adjacency sip SIPP2
 signaling-address ipv4 20.24.34.1
 statistics method summary
 signaling-port 5060
 remote-address ipv4 10.0.244.82 255.255.255.255
 signaling-peer 10.0.244.82
 dbe-location-id 0
attach
adjacency sip UE-RX
 inherit profile preset-access
 signaling-address ipv4 192.168.2.1
 statistics method summary
 remote-address ipv4 10.0.120.19 255.255.255.255
 signaling-peer 10.0.120.19
 dbe-location-id 0
 reg-min-expiry 200
 fast-register disable
attach
adjacency sip adj1-o
 inherit profile preset-access
visited network identifier ims.net
 signaling-address ipv4 192.168.2.1
 statistics method summary
 remote-address ipv4 192.168.1.1 255.255.255.255
 signaling-peer 192.168.1.1
media bypass tag 1 a
media bypass tag 2 b
media bypass tag 3 c
media bypass tag 4 d
attach
adjacency sip adj1-t
inherit profile preset-access
visited network identifier ims.net
 signaling-address ipv4 192.168.130.1
 statistics method summary
 remote-address ipv4 192.168.129.1 255.255.255.255
 signaling-peer 192.168.129.1
media bypass tag 1 a
media bypass tag 2 b
media bypass tag 3 c
media bypass tag 4 d
```

```
attach
adjacency sip CCM-132
preferred-transport tcp
signaling-address ipv4 20.24.34.1
statistics method summary
 signaling-port 5060
remote-address ipv4 10.0.50.132 255.255.255.255
 signaling-peer 10.0.50.132
 dbe-location-id 0
ping-enable
 ping-suppression ood-request
 ping-bad-rsp-codes 503
warrant match-order destination source diverted-by
attach
adjacency sip CCM-133
admin-domain ad1
vrf sigpinhole_customer_a
 signaling-address ipv4 10.190.6.33
 statistics method summary
 signaling-port 5060
remote-address ipv4 10.0.50.133 255.255.255.255
signaling-peer 10.0.50.133
dbe-location-id 0
 dtmf disable sip notify
attach
adjacency sip CCM-135
 admin-domain ad1
signaling-address ipv4 20.24.34.1
 statistics method summary
signaling-port 5060
remote-address ipv4 10.0.50.135 255.255.255.255
 signaling-peer 10.0.50.135
 dbe-location-id 0
dtmf disable sip info
attach
adjacency sip OpensipsV6
group IPv6
nat force-off
 inherit profile preset-core
 signaling-address ipv6 2001:A401::10:160:90:1
 statistics method summary
signaling-port 7060
remote-address ipv6 2001::216:ECFF:FE3B:40DD/128
 signaling-peer 2001:A401::33:33:36:1
 dbe-location-id 0
 registration target address 2001:A401::33:33:36:2
header-name From passthrough
dtmf prefer sip info
attach
adjacency sip OpenIMSCore
inherit profile preset-core
signaling-address ipv4 20.24.34.1
 statistics method summary
 signaling-port 4060
remote-address ipv4 10.0.48.236 255.255.255.255
 signaling-peer 10.0.48.236
 dbe-location-id 0
 registration target address open-ims.test
 registration monitor
header-name From passthrough
 ims pani e2
attach
adjacency sip SoftphoneV6
 group IPv6
```

```
nat force-on
 inherit profile preset-access
 signaling-address ipv6 2001:A401::10:160:90:1
 statistics method summary
signaling-port 5060
remote-address ipv6 2001::/64
 signaling-peer 2001::10:0:120:19
 dbe-location-id 0
registration rewrite-register
attach
cac-policy-set 1
first-cac-table SRC-ADJ
 first-cac-scope src-adjacency
 cac-table SRC-ADJ
  table-type limit src-adjacency
 entry 1
  match-value UE-RX
  caller inband-dtmf-mode always
  media police strip
  action cac-complete
  entry 2
  match-value CCM-132
  codec-preference-list pref-list1
  callee-privacy privacy-service always
  caller-privacy privacy-service never
  srtp support allow
  payload-type asymmetric allowed
   callee local-call-transfer allowed
   srtp caller forbid
  srtp callee mandate
  srtp interworking allow
  media police strip
  action cac-complete
  entry 3
  match-value CCM-133
  media police strip
   action next-table msmbtb1
 cac-table msmbtb1
  table-type policy-set
  entry 1
  media bypass type hairpin full
  media police strip
  action cac-complete
complete
cac-policy-set global 1
call-policy-set 1
 first-inbound-na-table natable1
 first-call-routing-table da1
 first-reg-routing-table REG-ROUTE-ON-SRC-ADJ
rtg-dst-address-table da1
 entry 1
  match-address kate string
  dst-adjacency CCM-135
  action complete
  entry 2
  match-address bob string
   dst-adjacency CCM-133
  action complete
  entry 3
  match-address 44 digits
  dst-adjacency CCM-135
  action complete
  prefix
  entry 4
```

```
match-address 86 digits
   dst-adjacency OpenIMSCore
   action complete
   prefix
 rtg-src-adjacency-table REG-ROUTE-ON-SRC-ADJ
  entry 1
   match-adjacency UE-RX
   dst-adjacency OpenIMSCore
   action complete
  entry 2
   match-adjacency SoftphoneV6
   dst-adjacency OpensipsV6
