



Cisco IOS SIP Configuration Guide

Release 12.4T

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Cisco IOS SIP Configuration Guide

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About Cisco IOS Software Documentation

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This document describes the objectives, audience, conventions, and organization used in Cisco IOS software documentation. Also included are resources for obtaining technical assistance, additional documentation, and other information from Cisco. This document is organized into the following sections:

- [Documentation Objectives, page i](#)
- [Audience, page i](#)
- [Documentation Conventions, page i](#)
- [Documentation Organization, page iii](#)
- [Additional Resources and Documentation Feedback, page xii](#)

Documentation Objectives

Cisco IOS documentation describes the tasks and commands available to configure and maintain Cisco networking devices.

Audience

The Cisco IOS documentation set is intended for users who configure and maintain Cisco networking devices (such as routers and switches) but who may not be familiar with the configuration and maintenance tasks, the relationship among tasks, or the Cisco IOS commands necessary to perform particular tasks. The Cisco IOS documentation set is also intended for those users experienced with Cisco IOS software who need to know about new features, new configuration options, and new software characteristics in the current Cisco IOS release.

Documentation Conventions

In Cisco IOS documentation, the term *router* may be used to refer to various Cisco products; for example, routers, access servers, and switches. These and other networking devices that support Cisco IOS software are shown interchangeably in examples and are used only for illustrative purposes. An example that shows one product does not necessarily mean that other products are not supported.

This section contains the following topics:

- [Typographic Conventions, page ii](#)
- [Command Syntax Conventions, page ii](#)
- [Software Conventions, page iii](#)
- [Reader Alert Conventions, page iii](#)

Typographic Conventions

Cisco IOS documentation uses the following typographic conventions:

Convention	Description
^ or Ctrl	Both the ^ symbol and Ctrl represent the Control (Ctrl) key on a keyboard. For example, the key combination ^D or Ctrl-D means that you hold down the Control key while you press the D key. (Keys are indicated in capital letters but are not case sensitive.)
<i>string</i>	A string is a nonquoted set of characters shown in italics. For example, when setting a Simple Network Management Protocol (SNMP) community string to <i>public</i> , do not use quotation marks around the string; otherwise, the string will include the quotation marks.

Command Syntax Conventions

Cisco IOS documentation uses the following command syntax conventions:

Convention	Description
bold	Bold text indicates commands and keywords that you enter as shown.
<i>italic</i>	Italic text indicates arguments for which you supply values.
[x]	Square brackets enclose an optional keyword or argument.
...	An ellipsis (three consecutive nonbolded periods without spaces) after a syntax element indicates that the element can be repeated.
	A vertical line, called a pipe, that is enclosed within braces or square brackets indicates a choice within a set of keywords or arguments.
[x y]	Square brackets enclosing keywords or arguments separated by a pipe indicate an optional choice.
{x y}	Braces enclosing keywords or arguments separated by a pipe indicate a required choice.
[x {y z}]	Braces and a pipe within square brackets indicate a required choice within an optional element.

Software Conventions

Cisco IOS software uses the following program code conventions:

Convention	Description
Courier font	Courier font is used for information that is displayed on a PC or terminal screen.
Bold Courier font	Bold Courier font indicates text that the user must enter.
< >	Angle brackets enclose text that is not displayed, such as a password. Angle brackets also are used in contexts in which the italic font style is not supported; for example, ASCII text.
!	An exclamation point at the beginning of a line indicates that the text that follows is a comment, not a line of code. An exclamation point is also displayed by Cisco IOS software for certain processes.
[]	Square brackets enclose default responses to system prompts.

Reader Alert Conventions

Cisco IOS documentation uses the following conventions for reader alerts:



Caution

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



Note

Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the manual.



Timesaver

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

Documentation Organization

This section describes the Cisco IOS documentation set, how it is organized, and how to access it on Cisco.com. It also lists the configuration guides, command references, and supplementary references and resources that comprise the documentation set. It contains the following topics:

- [Cisco IOS Documentation Set, page iv](#)
- [Cisco IOS Documentation on Cisco.com, page iv](#)
- [Configuration Guides, Command References, and Supplementary Resources, page v](#)

Cisco IOS Documentation Set

The Cisco IOS documentation set consists of the following:

- Release notes and caveats provide information about platform, technology, and feature support for a release and describe severity 1 (catastrophic), severity 2 (severe), and select severity 3 (moderate) defects in released Cisco IOS software. Review release notes before other documents to learn whether updates have been made to a feature.
- Sets of configuration guides and command references organized by technology and published for each standard Cisco IOS release.
 - Configuration guides—Compilations of documents that provide conceptual and task-oriented descriptions of Cisco IOS features.
 - Command references—Compilations of command pages in alphabetical order that provide detailed information about the commands used in the Cisco IOS features and the processes that comprise the related configuration guides. For each technology, there is a single command reference that supports all Cisco IOS releases and that is updated at each standard release.
- Lists of all the commands in a specific release and all commands that are new, modified, removed, or replaced in the release.
- Command reference book for **debug** commands. Command pages are listed in alphabetical order.
- Reference book for system messages for all Cisco IOS releases.

Cisco IOS Documentation on Cisco.com

The following sections describe the organization of the Cisco IOS documentation set and how to access various document types.

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

New Features List

The New Features List for each release provides a list of all features in the release with hyperlinks to the feature guides in which they are documented.

Feature Guides

Cisco IOS features are documented in feature guides. Feature guides describe one feature or a group of related features that are supported on many different software releases and platforms. Your Cisco IOS software release or platform may not support all the features documented in a feature guide. See the Feature Information table at the end of the feature guide for information about which features in that guide are supported in your software release.

Configuration Guides

Configuration guides are provided by technology and release and comprise a set of individual feature guides relevant to the release and technology.

Command References

Command reference books contain descriptions of Cisco IOS commands that are supported in many different software releases and on many different platforms. The books are organized by technology. For information about all Cisco IOS commands, use the Command Lookup Tool at <http://tools.cisco.com/Support/CLILookup> or the *Cisco IOS Master Command List, All Releases*, at http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all_book.html.

Cisco IOS Supplementary Documents and Resources

Supplementary documents and resources are listed in [Table 2 on page xi](#).

Configuration Guides, Command References, and Supplementary Resources

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

[Table 2](#) lists documents and resources that supplement the Cisco IOS 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>

Table 1 *Cisco IOS Configuration Guides and Command References*

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS AppleTalk Configuration Guide</i> • <i>Cisco IOS AppleTalk Command Reference</i> 	AppleTalk protocol.
<ul style="list-style-type: none"> • <i>Cisco IOS Asynchronous Transfer Mode Configuration Guide</i> • <i>Cisco IOS Asynchronous Transfer Mode Command Reference</i> 	LAN ATM, multiprotocol over ATM (MPoA), and WAN ATM.

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS Bridging and IBM Networking Configuration Guide</i> • <i>Cisco IOS Bridging Command Reference</i> • <i>Cisco IOS IBM Networking Command Reference</i> 	<p>Transparent and source-route transparent (SRT) bridging, source-route bridging (SRB), Token Ring Inter-Switch Link (TRISL), and token ring route switch module (TRRSM).</p> <p>Data-link switching plus (DLSw+), serial tunnel (STUN), block serial tunnel (BSTUN); logical link control, type 2 (LLC2), synchronous data link control (SDLC); IBM Network Media Translation, including Synchronous Data Logical Link Control (SDLLC) and qualified LLC (QLLC); downstream physical unit (DSPU), Systems Network Architecture (SNA) service point, SNA frame relay access, advanced peer-to-peer networking (APPN), native client interface architecture (NCIA) client/server topologies, and IBM Channel Attach.</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Broadband Access Aggregation and DSL Configuration Guide</i> • <i>Cisco IOS Broadband Access Aggregation and DSL Command Reference</i> 	<p>PPP over ATM (PPPoA) and PPP over Ethernet (PPPoE).</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Carrier Ethernet Configuration Guide</i> • <i>Cisco IOS Carrier Ethernet Command Reference</i> 	<p>Connectivity fault management (CFM), Ethernet Local Management Interface (ELMI), IEEE 802.3ad link bundling, Link Layer Discovery Protocol (LLDP), media endpoint discovery (MED), and Operation, Administration, and Maintenance (OAM).</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Configuration Fundamentals Configuration Guide</i> • <i>Cisco IOS Configuration Fundamentals Command Reference</i> 	<p>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.</p>
<ul style="list-style-type: none"> • <i>Cisco IOS DECnet Configuration Guide</i> • <i>Cisco IOS DECnet Command Reference</i> 	<p>DECnet protocol.</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Dial Technologies Configuration Guide</i> • <i>Cisco IOS Dial Technologies Command Reference</i> 	<p>Asynchronous communications, dial backup, dialer technology, dial-in terminal services and AppleTalk remote access (ARA), dial-on-demand routing, dial-out, ISDN, large scale dial-out, modem and resource pooling, Multilink PPP (MLP), PPP, and virtual private dialup network (VPDN).</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Flexible NetFlow Configuration Guide</i> • <i>Cisco IOS Flexible NetFlow Command Reference</i> 	<p>Flexible NetFlow.</p>
<ul style="list-style-type: none"> • <i>Cisco IOS High Availability Configuration Guide</i> • <i>Cisco IOS High Availability Command Reference</i> 	<p>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.</p>
<ul style="list-style-type: none"> • <i>Cisco IOS Integrated Session Border Controller Command Reference</i> 	<p>A VoIP-enabled device that is deployed at the edge of networks. An SBC is a toolkit of functions, such as signaling interworking, network hiding, security, and quality of service (QoS).</p>

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS Intelligent Services Gateway Configuration Guide</i> • <i>Cisco IOS Intelligent Services Gateway Command Reference</i> 	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.
<ul style="list-style-type: none"> • <i>Cisco IOS Interface and Hardware Component Configuration Guide</i> • <i>Cisco IOS Interface and Hardware Component Command Reference</i> 	LAN interfaces, logical interfaces, serial interfaces, virtual interfaces, and interface configuration.
<ul style="list-style-type: none"> • <i>Cisco IOS IP Addressing Services Configuration Guide</i> • <i>Cisco IOS IP Addressing Services Command Reference</i> 	Address Resolution Protocol (ARP), Network Address Translation (NAT), Domain Name System (DNS), Dynamic Host Configuration Protocol (DHCP), and Next Hop Address Resolution Protocol (NHRP).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Application Services Configuration Guide</i> • <i>Cisco IOS IP Application Services Command Reference</i> 	Enhanced Object Tracking (EOT), Gateway Load Balancing Protocol (GLBP), Hot Standby Router Protocol (HSRP), IP Services, Server Load Balancing (SLB), Stream Control Transmission Protocol (SCTP), TCP, Web Cache Communication Protocol (WCCP), User Datagram Protocol (UDP), and Virtual Router Redundancy Protocol (VRRP).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Mobility Configuration Guide</i> • <i>Cisco IOS IP Mobility Command Reference</i> 	Mobile ad hoc networks (MANet) and Cisco mobile networks.
<ul style="list-style-type: none"> • <i>Cisco IOS IP Multicast Configuration Guide</i> • <i>Cisco IOS IP Multicast Command Reference</i> 	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).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing Protocols Configuration Guide</i> • <i>Cisco IOS IP Routing Protocols Command Reference</i> 	Border Gateway Protocol (BGP), multiprotocol BGP, multiprotocol BGP extensions for IP multicast, bidirectional forwarding detection (BFD), Enhanced Interior Gateway Routing Protocol (EIGRP), Interior Gateway Routing Protocol (IGRP), Intermediate System-to-Intermediate System (IS-IS), On-Demand Routing (ODR), Open Shortest Path First (OSPF), and Routing Information Protocol (RIP).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing: BFD Configuration Guide</i> 	Bidirectional forwarding detection (BFD).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing: BGP Configuration Guide</i> • <i>Cisco IOS IP Routing: BGP Command Reference</i> 	Border Gateway Protocol (BGP), multiprotocol BGP, multiprotocol BGP extensions for IP multicast.
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing: EIGRP Configuration Guide</i> • <i>Cisco IOS IP Routing: EIGRP Command Reference</i> 	Enhanced Interior Gateway Routing Protocol (EIGRP).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing: ISIS Configuration Guide</i> • <i>Cisco IOS IP Routing: ISIS Command Reference</i> 	Intermediate System-to-Intermediate System (IS-IS).
<ul style="list-style-type: none"> • <i>Cisco IOS IP Routing: ODR Configuration Guide</i> • <i>Cisco IOS IP Routing: ODR Command Reference</i> 	On-Demand Routing (ODR).