   action complete
  entry 3
   match-adjacency OpenIMSCore
   dst-adjacency adj1-o
   action complete
na-dst-address-table natable1
  entry 1
   action next-table privacytb1
   edit-src add-prefix 1
  match-address 111 digits
  entry 2
   action accept
   edit-src add-prefix 12345
   match-address 112 digits
  entry 3
   action accept
   edit-src add-prefix abc
   match-address 113 digits
  entry 4
   action accept
  match-address ^201[a-d]ef regex
  entry 5
   action accept
 na-src-name-anonymous-table privacytb1
  entry 1
   action accept
   edit-dst add-prefix 3
   match-anonymous true
 complete
call-policy-set 2
 first-call-routing-table ROUTE-ON-DEST-NUM
 rtg-dst-address-table ROUTE-ON-DEST-NUM
  entry 1
   match-address 1320X digits
   dst-adjacency CCM-132
   action complete
   edit-dst del-prefix 4
   prefix
 complete
call-policy-set 3
 first-call-routing-table table1
 rtg-src-adjacency-table table1
  entry 1
   match-adjacency SIPP1
   dst-adjacency CCM-135
   action complete
  entry 2
   match-adjacency SIPP2
   dst-adjacency CCM-133
   action complete
 complete
call-policy-set default 1
```

```
admin-domain adl
  description This is a description for DOMAIN1
  call-policy-set inbound-na 3
  call-policy-set rtg 3
   ! using call-policy-set outbound-na default
  admin-domain ad2
  description This is a description for DOMAIN2
  call-policy-set inbound-na 2
  call-policy-set rtg 2
  call-policy-set outbound-na 2
  enum 1
  reg-timeout 60
  rsp-lifetime 34000
  nmr-buf-pool-size 500
   entry default
   server ipv4 10.0.120.33
  activate
 network-id 29599
  sip dns
  support-type sip-dns-srv
  cache lifetime 0
  cache limit 10
1
  codec list pref-list1
  codec G723 priority 1
  codec PCMU priority 2
1
  codec variant codec G7231L
   variant G7231L
   standard G723
   fmtp annexa=ves
    fmtp bitrate=5.3
 billing
  local-address ipv4 20.24.34.1
  ldr-check 23 30
  method packetcable-em
  method xml
  packetcable-em 0 transport radius Codenomicon
   local-address ipv4 20.24.34.1
   attach
  xml 1
   cdr path usb0:Billing/
   cdr alarm minor 500000
   ldr-check 23 30
   attach
   activate
1
Т
 blacklist global
  reason bad-address
   trigger-size 65535
  reason cac-policy-rejection
   trigger-size 65535
  reason spam
   trigger-size 65535
 blacklist vpn sigpinhole_customer_a
  reason authentication-failure
   trigger-size 65535
   reason endpoint-registration
   trigger-size 65535
   trigger-period 1 seconds
   reason cac-policy-rejection
   reason corrupt-message
```

```
trigger-size 65535
     trigger-period 1 seconds
   blacklist global ipv6 2001::10:0:233:113
    reason authentication-failure
     trigger-size 65535
     trigger-period 1 seconds
    reason bad-address
     trigger-size 65535
     trigger-period 1 seconds
    reason endpoint-registration
     trigger-size 65535
    trigger-period 1 seconds
    reason cac-policy-rejection
     trigger-size 65535
     trigger-period 1 seconds
    reason corrupt-message
     trigger-size 65535
     trigger-period 1 seconds
    reason spam
     trigger-size 65535
     trigger-period 1 seconds
 1
rtp-flood-detect
media-address ipv4 10.160.90.3
 port-range 10000 11000 voice tag CCM-132
 port-range 11001 12000 video tag CCM-135
media-address ipv6 2001:A401::10:160:90:1
 port-range 16384 32767 signaling
 activate
T
!
line con 0
exec-timeout 0 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
 exec-timeout 0 0
no login
!
exception data-corruption buffer truncate
!
monitor session 22 type erspan-source
description SOURCE_SESSION_FOR_Gi0/0/0
source interface Gi0/0/0
destination
 erspan-id 22
 ip address 10.0.100.100
  origin ip address 20.21.28.72
!
!
end
```



SIP Compliance and Interoperability

This appendix lists examples of Session Initiation Protocol (SIP) services and features and the type of support provided by Cisco Unified Border Element (SP Edition).

SIP Features and SBC Compliance

The following table lists some examples of SIP services and features and extent of interoperability and compliance provided by Cisco Unified Border Element (SP Edition) on the unified model.

Many of the SIP services or features in the table are listed in the draft-ietf-sipping-service-examples.txt specification.

The table covers most features offered in what are considered IP Centrex offerings from local exchange carriers and PBX (Private Branch Exchange) features. The table also includes services involving some extensions to SIP, including the REFER, SUBSCRIBE, and NOTIFY methods and the Replaces and Join headers.