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> <i>Cisco IOS IP Routing: OSPF Configuration Guide</i> <i>Cisco IOS IP Routing: OSPF Command Reference</i> 	Open Shortest Path First (OSPF).
<ul style="list-style-type: none"> <i>Cisco IOS IP Routing: Protocol-Independent Configuration Guide</i> <i>Cisco IOS IP Routing: Protocol-Independent Command Reference</i> 	IP routing protocol-independent features and commands. Generic policy-based routing (PBR) features and commands are included.
<ul style="list-style-type: none"> <i>Cisco IOS IP Routing: RIP Configuration Guide</i> <i>Cisco IOS IP Routing: RIP Command Reference</i> 	Routing Information Protocol (RIP).
<ul style="list-style-type: none"> <i>Cisco IOS IP SLAs Configuration Guide</i> <i>Cisco IOS IP SLAs Command Reference</i> 	Cisco IOS IP Service Level Agreements (IP SLAs).
<ul style="list-style-type: none"> <i>Cisco IOS IP Switching Configuration Guide</i> <i>Cisco IOS IP Switching Command Reference</i> 	Cisco Express Forwarding, fast switching, and Multicast Distributed Switching (MDS).
<ul style="list-style-type: none"> <i>Cisco IOS IPv6 Configuration Guide</i> <i>Cisco IOS IPv6 Command Reference</i> 	For IPv6 features, protocols, and technologies, go to the IPv6 “Start Here” document.
<ul style="list-style-type: none"> <i>Cisco IOS ISO CLNS Configuration Guide</i> <i>Cisco IOS ISO CLNS Command Reference</i> 	ISO Connectionless Network Service (CLNS).
<ul style="list-style-type: none"> <i>Cisco IOS LAN Switching Configuration Guide</i> <i>Cisco IOS LAN Switching Command Reference</i> 	VLANs, Inter-Switch Link (ISL) encapsulation, IEEE 802.10 encapsulation, IEEE 802.1Q encapsulation, and multilayer switching (MLS).
<ul style="list-style-type: none"> <i>Cisco IOS Mobile Wireless Gateway GPRS Support Node Configuration Guide</i> <i>Cisco IOS Mobile Wireless Gateway GPRS Support Node Command Reference</i> 	Cisco IOS Gateway GPRS Support Node (GGSN) in a 2.5-generation general packet radio service (GPRS) and 3-generation universal mobile telecommunication system (UMTS) network.
<ul style="list-style-type: none"> <i>Cisco IOS Mobile Wireless Home Agent Configuration Guide</i> <i>Cisco IOS Mobile Wireless Home Agent Command Reference</i> 	Cisco Mobile Wireless Home Agent, an anchor point for mobile terminals for which mobile IP or proxy mobile IP services are provided.
<ul style="list-style-type: none"> <i>Cisco IOS Mobile Wireless Packet Data Serving Node Configuration Guide</i> <i>Cisco IOS Mobile Wireless Packet Data Serving Node Command Reference</i> 	Cisco Packet Data Serving Node (PDSN), a wireless gateway that is between the mobile infrastructure and standard IP networks and that enables packet data services in a code division multiple access (CDMA) environment.
<ul style="list-style-type: none"> <i>Cisco IOS Mobile Wireless Radio Access Networking Configuration Guide</i> <i>Cisco IOS Mobile Wireless Radio Access Networking Command Reference</i> 	Cisco IOS radio access network products.
<ul style="list-style-type: none"> <i>Cisco IOS Multiprotocol Label Switching Configuration Guide</i> <i>Cisco IOS Multiprotocol Label Switching Command Reference</i> 	MPLS Label Distribution Protocol (LDP), MPLS Layer 2 VPNs, MPLS Layer 3 VPNs, MPLS traffic engineering (TE), and MPLS Embedded Management (EM) and MIBs.

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS Multi-Topology Routing Configuration Guide</i> • <i>Cisco IOS Multi-Topology Routing Command Reference</i> 	Unicast and multicast topology configurations, traffic classification, routing protocol support, and network management support.
<ul style="list-style-type: none"> • <i>Cisco IOS NetFlow Configuration Guide</i> • <i>Cisco IOS NetFlow Command Reference</i> 	Network traffic data analysis, aggregation caches, and export features.
<ul style="list-style-type: none"> • <i>Cisco IOS Network Management Configuration Guide</i> • <i>Cisco IOS Network Management Command Reference</i> 	Basic system management; system monitoring and logging; troubleshooting, logging, and fault management; Cisco Discovery Protocol; Cisco IOS Scripting with Tool Control Language (Tcl); Cisco networking services (CNS); DistributedDirector; Embedded Event Manager (EEM); Embedded Resource Manager (ERM); Embedded Syslog Manager (ESM); HTTP; Remote Monitoring (RMON); SNMP; and VPN Device Manager Client for Cisco IOS software (XSM Configuration).
<ul style="list-style-type: none"> • <i>Cisco IOS Novell IPX Configuration Guide</i> • <i>Cisco IOS Novell IPX Command Reference</i> 	Novell Internetwork Packet Exchange (IPX) protocol.
<ul style="list-style-type: none"> • <i>Cisco IOS Optimized Edge Routing Configuration Guide</i> • <i>Cisco IOS Optimized Edge Routing Command Reference</i> 	Optimized edge routing (OER) monitoring; Performance Routing (PfR); and automatic route optimization and load distribution for multiple connections between networks.
<ul style="list-style-type: none"> • <i>Cisco IOS Quality of Service Solutions Configuration Guide</i> • <i>Cisco IOS Quality of Service Solutions Command Reference</i> 	Traffic queueing, traffic policing, traffic shaping, Modular QoS CLI (MQC), Network-Based Application Recognition (NBAR), Multilink PPP (MLP) for QoS, header compression, AutoQoS, Resource Reservation Protocol (RSVP), and weighted random early detection (WRED).
<ul style="list-style-type: none"> • <i>Cisco IOS Security Command Reference</i> 	Access control lists (ACLs); authentication, authorization, and accounting (AAA); firewalls; IP security and encryption; neighbor router authentication; network access security; network data encryption with router authentication; public key infrastructure (PKI); RADIUS; TACACS+; terminal access security; and traffic filters.
<ul style="list-style-type: none"> • <i>Cisco IOS 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.
<ul style="list-style-type: none"> • <i>Cisco IOS Security Configuration Guide: Securing the Control Plane</i> 	Control Plane Policing, Neighborhood Router Authentication.
<ul style="list-style-type: none"> • <i>Cisco IOS Security Configuration Guide: Securing User Services</i> 	AAA (includes 802.1x authentication and 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.

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS Security Configuration Guide: Secure Connectivity</i> 	Internet Key Exchange (IKE) for IPsec VPNs; IPsec Data Plane features; IPsec Management features; Public Key Infrastructure (PKI); Dynamic Multipoint VPN (DMVPN); Easy VPN; Cisco Group Encrypted Transport VPN (GETVPN); SSL VPN.
<ul style="list-style-type: none"> • <i>Cisco IOS Service Advertisement Framework Configuration Guide</i> • <i>Cisco IOS Service Advertisement Framework Command Reference</i> 	Cisco Service Advertisement Framework.
<ul style="list-style-type: none"> • <i>Cisco IOS Service Selection Gateway Configuration Guide</i> • <i>Cisco IOS Service Selection Gateway Command Reference</i> 	Subscriber authentication, service access, and accounting.
<ul style="list-style-type: none"> • <i>Cisco IOS Software Activation Configuration Guide</i> • <i>Cisco IOS Software Activation Command Reference</i> 	An orchestrated collection of processes and components to activate Cisco IOS software feature sets by obtaining and validating Cisco software licenses.
<ul style="list-style-type: none"> • <i>Cisco IOS Software Modularity Installation and Configuration Guide</i> • <i>Cisco IOS Software Modularity Command Reference</i> 	Installation and basic configuration of software modularity images, including installations on single and dual route processors, installation rollbacks, software modularity binding, software modularity processes, and patches.
<ul style="list-style-type: none"> • <i>Cisco IOS Terminal Services Configuration Guide</i> • <i>Cisco IOS Terminal Services Command Reference</i> 	DEC, local-area transport (LAT), and X.25 packet assembler/disassembler (PAD).
<ul style="list-style-type: none"> • <i>Cisco IOS Virtual Switch Command Reference</i> 	Virtual switch redundancy, high availability, and packet handling; converting between standalone and virtual switch modes; virtual switch link (VSL); Virtual Switch Link Protocol (VSLP). Note For information about virtual switch configuration, see the product-specific software configuration information for the Cisco Catalyst 6500 series switch or for the Metro Ethernet 6500 series switch.
<ul style="list-style-type: none"> • <i>Cisco IOS Voice Configuration Library</i> • <i>Cisco IOS Voice Command Reference</i> 	Cisco IOS support for voice call control protocols, interoperability, physical and virtual interface management, and troubleshooting. The library includes documentation for IP telephony applications.
<ul style="list-style-type: none"> • <i>Cisco IOS VPDN Configuration Guide</i> • <i>Cisco IOS VPDN Command Reference</i> 	Layer 2 Tunneling Protocol (L2TP) dial-out load balancing and redundancy; L2TP extended failover; L2TP security VPDN; 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; tunnel authentication via RADIUS on tunnel terminator.

Table 1 Cisco IOS Configuration Guides and Command References (continued)

Configuration Guide and Command Reference Titles	Features/Protocols/Technologies
<ul style="list-style-type: none"> • <i>Cisco IOS Wide-Area Networking Configuration Guide</i> • <i>Cisco IOS Wide-Area Networking Command Reference</i> 	Frame Relay; Layer 2 Tunnel Protocol Version 3 (L2TPv3); L2VPN Pseudowire Redundancy; L2VPN Interworking; Layer 2 Local Switching; Link Access Procedure, Balanced (LAPB); and X.25.
<ul style="list-style-type: none"> • <i>Cisco IOS Wireless LAN Configuration Guide</i> • <i>Cisco IOS Wireless LAN Command Reference</i> 	Broadcast key rotation, IEEE 802.11x support, IEEE 802.1x authenticator, IEEE 802.1x local authentication service for Extensible Authentication Protocol-Flexible Authentication via Secure Tunneling (EAP-FAST), Multiple Basic Service Set ID (BSSID), Wi-Fi Multimedia (WMM) required elements, and Wi-Fi Protected Access (WPA).

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

Table 2 Cisco IOS Supplementary Documents and Resources

Document Title or Resource	Description
<i>Cisco IOS Master Command List, All Releases</i>	Alphabetical list of all the commands documented in all Cisco IOS releases.
<i>Cisco IOS New, Modified, Removed, and Replaced Commands</i>	List of all the new, modified, removed, and replaced commands for a Cisco IOS release.
<i>Cisco IOS Software System Messages</i>	List of Cisco IOS 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 system software.
<i>Cisco IOS Debug Command Reference</i>	Alphabetical list of debug commands including brief descriptions of use, command syntax, and usage guidelines.
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 software releases.
MIBs	Files used for network monitoring. To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator .
RFCs	Standards documents maintained by the Internet Engineering Task Force (IETF) that Cisco IOS documentation references where applicable. The full text of referenced RFCs may be obtained at the following URL: http://www.rfc-editor.org/

Additional Resources and Documentation Feedback

What's New in Cisco Product Documentation is released monthly and describes all new and revised Cisco technical documentation. The *What's New in Cisco Product Documentation* publication also provides information about obtaining the following resources:

- Technical documentation
- Cisco product security overview
- Product alerts and field notices
- Technical assistance

Cisco IOS technical documentation includes embedded feedback forms where you can rate documents and provide suggestions for improvement. Your feedback helps us improve our documentation.

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Using the Command-Line Interface in Cisco IOS Software

Last Updated: October 14, 2009

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

- [Initially Configuring a Device, page i](#)
- [Using the CLI, page ii](#)
- [Saving Changes to a Configuration, page xi](#)
- [Additional Information, page xii](#)

For more information about using the CLI, see the “[Using the Cisco IOS Command-Line Interface](#)” section of the *Cisco IOS Configuration Fundamentals Configuration Guide*.

For information about the software documentation set, see the “[About Cisco IOS 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/Technologies 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 changes that you can make to a console port and 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.

**Note**

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 ii](#)
- [Using the Interactive Help Feature, page v](#)
- [Understanding Command Syntax, page vi](#)
- [Understanding Enable and Enable Secret Passwords, page vii](#)
- [Using the Command History Feature, page viii](#)
- [Abbreviating Commands, page ix](#)
- [Using Aliases for CLI Commands, page ix](#)
- [Using the no and default Forms of Commands, page x](#)
- [Using the debug Command, page x](#)
- [Filtering Output Using Output Modifiers, page x](#)
- [Understanding CLI Error Messages, page xi](#)

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](#) lists common command modes with associated CLI prompts, access and exit methods, and a brief description of how each mode is used.