SIP Service or Feature	Cisco Unified Border Element (SP Edition) Compliance or Support
Call Hold	Supported.
Consultation Hold	Supported.
Music on Hold	Supported.
Call Hold with Music on Hold	Supported.
Find-Me	Passthrough only. SBC does not perform find-me function.
Call Park	Passthrough only. SBC does not perform call park function.
Call Forking	Supported.
Caller-ID	Passthrough only. SBC does not perform caller-ID function.
Calling Name Delivery	Passthrough only. SBC does not perform calling name delivery function.
Click to Dial	Partial (not supported with SBC between end user computer and phone).

 Table 1-1
 SBC Compliance and Support of SIP Services and Features

SIP Service or Feature	Cisco Unified Border Element (SP Edition) Compliance or Support
Message Waiting Indicator	Passthrough, depending on SIP SUBSCRIBER/NOTIFY messages
Call Forwarding - Busy	Partial (not supported with SBC between proxy and callee).
Call Forwarding - No Answer	Partial (not supported with SBC between proxy and callee).
Call Forwarding - Unconditional	Supported.
SIP Session Refreshment with re-INVITE	
SIP - Specific Event Notification	Partial support (RFC 3265).
Transfer - Unattended	Supported.
Transfer - Attended	Supported.
Transfer - Instant Messaging	Supported.
3-way Conference - Third Party is Added	Not supported.
3-way Conference - Third Party Joins	Not supported.
Single Line Extension	Not supported.
Call Management (Incoming Call Screening)	Passthrough only. SBC does not perform Call Screening function.
Call Management (Outgoing Call Screening)	Passthrough only. SBC does not perform Call Screening function.
Call Pickup	Not supported.
Automatic Redial	Not supported.

Table 1-1 SBC Compliance and Support of SIP Services and Features (continued)



XML Billing Schema

This appendix provides a detailed description of the XML elements used in the XML billing records that the CUBE (SP) XML billing method generates, an XML billing sample file generated by the SBC, the termination codes for the XML billing records, and the XML Document Type Definition (DTD).

XML Elements Generated by CUBE (SP)

This section provides details of the XML elements used in the XML billing records that the CUBE (SP) XML billing method generates.

The recordfile Element

Table 1-1 shows the attribute in the recordfile element.

Table 1-1 Attributes of the recordfile Element

Attribute	Optional	Description
sbc	Ν	IP address of a CUBE (SP) recording.

The call Element

Table 1-2 shows the attributes in the call element.

Table 1-2	Attributes of the call Element
Table 1-2	Attributes of the call Element

Attribute	Optional	Description
starttime	N	The time at which a call starts is the time at which signaling starts.
endtime	Y	The time at which a call ends is the time at which signaling ends and resources are released. This attribute is present if the call does not end, when the call detail record (CDR) is written. This is because the billing method instance is deactivated when the billingdeactivation element is present.

Attribute	Optional	Description
duration	Y	The length of the call, in milliseconds. This attribute is present if the call does not end, when the call detail record (CDR) is written. This is because the billing method instance is deactivated when the Billingdeactivation element is present.
release_side	Y	Indicates the side of the call control that initiates the call release, if any.
		The applicable values are as follows:
		• Orig—Indicates that the release is initiated on the originating side of the call.
		• Term—Indicates that the release is initiated on the terminating side of the call.
		• Neither—Indicates that the release is not initiated on either side of the call.
		Omission of this attribute indicates that no information is available about the release initiator. One possible reason for this could be that the call is not yet released.
bcid	N	A unique identifier of the CUBE (SP) instance pertaining to a call record.

Table 1-2	Attributes	of the call	Element	(continued)
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The subscriber Element

Table 1-3 shows the attribute in the subscriber element.

Table 1-3	Attributes	of the	subscriber	Element

Attribute	Optional	Description
public_id	Ν	The public identifier of the subscriber.

The billingdeactivation Element

Table 1-4 shows the attribute in the billingdeactivation element.

 Table 1-4
 Attributes of the billingdeactivation Element

Attribute	Optional	Description
time	Ν	The time at which the billing instance is deactivated.

The party Element

Attribute	Optional	Description
type	N	The applicable values are:
		• Orig—Indicates that this party is the originating endpoint of the call.
		• Term—Indicates that this party is the terminating endpoint of the call.
phone	Ν	The original phone number or the SIP user name of the party.
domain	Y	The original domain name of the phone number or the SIP user name.
cic	Y	The carrier identification code of the phone number or the SIP user name. This attribute is present only at the terminating endpoint.
editphone	Y	The edited phone number or the SIP user name of the party.
editcic	Y	The edited carrier identification code of the phone number or the SIP user name. This attribute is present only at the terminating endpoint.
sig_address	Y	The network address of the next-hop signaling entity. The signaling messages are received from this network address and are sent to this network address.
sig_port	Y	The network port of the next-hop signaling entity. The signaling messages are received from this network port and are sent to this network port.

Table 1-5 shows the attributes in the party element.

Table 1-5Attributes of the party Element

The adjacency Element

Table 1-6 shows the attributes in the adjacency element.