Table 1 CLI Command Modes

Command Mode	Access Method	Prompt	Exit Method	Mode Usage
User EXEC	Log in.	Router>	Issue the logout or exit command.	<ul style="list-style-type: none"> • 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.	<ul style="list-style-type: none"> • 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.
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 CLI Command Modes (continued)

Command Mode	Access Method	Prompt	Exit Method	Mode Usage
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.	<ul style="list-style-type: none"> 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 (available only on Cisco ASR 1000 series routers)	<p>The router boots or enters diagnostic mode in the following scenarios. When a Cisco IOS process or processes fail, in most scenarios the router will reload.</p> <ul style="list-style-type: none"> 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) #	<p>If a Cisco IOS process failure is the reason for entering diagnostic mode, the failure must be resolved and the router must be rebooted to exit diagnostic mode.</p> <p>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 CLI.</p> <p>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.</p>	<ul style="list-style-type: none"> Inspect various states on the router, including the Cisco IOS state. Replace or roll back the configuration. Provide methods of restarting the Cisco IOS 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.

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 2](#) describes the purpose of the CLI interactive Help commands.

Table 2 CLI Interactive Help Commands

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

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

```
<snip>
```

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 3](#) describes these conventions.

Table 3 CLI Syntax 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)	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.	—

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

Using the Command History Feature

The command history feature saves, in a command history buffer, the commands that you enter during a session. The default number of saved commands 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.



Note 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 **vrp** as a keyword in addition to **version**. (Command and keyword examples are from Cisco IOS Release 12.4(13)T.)

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 4 shows the default command aliases.

Table 4 Default Command Aliases

Command Alias	Original Command
h	help
lo	logout
p	ping
s	show
u or un	undebug
w	where

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_a1.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 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.

The **no** form is documented in the command pages of command references. The **default** form is generally documented in the command pages only when the **default** form performs a different function than the plain and **no** forms of the command. To see what **default** commands are available on your system, enter **default ?** in the appropriate command mode.

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 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 **undebg 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. Using output modifiers, you can 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 (|), 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 5](#) shows the common CLI error messages.

Table 5 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.

For more system error messages, see the following document:

- [Cisco IOS Release 12.4T System Message Guide](#)

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

- “Using the Cisco IOS Command-Line Interface” section of the *Cisco IOS Configuration Fundamentals Configuration Guide*
http://www.cisco.com/en/US/docs/ios/fundamentals/configuration/guide/cf_cli-basics.html
- Cisco Product/Technology Support
<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 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 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/cgi-bin/Support/OutputInterpreter/home.pl>

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SIP Features Roadmap

First Published: March 1992

Last Updated: February 27, 2009

This chapter contains a list of SIP features (Cisco IOS Release 12.3 and later) and the location of associated documentation.

Finding Support Information for Platforms and Cisco IOS and Catalyst OS Software Images

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

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(24)T	RSVP Preconditions for Video Gateway	Expands existing support for SIP video calls on H.324-SIP video gateways to include H.320-SIP video gateways. Additionally, this feature adds support for SIP video RSVP preconditions for SIP video calls on both H.320-SIP and H.324-SIP video gateways.	Refer to the “ Configuring SIP RSVP Features ” module in this guide.
12.4(22)T	RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express (Cisco Unified CME)	Provides application-specific reservations that enhance the granularity of local policy match criteria on Cisco IOS SIP devices. Additionally, this feature provides support for SIP audio RSVP preconditions for audio on both SIP time-division multiplexing (TDM) gateways and on SIP trunks for Skinny Client Control Protocol (SCCP) line-side Cisco Unified CME devices.	Refer to the “ Configuring SIP RSVP Features ” module in this guide.



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Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(22)T	SIP Diversion Header Enhancements	Upgrades the Diversion header draft implementation to the draft-levy-diversion-06.txt version. This upgrade adds the capability to send or receive two new parameters in the Diversion header. The stack adds two new fields to set or pass this information to and from the application.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.4(22)T	SIP History INFO	Provides support for the history-info header in SIP INVITE messages only. The SIP gateway generates history information in the INVITE message for all forwarded and transferred calls.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.4(22)T	SIP Multicast Music on Hold (MoH)	Enables the multicast music-on-hold (MOH) feature on a voice gateway.	Refer to the ccm-manager music-on-hold command included in the Cisco IOS Voice Command Reference .
12.4(20)T	Caller ID on FXO for MGCP	Provides caller ID on FXO for MGCP calls.	Refer to the various caller-id commands included in the Cisco IOS Voice Command Reference and the Cisco Unified CME Command Reference .
12.4(20)T	Control Media Cut-Through on SIP 18x Response	Provides the ability to send media backward before a call is established, allowing the remote side to send personalized ringback tones (usually music) as a response before a call is established. This is default behavior but, in some scenarios, causes clipping and may need to be disabled (CLI for disabling this behavior was added for this release).	Refer to the rtp send-recv command in the Cisco IOS Voice Command Reference .
12.4(20)T	Disable Outbound SIP Proxy on a per-Dial-Peer Basis	Provides a fix for the issue of calls coming in over a SIP trunk to Cisco Unified CME and being forwarded to the outbound SIP proxy rather than directly to the phone.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.4(20)T	G.729br8 Codec as a Superset of G.729r8 and G.729br8 Codecs	Provides the ability for a Cisco IOS SIP gateway to interoperate with Cisco Unified Communications Manager, formerly known as the Cisco Unified CallManager (CUCM) or Cisco CallManager (CCM).	Refer to the g729 annexb-all (global) and voice-class sip g729 annexb-all (dial peer) commands in the Cisco IOS Voice Command Reference .
12.4(20)T	ISDN FACILITY and NOTIFY mapping to SIP INFO	Maps ISDN FACILITY (supporting 4ESS and 5ESS switch types) and ISDN NOTIFY (supporting DMS 100 switch type) to SIP INFO messages. FACILITY and NOTIFY messages are mapped to GTD, which is then carried in the SIP INFO message. The GTD can also be populated with RAW message.	There is no associated CLI or configuration for this feature.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(20)T	Pass Data in SIP REFER to Triggered INVITE	Maps SIP REFER message data into SIP INVITE messages. This allows you to send customer-specific information to triggered SIP INVITE messages using Call-Info as the URL header of the SIP Refer-To header. Further, this feature allows the gateway to take SIP REFER data and create a new SIP INVITE message to a new destination when a call is being placed to an Interactive Voice Response (IVR) endpoint and the IVR refers the call to an agent or to another IVR system.	There is no associated CLI or configuration for this feature.
12.4(20)T	SIP SRTP Fallback to Nonsecure RTP	The SIP SRTP fallback to non-secure RTP feature allows Secure Real-Time Transport Protocol (SRTP) transactions to fallback to non-secure RTP using SIP 4xx responses on a Cisco IOS SIP gateway. Beginning with Cisco IOS Release 12.4(15)XY (integrated into Cisco IOS Release 12.4(20)T), you can use the srtp negotiate command to allow a Cisco IOS SIP gateway to accept and send an RTP Audio/Video Profile (AVP) in response to an RTP Secure AVP offer (also known as an SRTP profile) using 4xx messages.	Refer to the srtp (dial peer), srtp (global), srtp negotiate (global), and voice-class sip srtp negotiate (dial peer) commands in the Cisco IOS Voice Command Reference .
12.4(20)T	Transparent Tunneling of QSIG over SIP TDM Gateway	Allows tunneling of QSIG over SIP on Cisco IOS SIP TDM gateways.	Refer to the “ Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP CUBE ” module in this guide.
12.4(20)T	ISDN Q.931 Tunneling over SIP TDM Gateway	Allows tunneling of Q.931 over SIP on Cisco IOS SIP TDM gateways.	
12.4(20)T	Transparent Tunneling of QSIG and Q.931 over SIP-SIP Cisco Unified Border Element (CUBE)	Extends tunneling of QSIG and Q.931 over SIP to the CUBE.	
12.4(15)T	SIP Support for PAI	Configures either P-Asserted-Identity (PAI) or P-Preferred-Identity (PPI) privacy headers in outgoing SIP request or response messages to assert the identity of authenticated users in trusted domains.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.4(15)T	SIP Support for Asymmetric SDP	Configures SIP gateways to send and receive Dual Tone Multi-Frequency (DTMF) and dynamic codec Real Time Protocol (RTP) packets with different payloads.	Refer to the “ Configuring SIP DTMF Features ” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(15)T	SIP Support for SRTP	The Secure Real-Time Transfer protocol (SRTP) is an extension of the Real-Time Protocol (RTP) Audio/Video Profile and ensures the integrity of RTP and Real-Time Control Protocol (RTCP) packets providing authentication, integrity, and encryption of media packets between two SIP endpoints.	Refer to the “Configuring SIP Support for SRTP” module in this guide.
12.4(15)T	Outbound Proxy Support for the SIP Gateway	Configure an outbound-proxy server that receives all initiating request (INVITE and SUBSCRIBE) messages and routes them to the designated destination.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.4(11)XJ	SIP REFER outside the scope of a dialog created with a SIP INVITE	Out-of-dialog REFER (OOD-R) allows remote applications to establish calls by sending a REFER message to a SIP gateway without an initial INVITE.	Refer to the “Defining Network Parameters” module in the <i>Cisco Unified Communications Manager Express System Administrator Guide</i> .
12.4(11)XJ	Unified CME SIP Features: MoH, dialing, line updates, presence with BLF, provisioning new phones	You can disable REFER messages for call transfers and redirect responses for call forwarding from being sent by Unified CME or Unified SRST, if a destination gateway does not support supplementary services. Disabling supplementary services is supported if all endpoints use SCCP or all endpoints use SIP. It is not supported for a mix of SCCP and SIP endpoints.	Refer to the “Configuring Call Transfer and Forwarding” module in the <i>Cisco Unified Communications Manager Express System Administrator Guide</i> .
12.4(11)T	SIP Support for Hookflash	Configures IP Centrex supplementary services on SIP-enabled, Foreign Exchange Station (FXS) lines.	Refer to the “Configuring SIP Support for Hookflash” module in this guide.
12.4(11)T	RFC 2833 Dual-Tone Multifrequency (DTMF) Media Termination Point (MTP) Passthrough	Passes DTMF tones transparently between SIP endpoints that require either transcoding or use of the RSVP Agent feature.	Refer to the “Configuring SIP DTMF Features” module in this guide.
12.4(11)T	SIP MWI NOTIFY - QSIG MWI Translation	Enhances MWI functionality to include SIP-MWI-NOTIFY-to-QSIG-MWI translation between Cisco gateways or routers over a LAN or WAN.	Refer to the “Configuring SIP MWI Features” module in this guide.
12.4(9)T	SIP: SIP Gateway OOB DTMF Support with KPML	Provides a command-line interface (CLI) option that forwards DTMF tones using KeyPad Markup Language (KPML) by way of SIP SUBSCRIBE and NOTIFY messages.	Refer to the “Configuring SIP DTMF Features” module in this guide.
12.4(9)T	SIP: SIP Gateway Session Timer Support	Enhances session timer support for gateways to comply with IETF Session Timer RFC 4028.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.4(9)T	SIP: SIP Gateway Support for SDP Session Information and Permit Hostname CLI	Adds support for Session Protocol Description (SDP) session information to comply with IETF SDP RFC 2327. Adds support for validating up to 10 hostnames for incoming initial INVITE messages.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.4(4)T	SIP: CLI for Caller ID When Privacy Exists	Provides three CLI options that make the handling of caller ID information more flexible. Specifically, the SIP: CLI for Caller ID When Privacy Exists feature addresses the following situations: passing along caller ID information when privacy exists, handling the Display Name field when no display name exists; and allowing caller ID information to be passed to ISDN as network-provided.	Refer to the “Configuring SIP ISDN Support Features” module in this guide.
12.4(2)T	SIP: Domain Name Support in SIP Headers	Provides a host or domain name in the host portion of locally generated Session SIP headers.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.4(2)T	SIP: Multilevel Precedence and Priority Support	Enables gateways to interoperate with other MLPP-capable circuit-switched networks. An MLPP call has an associated priority level that applications that handle emergencies and congestions use to determine which lower-priority call to preempt in order to dedicate their end-system resources to high-priority communications.	Refer to the “Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” module in this guide.
12.4(2)T	SIP Stack Portability	Implements new capabilities to the SIP gateway Cisco IOS stack involving user-agent handling of messages, handling of unsolicited messages, support for outbound delayed media, and SIP headers and content in requests and responses.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.3(8)T	SIP Audible Message-Waiting Indicator for FXS Phones	Enables an FXS port on a voice gateway to receive audible MWI in a SIP network.	Refer to the “Configuring SIP MWI Features” module in this guide.
12.3(8)T	SIP Gateway Compliance to RFC 3261, RFC 3262, and RFC 3264	Provides compliance with RFC 3261, RFC 3262, and RFC 3264.	Refer to the “Achieving SIP RFC Compliance” module in this guide.
12.3(8)T	SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion	Implements support for Reason headers and buffered calling-name completion.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.3(8)T	SIP: Gateway HTTP Authentication Digest	Implements authentication using the digest access on the client side of a common SIP stack. The gateway responds to authentication challenges from an authenticating server, proxy server, or user-agent server. Also maintains parity between gateways, proxy servers, and SIP phones that already support authentication.	Refer to the “ Configuring SIP AAA Features ” module in this guide.
12.3(7)T	Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks	Enables call management applications to identify specific ISDN bearer (B) channels used during a voice gateway call for billing purposes. With the identification of the B channel, SIP gateways can enable port-specific features such as voice recording and call transfer.	Refer to the “ Configuring SIP ISDN Support Features ” module in this guide.
12.3(4)T	ISDN Calling Name Display	Provides end-to-end calling name display in SIP networks.	Refer to the “ Configuring SIP ISDN Support Features ” module in this guide.
12.3(4)T	SIP 300 Multiple Choice Messages	If multiple routes to a destination exist for a redirected number the SIP gateway sends a 300 Multiple Choice message, and the multiple routes in the Contact header are listed.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.3(4)T	SIP Gateway Support for the bind Command	Expands support for the bind command to allow specifying different source interfaces for signaling and media.	Refer to the “ Configuring SIP Bind Features ” module in this guide.
12.3(4)T	SIP Header/URL Support and Subscribe/Notify for External Triggers	Allows applications to send and receive SIP headers, to send SUBSCRIBE messages, and to receive NOTIFY events.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.
12.3(4)T	SIP NOTIFY-Based Out-of-Band DTMF Relay Support	Supports SCCP devices through SIP originating and terminating gateway use of Cisco proprietary NOTIFY-based out-of-band DTMF relay, which can also be used by analog phones attached to analog voice ports (FXS) on a router.	Refer to the “ Configuring SIP DTMF Features ” module in this guide.
12.3(4)T	SIP Redirect Processing Enhancement	Allows flexibility in the handling of incoming redirect or 3xx class of responses. Redirect processing is active by default, which means that SIP gateways handle incoming 3xx messages in compliance with RFC 2543.	Refer to the “ Basic SIP Configuration ” module in this guide.
12.3(4)T	SIP Register Support	Allows SIP gateways to register E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.	Refer to the “ Configuring SIP Message, Timer, and Response Features ” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.3(4)T	SIP: RFC 3261 Enhancements (RFC 3261)	Provides compliance with RFC 3261.	Refer to the “Achieving SIP RFC Compliance” module in this guide.
12.3(1)	SIP Accept-Language Header Support	Supports the Accept-Language header in SIP INVITE messages and in OPTIONS responses, which allows configuration of up to nine languages to be carried in SIP messages and to indicate multiple language preferences.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.3(1)	SIP PSTN Transport Using the Cisco Generic Transparency Descriptor (GTD)	Adds support for ISDN User Part (ISUP) Transport using Generic Transparency Descriptor (GTD).	Refer to the “Configuring SIP ISDN Support Features” module in this guide.
12.3(1)	SIP Support for Media Forking	Allows the creation of midcall multiple streams (or branches) of audio associated with a single call and then send those streams of data to different destinations.	Refer to the “Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” module in this guide.
12.2(15)T	Measurement-Based Call Admission Control for SIP	Monitors IP network capacity and rejects or redirects calls based on congestion detection. Provides an alternative to RSVP-based call admission control for VoIP service providers who do not deploy RSVP.	Refer to the “Configuring SIP QoS Features” module in this guide.
12.2(15)T	SIP: ISDN Suspend/Resume Support	Supports ISDN and ISDN User Part (ISUP) signaling basic functions, Suspend and Resume.	Refer to the “Configuring SIP ISDN Support Features” module in this guide.
12.3(13)	SIP Transfer Using the Refer Method and Call Forwarding	Adds support for initiating attended call transfer via REFER on Cisco IOS gateways.	Refer to the “Configuring SIP Call-Transfer Features” module in this guide.
12.3(13)	SIP: Hold Timer Support	Terminates a call that has been placed on hold in excess of a configurable time period, freeing up trunk resources.	Refer to the “Configuring SIP QoS Features” module in this guide.
12.2(13)T	SIP - Call Transfer Enhancements Using the Refer Method	Enhances the Refer method for call transfer.	Refer to the “Configuring SIP Call-Transfer Features” module in this guide.
12.2(13)T	SIP - Connection-Oriented Media (Comedia) Enhancements for SIP	Allows a gateway to check the media source of incoming Realtime Transport Protocol (RTP) packets, and the endpoint to advertise its presence inside or outside of Network Address Translation (NAT).	Refer to the “Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” module in this guide.
12.2(13)T	SIP Enhanced 180 Provisional Response Handling	Provides the ability to enable or disable early media cut-through on Cisco IOS gateways for SIP 180 response messages.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.2(13)T	SIP Extensions for Caller Identity and Privacy	Provides support for privacy indication, network verification, and screening of a call participant name and number.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.2(13)T	SIP: Core SIP Technology Enhancements (RFC 2543)	Provides compliance with RFC 2543 and RFC 2543-bis-04.	Refer to the “Achieving SIP RFC Compliance” module in this guide.
12.2(11)T	DTMF Events Through SIP Signaling	Supports sending DTMF event notifications from the local POTS interface via SIP NOTIFY messages from a SIP gateway.	Refer to the “Configuring SIP DTMF Features” module in this guide.
12.2(11)T	Enhanced Codec Support for SIP Using Dynamic Payloads	Enhances codec selection and payload negotiation between originating and terminating SIP gateways.	Refer to the “Configuring SIP QoS Features” module in this guide.
12.2(11)T	Internal Cause Code Consistency Between SIP and H.323	Establishes a standard set of categories for internal causes of voice call failures.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.2(11)T	Sending SIP 300 Pre-authentication for Voice Calls	Provides the means to evaluate and accept or reject call setup requests for both voice and dial calls received at universal gateways.	Refer to the “Configuring SIP AAA Features” module in this guide.
12.2(11)T	SIP - Call Transfer Using the Refer Method	Introduces the Refer method for call transfer, to supplement the Bye and Also methods implemented earlier.	Refer to the “Configuring SIP Call-Transfer Features” module in this guide.
12.2(11)T	SIP Carrier Identification Code	Enables transmission of the Carrier Identification Code (CIC) parameter from the SIP network to the ISDN.	Refer to the “Configuring SIP ISDN Support Features” module in this guide.
12.2(11)T	SIP INFO Method for DTMF Tone Generation	Adds support for out-of-band DTMF tone generation using the SIP INFO method.	Refer to the “Configuring SIP DTMF Features” module in this guide.
12.2(11)T	SIP Session Timer Support	Enables the periodical refresh of SIP sessions by sending repeated INVITE requests.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.2(8)T	Configurable Screening Indicator	Allows SIP terminating gateways to assign a specific value to octet 3a of the ISDN SETUP message screening indicator through the use of Tool Command Language (Tcl) Interactive Voice Response (IVR) 2.0 command set scripts.	Refer to the “Configuring SIP AAA Features” module in this guide.
12.2(8)T	DTMF Relay for SIP Calls Using Named Telephone Events	Provides reliable digit relay between VoIP gateways when a low-bandwidth codec is used and allows gateways to communicate with SIP phones that use NTE packets to indicate DTMF digits.	Refer to the “Configuring SIP DTMF Features” module in this guide.
12.2(8)T	Interaction with Forking Proxies	Enables the terminating gateway to handle multiple requests and the originating gateway to handle multiple provisional responses for the same call.	Refer to the “Basic SIP Configuration” module in this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
12.2(8)T	SIP - DNS SRV RFC 2782 Compliance (RFC 2782)	Provides compliance with RFC 2782 in appending protocol labels.	Refer to the “Achieving SIP RFC Compliance” module in this guide.
12.2(8)T	SIP - Enhanced Billing Support for Gateways	Provides changes to authentication, authorization, and accounting (AAA) records and Remote Authentication Dial-In User Service (RADIUS) implementations on SIP gateways to enable billing for traffic transported over SIP networks.	Refer to the “Configuring SIP AAA Features” module in this guide.
12.2(8)T	SIP Configurable PSTN Cause Code Mapping	Allows customization of the standard RFC 2543 mappings between SIP and PSTN networks.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.2(8)T	SIP Gateway Support of RSVP SIP Gateway Support of ‘tel’ URL	Allows resource reservation on SIP gateways that synchronize RSVP and SIP call-establishment procedures, ensuring that the required quality of service for a call is maintained across the IP network.	Refer to the “Configuring SIP QoS Features” module in this guide.
12.2(8)T	SIP Intra-Gateway Hairpinning	Provides call-routing capability in which an incoming call on a specific gateway is signaled through the IP network and back out the same gateway.	Refer to the “Basic SIP Configuration” module in this guide.
12.2(8)T	SIP INVITE Request with Malformed Via Header	Allows enabling of a response to an INVITE even if the Via header becomes malformed and cannot deliver the required information.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.
12.2(8)T	SIP Media Inactivity Timer	Enables gateways to monitor and disconnect VoIP calls if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period.	Refer to the “Configuring SIP Message, Timer, and Response Features” module in this guide.

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Overview of SIP

This chapter provides an overview of the Session Initiation Protocol (SIP).

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- [How SIP Works, page 4](#)
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Information About SIP

Session Initiation Protocol (SIP) is an ASCII-based, application-layer control protocol that can be used to establish, maintain, and terminate calls between two or more endpoints. SIP is an alternative protocol developed by the Internet Engineering Task Force (IETF) for multimedia conferencing over IP. SIP features are compliant with IETF RFC 2543, *SIP: Session Initiation Protocol*, published in March 1999.

The Cisco SIP implementation enables supported Cisco platforms to signal the setup of voice and multimedia calls over IP networks.

Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP Capabilities

SIP provides the following capabilities:

- Determines the location of the target endpoint—SIP supports address resolution, name mapping, and call redirection.



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- Determines the media capabilities of the target endpoint—SIP determines the lowest level of common services between the endpoints through Session Description Protocol (SDP). Conferences are established using only the media capabilities that can be supported by all endpoints.
- Determines the availability of the target endpoint—If a call cannot be completed because the target endpoint is unavailable, SIP determines whether the called party is connected to a call already or did not answer in the allotted number of rings. SIP then returns a message indicating why the target endpoint was unavailable.
- Establishes a session between the originating and target endpoints—If the call can be completed, SIP establishes a session between the endpoints. SIP also supports midcall changes, such as the addition of another endpoint to the conference or the changing of a media characteristic or codec.
- Handles the transfer and termination of calls—SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP simply establishes a session between the transferee and a new endpoint (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions among all parties.

**Note**

The term “conference” describes an established session (or call) between two or more endpoints. Conferences consist of two or more users and can be established using multicast or multiple unicast sessions.

SIP Components

SIP is a peer-to-peer protocol. The peers in a session are called user agents (UAs). A UA can function in one of the following roles:

- User-agent client (UAC)—A client application that initiates the SIP request.
- User-agent server (UAS)—A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

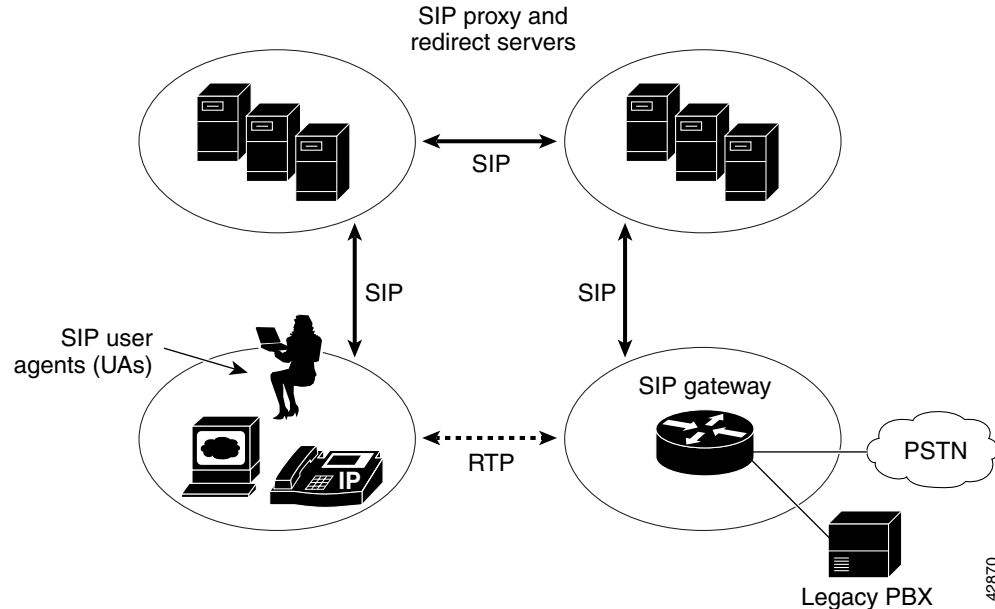
Typically, a SIP endpoint is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the user agent that initiated the request.

From an architectural standpoint, the physical components of a SIP network can be grouped into two categories: clients (endpoints) and servers. [Figure 1 on page 3](#) illustrates the architecture of a SIP network.