Table 1-6Attributes of the adjacency Element

Attribute	Optional	Description
type	N	The applicable values are:
		• Orig—Indicates that this adjacency is the originating adjacency of the call.
		• Term—Indicates that this adjacency is the outgoing adjacency of the call.
name	Ν	The adjacency name, as configured by the administrator on the SBC.
account	N	The account name to which the originating branch or terminating branch of the call belong to, as configured by the administrator on the SBC.

Table 1-6	Attributes of the adjacency Element	
-----------	-------------------------------------	--

Attribute	Optional	Description
vpn	Y	The VPN ID associated with the adjacency, if any.
mediarealm	Y	The IP realm associated with the adjacency, if any.

The connect Element

Table 1-7 shows the attribute in the connect element.

```
        Table 1-7
        Attributes of the connect Element
```

Attribute	Optional	Description
time	N	The time at which the call is connected, that is, when the media gate is opened.

The firstendrequest Element

Table 1-8 shows the attribute in the firstendrequest element.

```
        Table 1-8
        Attributes of the firstendrequest Element
```

Attribute	Optional	Description
time	Ν	The time at which the first BYE request is received.

The disconnect Element

Table 1-9 shows the attribute in the disconnect element.

Table 1-9 Attributes of the disconnect Element

Attribute	Optional	Description
time	N	The time at which the call is disconnected, that is, when the final BYE response is received.
reason	N	The reason for the disconnection. For more information about the various reasons for call termination, see Table 1-17.

The release Element

Table 1-10 shows the attribute in the release element.

Table 1-10Attributes of the release Element

Attribute	Optional	Description
reason	Ν	The reason for not connecting to the call. For more information about the various reasons for call termination, see Table 1-17.

The im_stats Element

Table 1-11 shows the attributes in the im_stats element.

 Table 1-11
 Attributes of the im_stats Element

Attribute	Optional	Description
incomplete	Y	Indicates whether the msgs_from_orig and msgs_from_term statistics attributes are applicable to the entire call. This element is omitted when the value is false.
msgs_from_orig	Ν	The number of IM messages sent from the caller.
msgs_from_term	Ν	The number of IM messages sent from the callee.

The QoS Element

Table 1-12 shows the attributes in the Quality of Service (QoS) element.

Table 1-12 Attributes of the QoS Elei	nent
---------------------------------------	------

Attribute	Optional	Description
reservetime	Y	The time at which the QoS is reserved.
committime	Y	The time at which the QoS is committed. This information is mandatory if the QoS is committed.
releasetime	N	The time at which the QoS is released. This field value may be inaccurate in certain post-failover scenarios such as RP switch over.

The gate Element

The gate element contains no attributes.

The flowinfo Element

Table 1-13 shows the attribute in the flowinfo element.

Table 1-13 Attributes of the flowinfo Element

Attribute	Optional	Description
transport_type	Y	This attribute can have the following values:
		• RTP—This indicates that the media stream is using real-time transport protocol (RTP). The RTP is the default value used, if the transport_type attribute is absent from the flowinfo element.
		• SRTP—This indicates that the media stream is using secure real-time transport protocol (SRTP).
		• UDPTL—This indicates that the media stream is carrying T.38 over user datagram protocol transport layer (UDPTL).

The local Element and the remote Element

Table 1-14 shows the attribute in the local element and the remote element.

 Table 1-14
 Attributes of the local Element and the remote Element

Attribute	Optional	Description
address	N	The IP address that sends and receives packets.
port	N	The port number that packets are sent from and received on.
transrated	Y	Indicates whether the media packets sent to this element are transrated or not.
		If this attribute is absent, transrating is not provisioned. In the current implementation of the SBC, this attribute appears only in the remote element, because transrating is always performed as late as possible.

The sd Element

Table 1-15 shows the attribute in the sd element.

Table 1-15	Attributes of the sd Element

Attribute	Optional	Description
direction	Y	This attribute can have the following values:
		• Inbound – Indicates that the element provides inbound SDP information.
		• Outbound – Indicates that the element provides outbound SDP information.
		If this attribute is not included, it implies that the negotiated SDP is symmetric.

The RTCPStats Element

The RTCPStats element contains no attributes.

The admin_domains Element

The admin_domains element contains no attributes.

The ad Element

Table 1-16 shows t	the attribute	in the ad	element.
--------------------	---------------	-----------	----------

Attribute	Optional	Description
Name	Ν	The name of the admin domain associated with the endpoint.

Sample XML Billing File

The following is an example of an XML billing file.