**Note**

In addition, the SIP servers can interact with other application services, such as Lightweight Directory Access Protocol (LDAP) servers, location servers, a database application, or an extensible markup language (XML) application. These application services provide back-end services, such as directory, authentication, and billing services.

Figure 1 SIP Architecture



SIP Clients

- Phones—Can act as either UAS or UAC.
 - Softphones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.
 - ephones—IP phones that are not configured on the gateway.
- Gateways—Provide call control. Gateways provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side.

SIP Servers

- Proxy server—Receives SIP requests from a client and forwards them on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
- Redirect server—Provides the client with information about the next hop or hops that a message should take and then the client contacts the next-hop server or UAS directly.
- Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

How SIP Works

SIP is a simple, ASCII-based protocol that uses requests and responses to establish communication among the various components in the network and to ultimately establish a conference between two or more endpoints.

Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of `sip:userID@gateway.com`. The user ID can be either a user name or an E.164 address. The gateway can be either a domain (with or without a hostname) or a specific internet IP address.

**Note**

An E.164 address is a telephone number with a string of decimal digits that uniquely indicates the public network termination point. Thus number contains all information necessary to route the call to this termination point.

Users register with a registrar server using their assigned SIP addresses. The registrar server provides this information to the location server upon request.

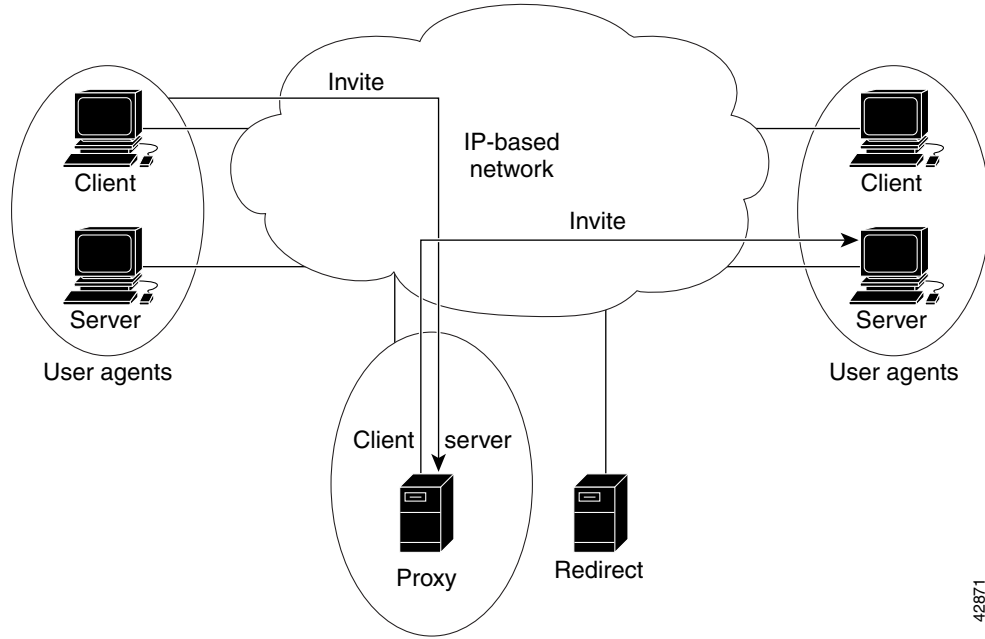
When a user initiates a call, a SIP request is sent to a SIP server (either a proxy or a redirect server). The request includes the address of the caller (in the From header field) and the address of the intended called party (in the To header field).

Over time, a SIP end user might move between end systems. The location of the end user can be dynamically registered with the SIP server. The location server can use one or more protocols (including finger, rwhois, and LDAP) to locate the end user. Because the end user can be logged in at more than one station and because the location server can sometimes have inaccurate information, it might return more than one address for the end user. If the request is coming through a SIP proxy server, the proxy server tries each of the returned addresses until it locates the end user. If the request is coming through a SIP redirect server, the redirect server forwards all the addresses to the caller in the Contact header field of the invitation response.

How SIP Works with a Proxy Server

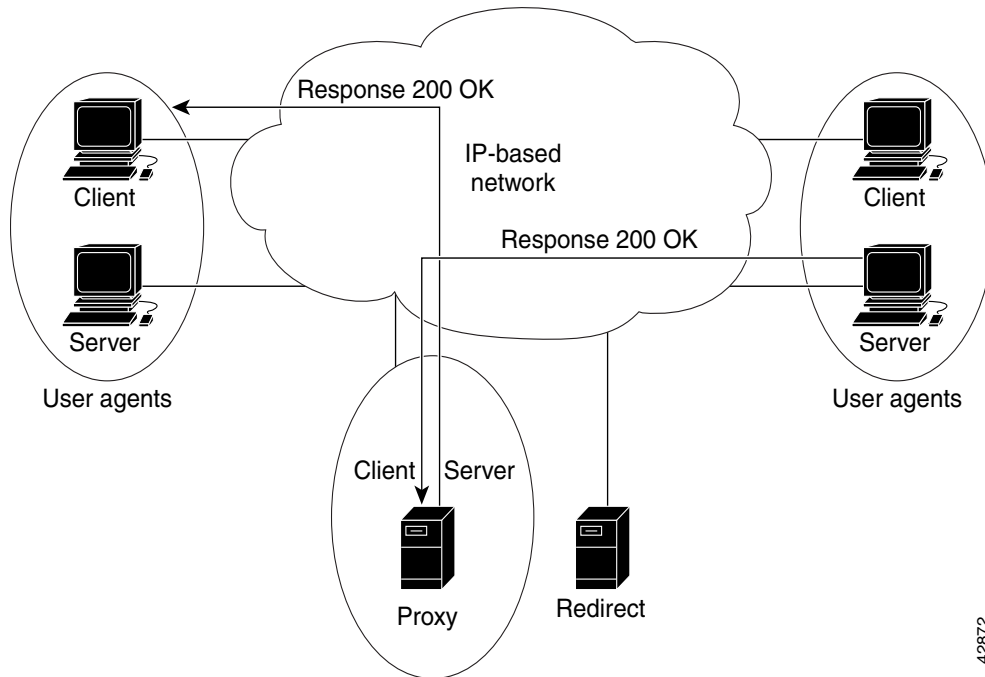
When communicating through a proxy server, the caller UA sends an INVITE request to the proxy server and then the proxy server determines the path and forwards the request to the called party (see [Figure 2](#)).

Figure 2 SIP INVITE Request Through a Proxy Server



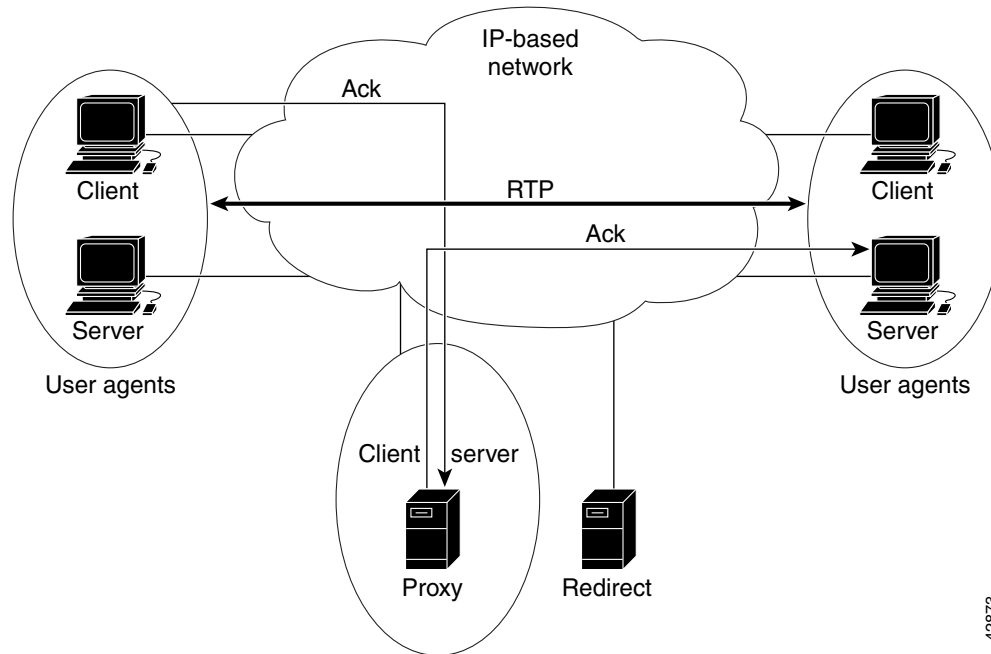
The called UA responds to the proxy server, which then forwards the response to the caller (see [Figure 3](#)).

Figure 3 SIP Response Through a Proxy Server



When both parties respond with an acknowledgement (SIP ACK message), the proxy server forwards the acknowledgments to their intended party and a session, or conference, is established between them. The Real-time Transfer Protocol (RTP) is then used for communication across the connection now established between the caller and called UA (see [Figure 4](#)).

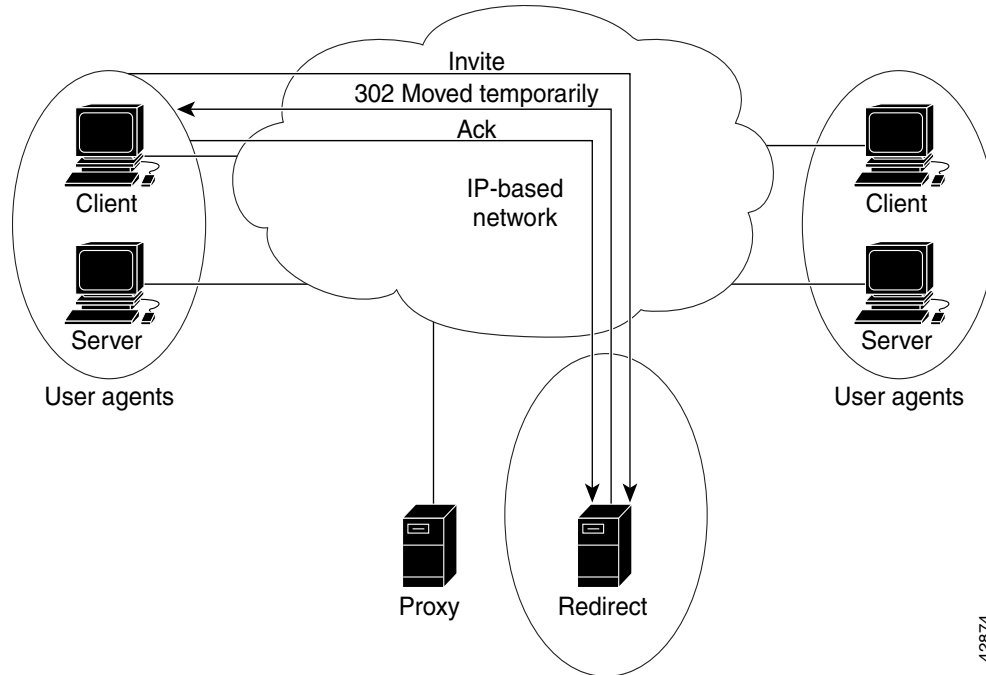
Figure 4 SIP Session Through a Proxy Server



How SIP Works with a Redirect Server

When communicating through a redirect server, the caller UA sends a SIP INVITE request to the redirect server and then the redirect contacts the location server to determine the path to the called party and sends that information back to the caller UA. The caller UA then acknowledges receipt of the information (see [Figure 5](#)).

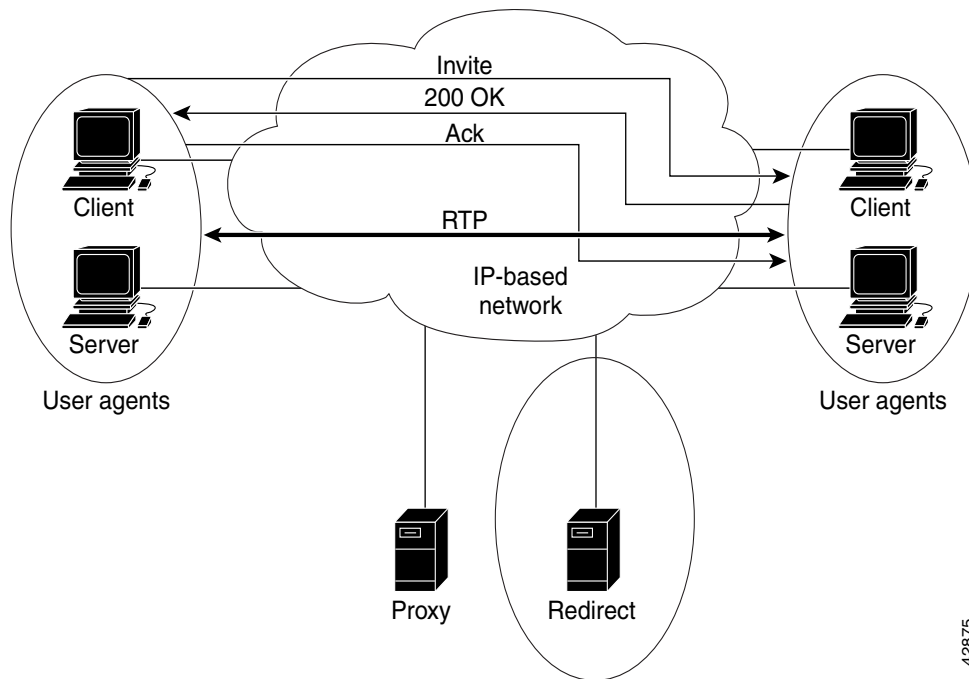
Figure 5 *SIP INVITE Through a Redirect Server*



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The caller UA then sends a SIP INVITE request directly to the device indicated in the redirect information, bypassing the redirect server. (The target device at this stage could be either the called UA itself or a proxy server that will forward the request.) Once the request reaches the called UA, the called UA sends a response and, if it is a SIP 200 OK message, the caller UA responds with a SIP ACK message to acknowledge 200 OK response. A session is then established between the two endpoints using RTP for communication between the caller and called UAs (see [Figure 6](#)).

Figure 6 SIP Session Through a Redirect Server



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SIP Call Flows

This topic describes call flows for the following scenarios, which illustrate successful calls:

- [SIP Gateway-to-SIP Gateway—Call Setup and Disconnect, page 8](#)
- [SIP Gateway-to-SIP Gateway—Call via SIP Redirect Server, page 12](#)
- [SIP Gateway-to-SIP Gateway—Call via SIP Proxy Server, page 16](#)

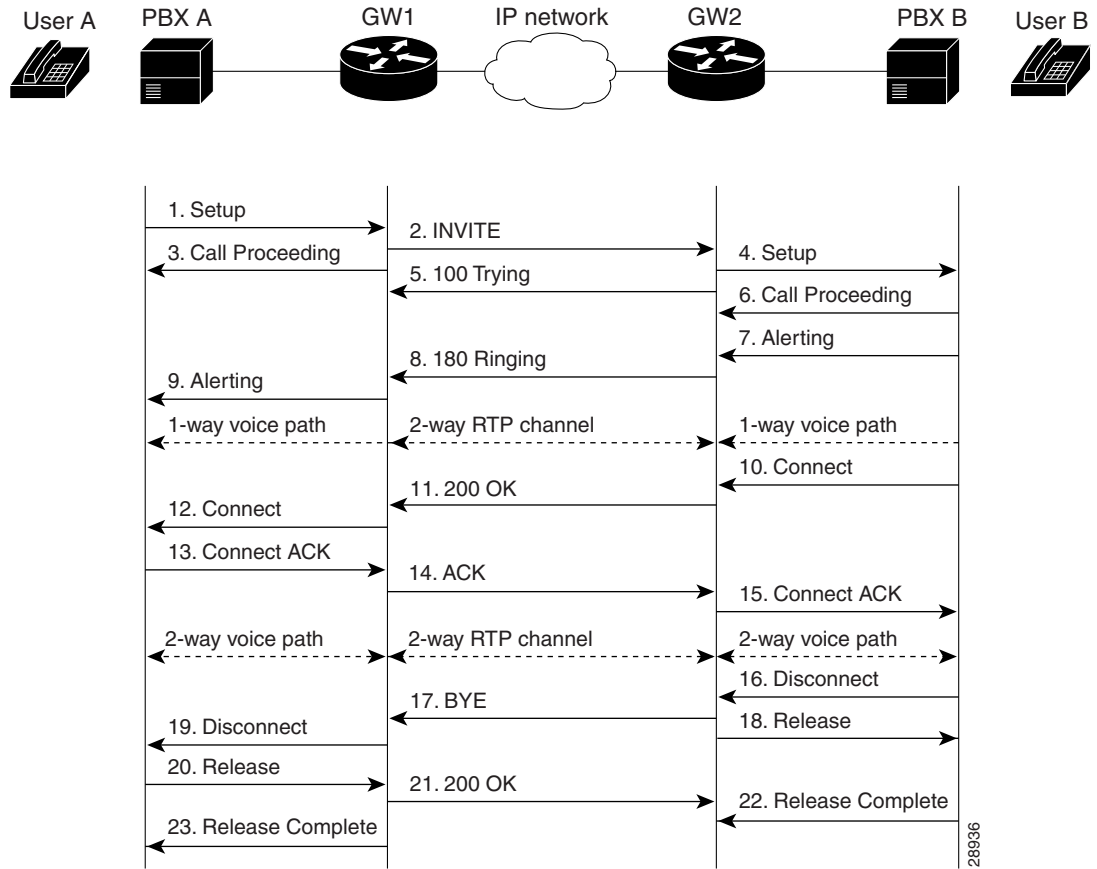
SIP Gateway-to-SIP Gateway—Call Setup and Disconnect

[Figure 7](#) shows a successful gateway-to-gateway call setup and disconnect. The two end users are User A and User B. User A is located at PBX A, which is connected to SIP gateway 1 via a T1/E1. User B is located at PBX B, which is connected to SIP gateway 2 via a T1/E1. User B's phone number is 555-0100. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B hangs up.

Figure 7 SIP Gateway-to-SIP Gateway—Call Setup and Disconnect



Note

RFC 2543-bis-04 requires that a UAS that receives a BYE request first send a response to any pending requests for that call before disconnecting. After receiving a BYE request, the UAS should respond with a 487 (Request Cancelled) status message.

The following processes occur in [Figure 7](#).

Process	Description
1. Setup—PBX A to SIP gateway 1	Call setup is initiated between PBX A and SIP gateway 1. Setup includes the standard transactions that take place as User A attempts to call User B.
2. INVITE—SIP gateway 1 to SIP gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request the following is the case:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is domain (with or without a hostname) or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0100@example.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3. Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the setup request.
4. Setup—SIP gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates call setup with User B via PBX B.
5. 100 Trying—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
6. Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the setup request.
7. Alerting—PBX B to SIP gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begins ringing.
8. 180 Ringing—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.

Process	Description
9. Alerting—SIP gateway 1 to PBX A	<p>SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from SIP gateway 2. User A hears the ringback tone that indicates that User B is being alerted.</p> <p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>
10. Connect—PBX B to SIP gateway 2	User B answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
11. 200 OK—SIP gateway 2 to SIP gateway 1	<p>SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p>
12. Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
13. Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
14. ACK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.
15. Connect ACK—SIP gateway 2 to PBX B	<p>SIP gateway 2 acknowledges PBX B's Connect message.</p> <p>The call session is now active over a two-way voice path via Real-time Transport Protocol (RTP).</p> <p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>
16. Disconnect—PBX B to SIP gateway 2	Once User B hangs up, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.
17. BYE—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL. The cseq value is incremented by one.
18. Release—SIP gateway 2 to PBX B	SIP gateway 2 sends a Release message to PBX B.
19. Disconnect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.

Process	Description
20. Release—PBX A to SIP gateway 1	PBX A sends a Disconnect message to SIP gateway 1.
21. 200 OK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
22. Release Complete—PBX B to SIP gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
23. Release Complete—SIP gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the session is terminated.

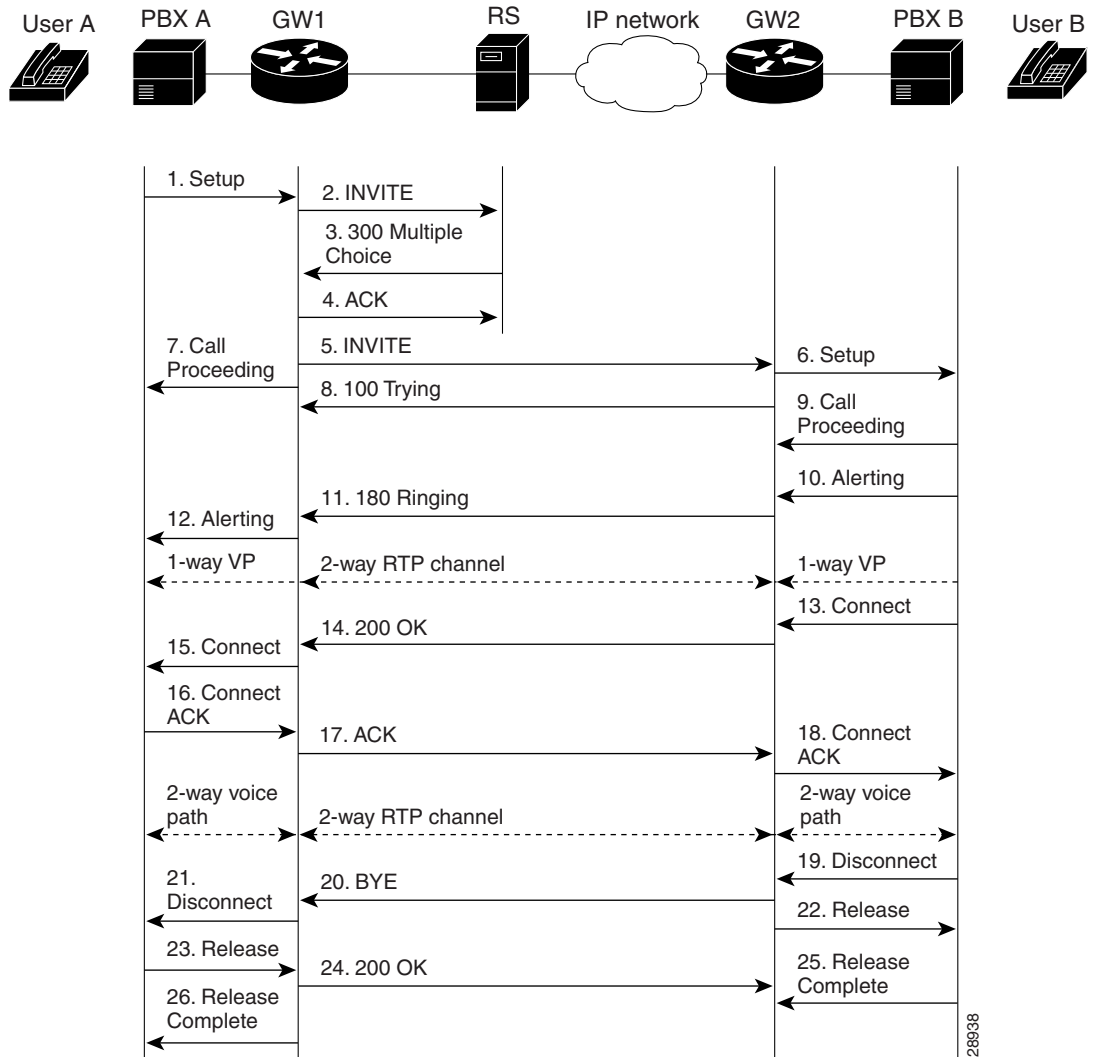
SIP Gateway-to-SIP Gateway—Call via SIP Redirect Server

Figure 8 on page 13 shows a successful gateway-to-gateway call setup and disconnect via a SIP redirect server. In this scenario, the two end users are identified as User A and User B. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. SIP gateway 1 is using a SIP redirect server. User B is located at PBX B. PBX B is connected to SIP gateway 2 via a T1/E1. User B's phone number is 555-0100. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. User A calls User B through the SIP gateway 1 using a SIP redirect server.
2. User B answers the call.
3. User B hangs up.

Figure 8 SIP Gateway-to-SIP Gateway—Call via SIP Redirect Server



The following processes occur in [Figure 8](#).

Process	Description
1. Setup—PBX A to SIP gateway 1	Call setup is initiated between PBX A and SIP gateway 1. Setup includes the standard transactions that take place as User A attempts to call User B.
2. INVITE—SIP gateway 1 to SIP redirect server	<p>SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request the following is the case:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is a domain (with or without a hostname) or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0100@example.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3. 300 Multiple Choice—SIP redirect server to SIP gateway 1	The SIP redirect server sends a 300 Multiple Choice response to SIP gateway 1. The 300 Multiple Choice response indicates that the SIP redirect server accepted the INVITE request, contacted a location server with all or part of User B’s SIP URL, and the location server provided a list of alternative locations where User B might be located. The SIP redirect server returns these possible addresses to SIP gateway 1 in the 300 Multiple Choice response.
4. ACK—SIP gateway 1 to SIP redirect server	SIP gateway 1 acknowledges the 300 Multiple Choice response with an ACK.
5. INVITE—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends a new INVITE request to SIP gateway 2. The new INVITE request includes the first contact listed in the 300 Multiple Choice response as the new address for User B, a higher transaction number in the CSeq field, and the same Call-ID as the first INVITE request.
6. Setup—SIP gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates call setup with User B through PBX B.
7. Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the setup request.