```
<?xml version="1.0"?><recordfile sbc-sig="20.24.34.1"><call starttime="1277766440306"</pre>
endtime="1277766552984" duration="112678" release_side="orig"
bcid="4C292B28202020203830339302B3038303030300000004"><party type="orig" phone="2013"
domain="10.0.50.135" sig_address="10.0.50.135" sig_port="58790"/><party type="term"
phone="13208011" editphone="8011" sig_address="10.0.50.132" sig_port="5060"/><adjacency
type="orig" name="CCM-135" account="" mediarealm = "sgn1"/><adjacency type="term"
name="CCM-132" account="" mediarealm = "sgn1"/><connect</pre>
time="1277766442516"/><firstendrequest time="1277766552976"/><disconnect
time="1277766552984" reason="0"/><QoS stream_id="1" instance="0"
reservetime="1277766440306" committime="1277766442516"
releasetime="1277766552987"><gate><flowinfo><local address="20.21.4.3"
port="16388"/><remote address="10.0.50.135" port="26880"/><sd>m=audio 0 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0, PC/RJI=0,
PC/RLA=0</RTCPstats></flowinfo><flowinfo><local address="20.21.4.3" port="16390"/><remote
address="10.0.50.132" port="24580"/><sd>m=audio 0 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
</sd></re>
PC/RPS=0, PC/ROS=0, PC/RPR=0, PC/RPL=0, PC/RJI=0,
PC/RLA=0</RTCPstats></flowinfo></gate></QoS></call></recordfile>
```

Termination Codes

The following table contains the codes that describe the causes for call terminations in XML billing records.

Value	Description
00	Normal call termination (no error).
01	A storage resource shortage has occurred on the local device.
02	A storage resource shortage has occurred on a remote device controlled by the local device.
03	A media resource shortage has occurred.

Table 1-17 Termination Codes for XML Billing Records

Value	Description
04	A media failure has occurred because of the failure in the underlying hardware or through management action.
05	A continuity test has failed.
06	The requested media resource is blocked or has been quiesced.
07	Media is in use by another call.
08	Media is not configured.
09	An error has occurred due to a configuration inconsistency.
10	Media is unavailable.
11	Media is congested.
12	An internal error has occurred.
13	No terminations are available.
14	An error other than a failure, resource, or bandwidth shortage has occurred in the media layers.
15	A request to reset a termination has failed.
16	An interworking error has occurred.
17	A security error has occurred.
18	This is not a valid address.
19	This is not a valid transit network.
20	There is no route available to the specified destination address.
21	There is no route available to the specified transit network.
22	This number is unavailable because the number has changed recently.
23	This is an unallocated number.
24	There is no route-to-destination address due to congestion.
25	There is no route-to-transit network due to congestion.
26	LNP call is misrouted to the exchange that does not serve the destination number.
27	Internal congestion has occurred.
28	The media capabilities requested for the call are not supported.
29	The maximum number of routing retries are exceeded.
30	The resources are unavailable for SBC.
31	The destination resource is incompatible with request.
32	This is an invalid message.
33	This is an unrecognized signaling message type.
34	Recovery on timer expiry.
35	Unrecognized or unimplemented signaling parameter has been passed on.
36	Unrecognized or unimplemented signaling parameter has been discarded.
37	The signaling protocol error has occurred.
38	This is a temporary failure.

 Table 1-17
 Termination Codes for XML Billing Records (continued)

Value	Description
39	No answer.
40	The destination is out of order.
43	Unauthorized request.
44	Network congestion.
45	The request is not supported for an unspecified reason.
46	The specified resource is not equipped.
47	Call to call services.
48	An unspecified or miscellaneous error has occurred.
49	The named digit map requested by the call agent is unknown to the media gateway.
50	The media bandwidth is insufficient.
51	The routing has failed because no digits were dialed.
52	A subscriber has attempted to dial a number that is restricted.
53	QoR call to a subscriber has failed because the subscriber was not found.
55	Called user has rejected the call.
56	The call could not be routed to a subscriber because the subscriber's termination could not be located.
57	Called subscriber is busy even though media can be allocated to the subscriber.
64	A branch that was successfully audited internally following a Call Agent failover does not indicate that the call failed during the Call Agent failover.
65	A subscriber attempted to register for an interval that was too brief.
66	This request is unauthorized by proxy.
67	The call's early media exceeded the time limit set by access control before the call was connected.
68	A glare scenario was detected, where each party in the call sent a message of the same type simultaneously.
69	An endpoint has attempted a renegotiation at an illegal point.
70	An endpoint has sent media parameters that were unparseable.
71	A message or one of its subcomponents was too large to process.
72	An endpoint indicated that a request must be redirected.
73	(CAC-specific) Call setup rate have exceeded a maximum limit.
74	(CAC-specific) Number of call updates have exceeded the maximum limit.
75	(CAC-specific) Number of calls have exceeded the maximum limit.
76	(CAC-specific) Number of media channels used have exceeded the maximum limit.
77	(CAC-specific) Bandwidth used have exceeded the maximum limit.
78	(CAC-specific) Number of registered endpoints have exceeded the maximum limit.
79	(CAC-specific) Rate of endpoint registrations have exceeded the maximum limit.

 Table 1-17
 Termination Codes for XML Billing Records (continued)

Value	Description
80	Media could not be established because an acceptable media transport type could not be negotiated for any media stream.
81	Media is not yet established because of redirection. The system is retrying.
82	No subscriber record with the specified search keys is found.
83	(CAC-specific) Rate of in-call messages have exceeded the maximum limit.