Process	Description
8. 100 Trying—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located.
9. Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the setup request.
10. Alerting—PBX B to SIP gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B's phone begins to ring.
11. 180 Ringing—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.
12. Alerting—SIP gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. User A hears ringback tone. At this point, a one-way voice path is established between SIP gateway 1 and PBXA and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.
13. Connect—PBX B to SIP gateway 2	User B answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
14. 200 OK—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.
15. Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
16. Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
17. ACK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 200 OK response has been received. The call is now in progress over a two-way voice path via RTP.
18. Connect ACK—SIP gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message. At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.
19. Disconnect—PBX B to SIP gateway 2	Once User B hangs up, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.

Process	Description
20. BYE—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
21. Disconnect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
22. Release—SIP gateway 2 to PBX B	SIP gateway 2 sends a Release message to PBX B.
23. Release—PBX A to SIP gateway 1	PBX A sends a Release message to SIP gateway 1.
24. 200 OK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
25. Release Complete—PBX B to SIP gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
26. Release Complete—SIP gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the session is terminated.

SIP Gateway-to-SIP Gateway—Call via SIP Proxy Server

Figure 9 and Figure 10 show a successful gateway-to-gateway call setup and disconnect via a proxy server. The two end users are User A and User B. User A is located at PBX A, which is connected to SIP gateway 1 via a T1/E1. SIP gateway 1 is using a proxy server. SIP gateway 1 is connected to SIP gateway 2 over an IP network. User B is located at PBX B, which is connected to SIP gateway 2 (a SIP gateway) via a T1/E1. User B's phone number is 555-0100.



Note

The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy.

In [Figure 9](#), the record route feature is enabled on the proxy server. In [Figure 10](#), record route is disabled on the proxy server.

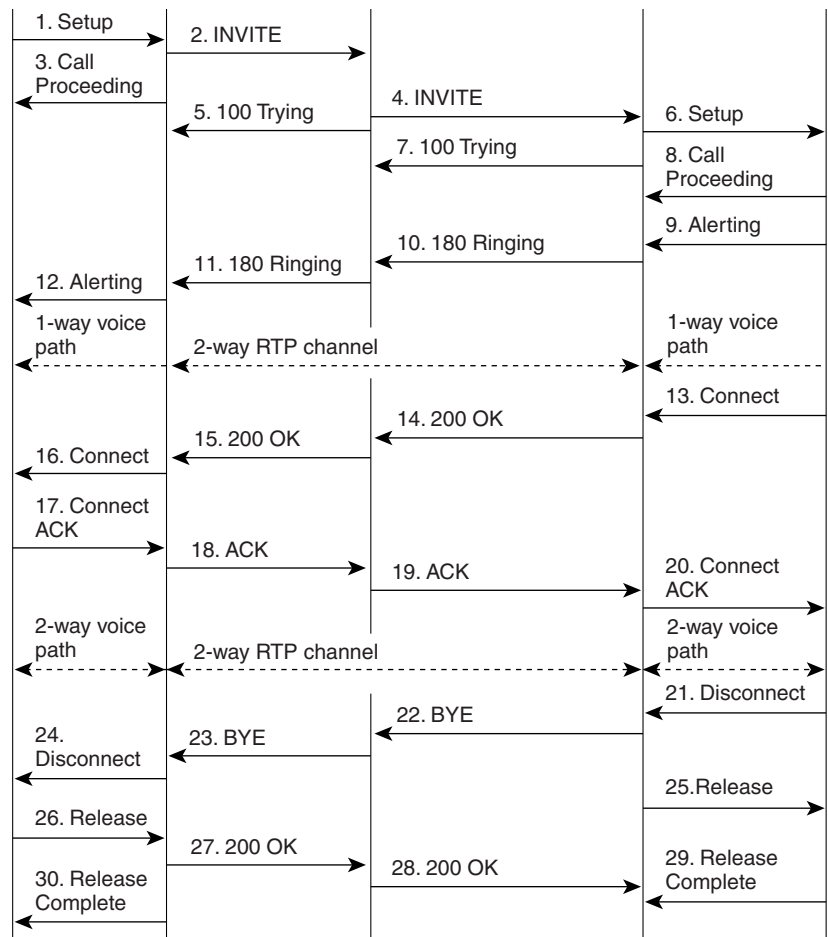
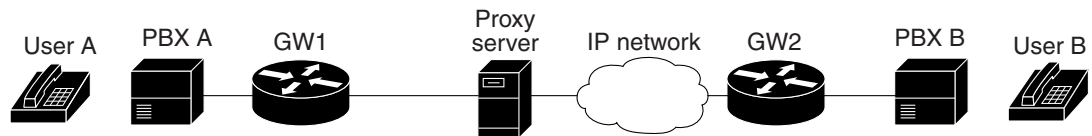
When record route is enabled, the proxy server adds the Record-Route header to the SIP messages to ensure that it is in the path of subsequent SIP requests for the same call leg. The Record-Route field contains a globally reachable Request-URI that identifies the proxy server. When record route is enabled, each proxy server adds its Request-URI to the beginning of the list.

When record route is disabled, SIP messages flow directly through the SIP gateways once a call has been established.

The call flow is as follows:

1. User A calls User B via SIP gateway 1 using a proxy server.
2. User B answers the call.
3. User B hangs up.

Figure 9 SIP Gateway-to-SIP Gateway—Call via SIP Proxy Server with Record Route Enabled



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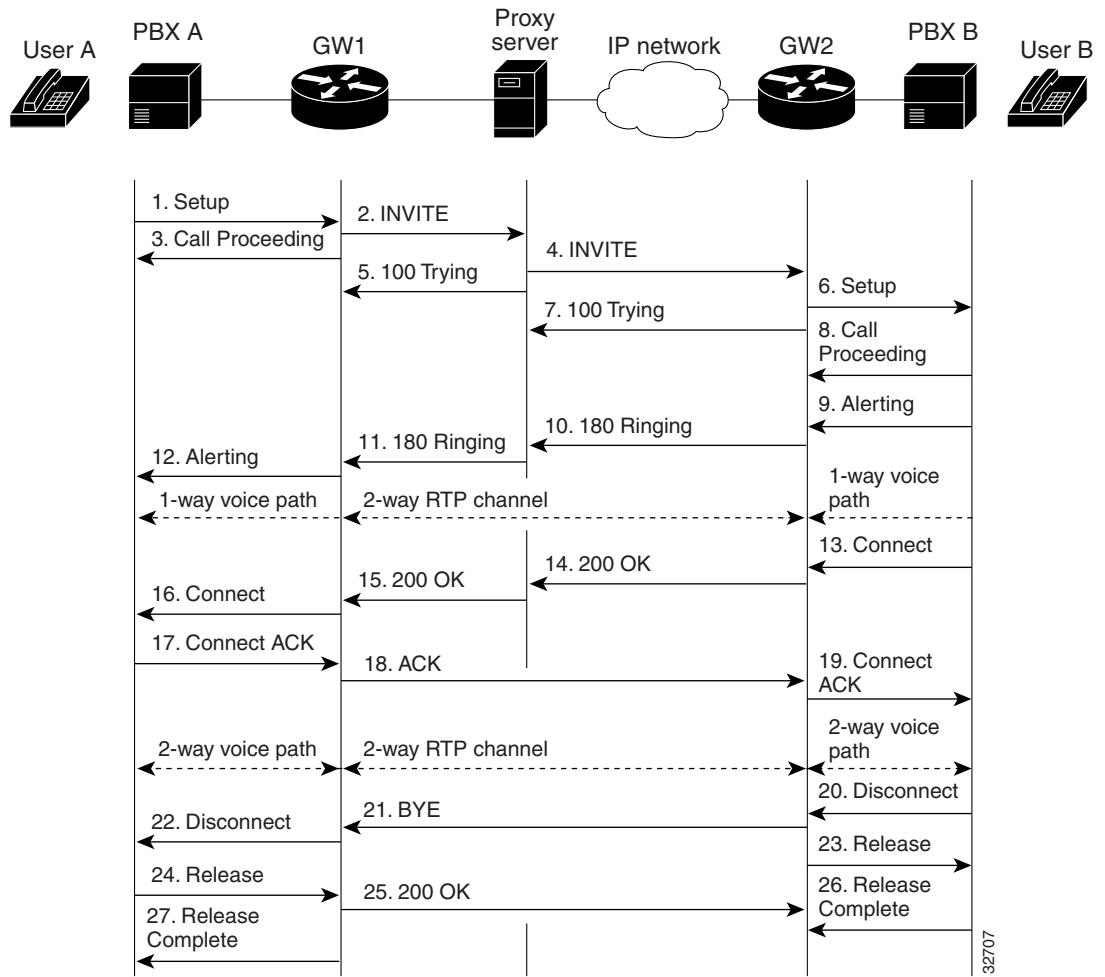
The following processes occur in [Figure 9](#).

Process	Description
1. Setup—PBX A to SIP gateway 1	Call setup is initiated between PBX A and SIP gateway 1. Setup includes the standard transactions that take place as User A attempts to call User B.
2. INVITE—SIP gateway 1 to proxy server	<p>SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is a domain (with or without a hostname) or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0100@example.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3. Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the setup request.
4. INVITE—SIP proxy server to SIP gateway 2	The SIP proxy server checks whether its own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.
5. 100 Trying—SIP proxy server to SIP gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
6. Setup—SIP gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates call setup with User B via PBX B.
7. 100 Trying—SIP gateway 2 to SIP proxy server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server. The SIP proxy server might or might not forward the 100 Trying response to SIP gateway 1.
8. Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the setup request.

Process	Description
9. Alerting—PBX B to SIP gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B's phone begins to ring.
10. 180 Ringing—SIP gateway 2 to SIP proxy server	SIP gateway 2 sends a 180 Ringing response to the SIP proxy server.
11. 180 Ringing—SIP proxy server to SIP gateway 1	The SIP proxy server forwards the 180 Ringing response to SIP gateway 1.
12. Alerting—SIP gateway 1 to PBX A	<p>SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response. User A hears the ringback tone that indicates that User B is being alerted.</p> <p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>
13. Connect—PBX B to SIP gateway 2	User B answers the phone. PBX B sends a Connect message to SIP gateway 2. The connect message notifies SIP gateway 2 that the connection has been made.
14. 200 OK—SIP gateway 2 to SIP proxy server	<p>SIP gateway 2 sends a 200 OK response (including the Record-Route header received in the INVITE) to the SIP proxy server. The 200 OK response notifies the SIP proxy server that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by the SIP proxy server, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p> <p>The SIP proxy server must forward 200 OK responses upstream.</p>
15. 200 OK—SIP proxy server to SIP gateway 1	The SIP proxy server forwards the 200 OK response that it received from SIP gateway 2 to SIP gateway 1. In the 200 OK response, the SIP proxy server forwards the Record-Route header to ensure that it is in the path of subsequent SIP requests for the same call leg. In the Record-Route field, the SIP proxy server adds its Request-URI.
16. Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
17. Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
18. ACK—SIP gateway 1 to SIP proxy server	SIP gateway 1 sends an ACK to the SIP proxy server. The ACK confirms that SIP gateway 1 has received the 200 OK response from the SIP proxy server.

Process	Description
19. ACK—SIP proxy server to SIP gateway 2	<p>Depending on the values in the To, From, CSeq, and Call-ID field, the SIP proxy server might process the ACK locally or proxy it. If the fields in the ACK match those in previous requests processed by the SIP proxy server, the server proxies the ACK. If there is no match, the ACK is proxied as if it were an INVITE request.</p> <p>The SIP proxy server forwards SIP gateway 1's ACK response to SIP gateway 2.</p>
20. Connect ACK—SIP gateway 2 to PBX B	<p>SIP gateway 2 acknowledges PBX B's Connect message. The call session is now active.</p> <p>The 2-way voice path is established directly between SIP gateway 1 and SIP gateway 2; not via the SIP proxy server.</p> <p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>
21. Disconnect—PBX B to SIP gateway 2	After the call is completed, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.
22. BYE—SIP gateway 2 to SIP proxy server	SIP gateway 2 sends a BYE request to the SIP proxy server. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
23. BYE—SIP proxy server to SIP gateway 1	The SIP proxy server forwards the BYE request to SIP gateway 1.
24. Disconnect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
25. Release—SIP gateway 2 to PBX B	After the call is completed, SIP gateway 2 sends a Release message to PBX B.
26. Release—PBX A to SIP gateway 1	PBX A sends a Release message to SIP gateway 1.
27. 200 OK—SIP gateway 1 to SIP proxy server	SIP gateway 1 sends a 200 OK response to the SIP proxy server. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
28. 200 OK—SIP proxy server to SIP v	The SIP proxy server forwards the 200 OK response to SIP gateway 2.
29. Release Complete—PBX B to SIP gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
30. Release Complete—SIP gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

Figure 10 SIP Gateway-to-SIP Gateway—Call via a Proxy Server with Record Route Disabled



The following processes occur in [Figure 10](#).