84	(CAC-specific) Rate of out-of-call requests have exceeded the maximum limit.
85	Register request from endpoint was rejected because a delegate subscriber exists in subscriber database (SUBDB) with matching search keys.
86	(CAC-specific) Media transport settings of the call caused it to fail.
87	Routing failed because the route to the address is unavailable.
88	No acceptable codec that can be used for a call.
89	The number of media channels requested is greater than the maximum number the SBC supports.
90	An attempt to transfer the call has failed. This is used when the reason for a call is released after an attempt to transfer it to a third party has failed.
91	The E.164 number mapping (ENUM) processing encountered an error.
92	The SBC received a message with SDP parameters that were unparseable.
93	A subscriber signaling bearer channel is unavailable.
94	A subscriber media bearer channel has failed mid-call.
95	A subscriber media bearer channel was rejected, either during call setup or during renegotiation.
96	Privacy requirements could not be satisfied for the call.
97	A CAC-specific error code indicating that a policy disallowing the RTP for the call caused it to fail. Note: This error code is used only in internal-to-ICC, and should not be communicated to the signaling stacks.
98	A CAC-specific error code indicating that a policy disallowing the SRTP for the call caused it to fail.
	Note: This error code is used only in internal-to-ICC, and must not be communicated to the signaling stacks.
99	Policy disallowing the RTP or the SRTP IW for the call caused it to fail.
100	No media gateway (MG) that is able to support the SRTP was found for the call, causing it to fail.
101	SRTP processing encountered a miscellaneous error.
102	Call released because media packets forwarding (MPF) has detected a fatal error.

 Table 1-17
 Termination Codes for XML Billing Records (continued)

XML Document Type Definition

This section provides the complete XML document type definition (DTD) for the XML billing records that the XML billing method produces.

```
<!DOCTYPE recordfile [
  <!ELEMENT recordfile (call | longcall | partialcall | audit)*>
  <!ATTLIST recordfile sbc CDATA #REQUIRED>
  <!ELEMENT call (subscriber, billingdeactivation, party, party, adjacency, adjacency,
connect?, disconnect?, QoS*)>
  <!ATTLIST call starttime
                             CDATA #REQUIRED
                endtime
                            CDATA #REQUIRED
                duration CDATA #REQUIRED
                release_side CDATA #IMPLIED
                             CDATA #REQUIRED>
                bcid
  <!ELEMENT subscriber EMPTY>
  <!ATTLIST subscriber public_id CDATA #REQUIRED>
  <!ELEMENT billingdeactivation EMPTY>
  <!ATTLIST billingdeactivation time CDATA #REQUIRED>
  <!ELEMENT party (admin_domains?)>
  <!ATTLIST party type
                        CDATA #REQUIRED
                             CDATA #REQUIRED
                 phone
                 domain CDATA #IMPLIED
                 cic
                             CDATA #IMPLIED
                 editphone CDATA #IMPLIED
editcic CDATA #IMPLIED
                              CDATA #IMPLIED
                 sig_address CDATA #IMPLIED
                 sig_port
                              CDATA #IMPLIED
                 trunk_group CDATA #IMPLIED
                 trunk_context CDATA #IMPLIED>
  <!ELEMENT admin domains (ad*)>
  <!ELEMENT ad EMPTY>
  <!ATTLIST adname
                              CDATA #REQUIRED>
  <!ELEMENT adjacency EMPTY>
  <!ATTLIST adjacency type CDATA #REQUIRED
                     name CDATA #REQUIRED
                     account CDATA #REQUIRED
                     vpn CDATA #IMPLIED
                     mediarealm CDATA #IMPLIED>
  <!ELEMENT connect EMPTY>
  <!ATTLIST connect time CDATA #REQUIRED>
  <!ELEMENT firstendrequest EMPTY>
  <!ATTLIST firstendrequest time CDATA #REQUIRED>
  <!ELEMENT disconnect EMPTY>
  <!ATTLIST disconnect time CDATA #REQUIRED
                      reason CDATA #REQUIRED>
  <!ATTLIST release reason CDATA #REQUIRED>
  <! ELEMENT im stats EMPTY>
  <!ATTLIST im_stats incomplete CDATA #IMPLIED
                    msgs_from_orig CDATA #REQUIRED
                    msgs_from_orig CDATA #REQUIRED>
  <!ELEMENT QoS (gate, gate*)>
  <!ATTLIST QoS reservetime CDATA #IMPLIED
               committime CDATA #IMPLIED
               releasetime CDATA #IMPLIED>
  <!ELEMENT gate (flowinfo, flowinfo)>
  <!ELEMENT flowinfo (local, remote, sd, RTCPStats)>
  <!ATTLIST flowinfo transport_type CDATA #IMPLIED>
  <!ELEMENT local EMPTY>
  <!ATTLIST local address CDATA
                                        #REOUIRED
                           CDATA
                 port
                                         #REOUIRED
                 transrated (true false) "false">
  <!ELEMENT remote EMPTY>
```

L

```
<!ATTLIST remote address CDATA
port CDATA
                                         #REQUIRED
                                        #REQUIRED
                transrated (true|false) "false">
<!ELEMENT sd (#PCDATA)>
<!ATTLIST sd direction CDATA #IMPLIED>
<!ELEMENT RTCPblock (#PCDATA)>
<!ELEMENT longcall (party, party)>
<!ATTLIST longcall starttime CDATA #REQUIRED
                  duration CDATA #REQUIRED
                  bcid
                         CDATA #REQUIRED>
<!ELEMENT partialcall (QoS)>
<!ATTLIST partialcall bcid CDATA #REQUIRED>
<!ELEMENT audit (log*)>
<!ELEMENT log (name, value)>
<!ELEMENT name (#PCDATA)>
<!ELEMENT value (#PCDATA)>
```

] >



GLOSSARY

#

- **1:1 redundancy** Mechanism to provide redundancy by ensuring that for each piece of hardware there is a backup that can take over non disruptively.
- **1:n redundancy** Mechanism to provide redundancy by ensuring that for each *n* identical pieces of hardware, there is a single backup that can take over non disruptively in the case of a single failure.

Α

AAA address	Authentication, authorization, accounting address. This is the IP address used when contacting billing or authentication servers. AAA performs user/endpoint authentication prior to forwarding a request to an upstream.
	Call Admission Control (CAC) to control DBE
	• Quality of service (QoS)
	Network Address Port Translation (NAPT) binding
	• Firewall pinhole
	• Call detail record (CDR) generation for billing
account	An account represents a service relationship with a remote organization on the SBE. Each adjacency is assigned to an account, which is used to define customer-specific Call Admission Control and routing policy configuration.
admission control policy	A set of rules on the SBE that define system and call level restrictions.
ALG	Application layer gateway. A bridge for traffic between two networks. It has knowledge of, and operates at the level of, the application generating the traffic.
В	
B2BUA	Back-to-back user agent. This is a piece of software that links together the signaling flows for two legs of a call, providing a bridge between them with local termination for each leg.

С

CAC	Call Admission Control. This is the set of actions taken by a network during the set-up phase of a call event to determine whether the event should be accepted or rejected.
call policy	An interconnected set of rules used to configure how SBC responds to new call events. It includes number analysis, routing, and CAC.
CALEA	Communications Assistance for Law Enforcement Act. Passed in 1994, CALEA requires telecommunications carriers in the United States to modify their equipment, facilities, and services to ensure that they are able to comply with authorized electronic surveillance.
CDR	Call detail record. The billing record for a phone call.
CE	See PE.
codec	Compressor/decompressor. A codec is any technology for compressing and decompressing data, typically audio or video.
control address	IP address on the SBE or DBE used for terminating the H.248 control traffic between SBE and SBE. Also used in AAA control traffic.
COPS-PR	Common Open Policy Service. This is an IETF standard, supplying network switches and hubs with policy rules to help maintain quality of service.
CORBA	Common Object Request Broker Architecture. CORBA is an architecture and specification for creating, distributing, and managing distributed program objects in a network.

D

DoS protection	Protects SBE from DoS (Denial of Service) attack.
DBE	Data border element, also known as the <i>media proxy</i> . Represents the media-handling portion (RTP, RTCP, and so on) of the SBC. There can be only one DBE per service card. However, the DBE can be partitioned into several virtual DBEs (VDBEs). The DBE supports the following services:
	• Bandwidth allocation, Call Admission Control (CAC), and Service Level Agreement (SLA) Monitoring
	• Policing, marking (DSCP), and rate limiting
	• RSVP proxy
	• Firewall (media pinholes)
	Security functions
	• NAPT traversing
	Topology hiding
	• VPN aware (VPN interconnect)
	• Quality monitoring and statistics gathering

DSP service control	Engages in the codec negotiation procedures and enforces policy on codecs being negotiated to control digital signal processor (DSP) service.
DiffServ	Differentiated services. A mechanism for marking IP traffic with different priorities.
DoS	Denial of service. A malicious attempt to overload a piece of hardware in some way.
DMZ	Demilitarized zone. This is a small subnetwork that sits between a trusted private network, such as a corporate LAN, and an untrusted public network, such as the public Internet.
F	
firewall	A system designed to protect a computer network from unauthorized access, especially through the Internet.
н	
H.248	H.248 (or Megaco) is a VoIP signaling protocol, usually used between a dumb device and a clever controller. It is similar in functionality (if not syntax) to MGCP. It is used to communicate between SBC and DBE in a distributed SBC system.
H.323	A protocol used for signaling for VoIP.
HSD	Hot software downgrade.
ı	
IAD	Integrated access device. An IAD is a one-box DSL voice and data solution equipment typically installed at the customer's site.
L	
Lawful intercept	Provides intercept-related information (IRI) and call content intercept (replication of the media streams).
load-related services (sharing and balancing)	SBE may also perform load balancing when it sends a message to multiple upstream or downstream servers.
location ID	Identifies the location of DBE within the network.
LSP	Label switched path. The name for a single traffic flow in MPLS.

Μ

media address	Pool of IP addresses on the DBE for media relay functionality. A separate pool of addresses is defined for each VPN that the DBE is attached to. All vDBEs within the DBE draw media addresses from these pools.
media bypass	An SBC function allowing media to bypass DBE and flow directly between two endpoints within the same customer network or VPN.
media transcoding device	A type of media gateway that can convert between media codec types in real time. SBEs sometimes include a combination of vDBE and a media transcoding device in the data path of a single call.
megaco	See H.248.
MGCP	Media Gateway Control Protocol. This is a VoIP signaling protocol, usually used between a dumb device and a clever controller. It is similar in functionality (if not syntax) to H.248/Megaco. It is defined in RFC 2705.
MPLS	Multiprotocol Label Switching. Protocol used for network traffic flow shaping and management.
message scrubbing for identity and address hiding	Hiding end-user identifying information and end-user IP-addresses by adding, removing, or modifying the identity and IP address information in the signaling headers.
N	
NAT	Network Address Translator. This is a program or piece of hardware that converts an IP address from a private address to a public address in real time. It allows multiple users to share a single public IP address.
NAT traversal	Detects that the endpoints are behind a NAT device and provide NAT traversal.
NNI	Network to network interface. The border between two carriers.

Number analysis A set of rules to determine whether a called number is valid and, optionally, to assign a category to the call or edit the called number.

0

OAM Operation, administration, and maintenance.

Ρ

PEProvider edge. This is a piece of equipment situated at the edge of a service provider's network,
typically contrasted with Customer Edge (CE) equipment.

POTS	Plain old telephone service. This is the standard telephone service that most homes use. It is also referred to as the PSTN.
PSTN	Public Switched Telephone Network. The world's collection of interconnected voice-oriented public telephone networks.
R	
RADIUS	Remote Authentication Dial-In User Service. Protocol used by SIG to connect to call accounting services or authentication services.
routing policy	A set of rules on the SBE to determine the next-hop VoIP signaling entity to which a signaling request should be sent. It defines whether a given called number is valid, and if so, where to send outbound signaling.
RSIP	Realm-Specific Internet Protocol. An IP address translation technique that is an alternative to NAT. RSIP lets an enterprise safeguard many private Internet addresses behind a single public Internet address.
RTCP	Real-Time Control Protocol. A protocol to carry information on the performance of RTP traffic.
RTP	Real-Time Protocol. This is the dominant protocol for carrying VoIP media data. It is defined in RFC 3550.

S

SBE

Signaling border element (also known as *signaling proxy*). Represents the signaling agent of the SBC to handle all call processing through SIP or H.323 protocols. There can be only one signaling agent per service card. An SBE typically controls one or more media gateways. The SBE supports the following services:

- Call Admission Control (CAC)
- Signaling scrubbing
- Security functions
- Routing
- Registration/authentication
- Identity hiding
- Topology hiding
- Protocol conversion
- Facilitate transcoding by communicating with the media gateway or media server

SDPSession Description Protocol. A syntax for describing key features of media streams, including
codecs, IP addresses and ports, bit rates, and other information. It is defined in RFC 2327.Session Control
Interface (SCI)SCI controls the various DBE entities in a distributed mode of operation.

signaling address	IP address on the SBE for terminating VoIP signaling (that is, SIP, H.323). A signaling address may be qualified by a VPN ID (VRF name) if the SBE needs to be assigned private addresses specific to particular VPNs.
signaling protocol translation and interworking	Performs protocol translation between different signaling protocols such as SIP and H.323.
SIP	Session Initiation Protocol. A protocol used for signaling for VoIP.
SLA	Service Level Agreement. The contract between a service provider and the customer that specifies the level of service that will be provided.
SNMP	Simple Network Management Protocol. An Internet standard that defines methods for remotely managing active network components such as hubs, routers, and bridges.
SOAP	Simple Object Access Protocol. A way for a web server to call a procedure on another, physically separate web server, and get back a machine-readable result in standard XML format.
SP	Service provider.
SVI	Service virtual interface.
т	
ТСР	Transmission Control Protocol. The connection-oriented, transport-level protocol used in the TCP/IP suite of communications protocols.
TLS	Transport Layer Security. A protocol that provides data integrity and privacy on a communications link over the Internet. It allows client/server applications to communicate and is designed to prevent eavesdropping, message forgery, and interference.
topology and infrastructure hiding	Hiding organization topology and infrastructure by removing routing information or by modifying the From/Contact information in the signaling headers.
transcoder	Technology for converting between different codecs.
U	
UDP	User Datagram Protocol. This is a transport layer protocol in the TCP/IP protocol suite, used in the Internet. UDP is used at the two ends of a data transfer. It does not establish a connection or provide reliable data transfer like TCP.
UNI	User-to-Network Interface. The border between a service provider and the customer.

V

I

VDBE	Represents a resource partition within a DBE. A VDBE is a type of media gateway. Each VDBE can be controlled by a separate SBE using the H.248 (Megaco) protocol.
VoIP	Voice over IP.
VPN	Virtual Private Network.
VRF	Virtual Routing and Forwarding Instances
VoIP signaling peer	Peer device within the VoIP signaling network.
VoIP event	Significant events within the VoIP network, such as new calls, call updates, and subscriber registrations.