Process	Description
1. Setup—PBX A to SIP gateway 1	Call setup is initiated between PBX A and SIP gateway 1. Setup includes the standard transactions that take place as User A attempts to call User B.
2. INVITE—SIP gateway 1 to SIP proxy server	<p>SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request the following is the case:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is a domain (with or without a hostname) or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0100@example.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3. Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the setup request.
4. INVITE—SIP proxy server to SIP gateway 2	The SIP proxy server checks whether its own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.
5. 100 Trying—SIP proxy server to SIP gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
6. Setup—SIP gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates call setup with User B via PBX B.
7. 100 Trying—SIP gateway 2 to SIP proxy server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server. The SIP proxy server might or might not forward the 100 Trying response to SIP gateway 1.
8. Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge setup request.
9. Alerting—PBX B to SIP gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begins to ring.

Process	Description
10. 180 Ringing—SIP gateway 2 to SIP proxy server	SIP gateway 2 sends a 180 Ringing response to the SIP proxy server.
11. 180 Ringing—SIP proxy server to SIP gateway 1	The SIP proxy server forwards the 180 Ringing response to SIP gateway 1.
12. Alerting—SIP gateway 1 to PBX A	<p>SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response. User A hears the ringback tone that indicates that User B is being alerted.</p> <p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>
13. Connect—PBX B to SIP gateway 2	User B answers the phone. PBX B sends a Connect message to SIP gateway 2. The connect message notifies SIP gateway 2 that the connection has been made.
14. 200 OK—SIP gateway 2 to SIP proxy server	<p>SIP gateway 2 sends a 200 OK response to the SIP proxy server. The 200 OK response notifies the SIP proxy server that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by the SIP proxy server, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p> <p>The SIP proxy server must forward 200 OK responses upstream.</p>
15. 200 OK—SIP proxy server to SIP gateway 1	The SIP proxy server forwards the 200 OK response that it received from SIP gateway 2 to SIP gateway 1.
16. Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
17. Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
18. ACK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from the SIP proxy server.
19. Connect ACK—SIP gateway 2 to PBX B	<p>SIP gateway 2 acknowledges PBX B's Connect message. The call session is now active.</p> <p>The 2-way voice path is established directly between SIP gateway 1 and SIP gateway 2; not via the SIP proxy server.</p> <p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>

Process	Description
20. Disconnect—PBX B to SIP gateway 2	After the call is completed, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.
21. BYE—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
22. Disconnect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
23. Release—SIP gateway 2 to PBX B	After the call is completed, SIP gateway 2 sends a Release message to PBX B.
24. Release—PBX A to SIP gateway 1	PBX A sends a Release message to SIP gateway 1.
25. 200 OK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
26. Release Complete—PBX B to SIP gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
27. Release Complete—SIP gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

Additional References

The following sections provide references related to SIP.



Note

- In addition to the references listed below, each chapter provides additional references related to SIP.
- Some of the products and services mentioned in this guide may have reached end of life, end of sale, or both. Details are available at http://www.cisco.com/en/US/products/prod_end_of_life.html.

Related Documents

Related Topic	Document Title
Basic router configuration	<ul style="list-style-type: none"> • Cisco 2600 series documentation at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis2600/index.htm • Cisco 3600 series documentation at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3600/index.htm • Cisco 3700 series documentation at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3700/index.htm • Cisco AS5300 documentation at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/5300/index.htm
Cisco IOS command references	<ul style="list-style-type: none"> • <i>Cisco IOS Debug Command Reference, Release 12.3T</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123dbr/index.htm • <i>Cisco IOS Voice Command Reference, Release 12.3T</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123tvr/index.htm
Cisco IOS configuration fundamentals and examples	<ul style="list-style-type: none"> • <i>Cisco IOS Configuration Fundamentals Configuration Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/ffun_c/ • <i>Cisco IOS Interface Command Reference</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/finter_r/index.htm • <i>Cisco IOS Interface Configuration Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/finter_c/ • <i>Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvfax_c/index.htm • Cisco Systems Technologies website at http://cisco.com/en/US/tech/index.html From the website, select a technology category and subsequent hierarchy of subcategories, then click Technical Documentation > Configuration Examples.
Cisco IOS Voice Configuration Library, including library preface and glossary	<ul style="list-style-type: none"> • Cisco IOS Voice Configuration Library at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm
Developer support	<ul style="list-style-type: none"> • Developer Support Agreement at http://www.cisco.com/go/developersupport
IVR script information	<ul style="list-style-type: none"> • <i>TCL IVR API 2.0 Programming Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/vapp_dev/tclivr2/index.htm
NAT configuration	<ul style="list-style-type: none"> • <i>Configuring Network Address Translation: Getting Started</i> at http://www.cisco.com/warp/public/556/12.htm

Related Topic	Document Title
SIP documents	<ul style="list-style-type: none"> • Cisco SIP proxy server documents at http://www.cisco.com/univercd/cc/td/doc/product/voice/sipproxy/index.htm • <i>Guide to Cisco Systems' VoIP Infrastructure Solution for SIP</i> at http://www.cisco.com/univercd/cc/td/doc/product/voice/sipsols/biggulp/index.htm • <i>Session Initiation Protocol Gateway Call Flows</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/rel_docs/sip_flo/index.htm
SS7 for voice gateways	<ul style="list-style-type: none"> • <i>Configuring Media Gateways for the SS7 Interconnect for Voice Gateways Solution</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/sc/rel7/soln/das22/gateway/dascfg5.htm
Tcl IVR programming	<ul style="list-style-type: none"> • <i>Tcl IVR API Version 2.0 Programmer's Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/vapp_dev/tclivr2/index.htm
Troubleshooting	<ul style="list-style-type: none"> • <i>Cisco IOS Debug Command Reference, Release 12.3T</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123dbr/index.htm • <i>Cisco IOS Voice Troubleshooting and Monitoring Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/index.htm • <i>Internetwork Troubleshooting Guide</i> at http://www.cisco.com/univercd/cc/td/doc/cisintwk/itg_v1/index.htm • <i>Troubleshooting and Debugging VoIP Call Basics</i> at http://www.cisco.com/warp/public/788/voip/voip_debugcalls.html • <i>Voice over IP Troubleshooting and Monitoring</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/index.htm • <i>VoIP Debug Commands</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/1700/1750/1750voip/debug.htm
VoATM configuration	<ul style="list-style-type: none"> • <i>Configuring AAL2 and AAL5 for the High-Performance Advanced Integration Module on the Cisco 2600 Series</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122limit/122x/122xa/122xa_2/ft_ataim.htm
VoIP configuration	<ul style="list-style-type: none"> • <i>Voice over IP for the Cisco 2600/3600 Series</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip3600/index.htm • <i>Voice over IP for the Cisco AS5300</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip5300/index.htm • <i>Voice over IP for the Cisco AS5800</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip5800/index.htm
VSA information	<ul style="list-style-type: none"> • <i>RADIUS Vendor-Specific Attributes Voice Implementation Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/vapp_dev/vsaig3.htm

Related Topic	Document Title
WAN configuration	<ul style="list-style-type: none">• <i>Cisco IOS Wide-Area Networking Command Reference</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fwan_r/index.htm• <i>Cisco IOS Wide-Area Networking Configuration Guide</i> at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fwan_c/wfatm.htm
Other documents	<p><i>Cisco Internetworking Terms and Acronyms</i> at http://www.cisco.com/univercd/cc/td/doc/cisintwk/ita/</p> <p><i>Cisco Resource Policy Management System 2.0</i> at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_soft/rpms/rpms_2-0/</p> <p><i>VoIP Call Admission Control</i> at http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/cac.htm</p>

Standards

Standards ¹	Title
ANSI TI.619/a	<i>ISDN Multi-Level Precedence and Preemption (MLPP) Service Capability</i>
draft-ietf-avt-profile-new-12.txt	<i>RTP Profile for Audio and Video Conferences with Minimal Control</i>
draft-ietf-avt-rtp-cn-06.txt	<i>RTP Payload for Comfort Noise, Internet Draft of the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group</i>
draft-ietf-avt-rtp-mime-06.txt	<i>MIME Type Registration of RTP Payload Formats</i>
draft-ietf-mmusic-sdp-comedia-04.txt	<i>Connection-Oriented Media Transport in SDP</i>
draft-ietf-sipping-reason-header-for-preemption-00	<i>Extending the SIP for Preemption Events</i>
draft-ietf-sip-privacy-02	<i>SIP Extensions for Caller Identity and Privacy</i>
draft-ietf-sip-resource-priority-05	<i>Communications Resources Priority for SIP</i>
draft-levy-diversion-06.txt	[Sip] verification of diversion header (draft-levy)
GR-268-CORE	<i>ISDN Basic Rate Interface Call Control Switching and Signalling Generic Requirements</i>

1. Not all supported standards are listed.

MIBs

MIBs	MIBs Link
CISCO-SIP-UA-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC ¹	Title
<ul style="list-style-type: none"> RFC 1889 (obsoleted by RFC 3550 in July 2003) 	RTP: A Transport Protocol for Real-Time Applications
<ul style="list-style-type: none"> RFC 2052 (obsoleted by RFC 2782 in Feb. 2000) 	A DNS RR for Specifying location of services (DNS SRV)
<ul style="list-style-type: none"> RFC 2543 (and RFC 2543-bis-04) (obsoleted by RFCs 3261, 3262, 3263, 3264, and 3265 in June 2002) 	SIP: Session Initiation Protocol
<ul style="list-style-type: none"> RFC 2617 	HTTP Authentication: Basic and Digest Access Authentication
<ul style="list-style-type: none"> RFC 2782 (replaced RFC 2052 in Feb. 2000) 	A DNS RR for specifying the location of services (DNS SRV)
<ul style="list-style-type: none"> RFC 2806 	URLs for Telephone Calls

RFC ¹	Title
<ul style="list-style-type: none"> • RFC 2833 (obsoleted by RFCs 4733 and 4734 in Dec. 2006) 	<i>RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals</i>
<ul style="list-style-type: none"> • RFC 2976 	<i>SIP INFO Method</i>
<ul style="list-style-type: none"> • RFC 3261 (replaced RFC 2543 in June 2002 and updated by RFCs 3853 (July 2004) and 4320 (Jan. 2006)) 	<i>SIP: Session Initiation Protocol</i>
<ul style="list-style-type: none"> • RFC 3262 (replaced RFC 2543 in June 2002) 	<i>Reliability of Provisional Responses in Session Initiation Protocol (SIP)</i>
<ul style="list-style-type: none"> • RFC 3263 (replaced RFC 2543 in June 2002) 	<i>Session Initiation Protocol (SIP): Locating SIP Servers</i>
<ul style="list-style-type: none"> • RFC 3264 (replaced RFC 2543 in June 2002) 	<i>An Offer/Answer Model with Session Description Protocol (SDP)</i>
<ul style="list-style-type: none"> • RFC 3265 (replaced RFC 2543 in June 2002) 	<i>Session Initiation Protocol (SIP)-Specific Event Notification</i>
<ul style="list-style-type: none"> • RFC 3311 	<i>The Session Initiation Protocol (SIP) UPDATE Method</i>
<ul style="list-style-type: none"> • RFC 3312 (updated by RFC 4032 in March 2005) 	<i>Integration of Resource Management and Session Initiation Protocol (SIP)</i>
<ul style="list-style-type: none"> • RFC 3326 	<i>The Reason Header Field for the Session Initiation Protocol</i>
<ul style="list-style-type: none"> • RFC 3420 	<i>Internet Media Type message/sipfrag</i>
<ul style="list-style-type: none"> • RFC 3515 	<i>The Session Initiation Protocol (SIP) Refer Method</i>
<ul style="list-style-type: none"> • RFC 3550 (replaced RFC 1889 in July 2003) 	<i>RTP: A Transport Protocol for Real-Time Applications</i>
<ul style="list-style-type: none"> • RFC 3853 (updated RFC 3261 in July 2004) 	<i>S/MIME Advanced Encryption Standard (AES) Requirement for the Session Initiation Protocol (SIP)</i>
<ul style="list-style-type: none"> • RFC 4032 (updated RFC 3312 in March 2005) 	<i>Update to the Session Initiation Protocol (SIP) Preconditions Framework</i>
<ul style="list-style-type: none"> • RFC 4320 (updated RFC 3261 in July 2004) 	<i>Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction</i>
<ul style="list-style-type: none"> • RFC 4733 (replaced RFC 2833 and was updated by RFC 4734 in Dec. 2006) 	<i>RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals</i>
<ul style="list-style-type: none"> • RFC 4734 (replaced RFC 2833 and updated RFC 4733 in Dec. 2006) 	<i>Definition of Events for Modem, Fax, and Text Telephony Signals</i>

1. Not all supported RFCs are listed.

Technical Assistance

Description	Link
The Cisco Technical Support website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

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Basic SIP Configuration

This chapter provides basic configuration information for the following features:

- SIP Register Support
- SIP Redirect Processing Enhancement
- SIP 300 Multiple Choice Messages
- SIP implementation enhancements:
 - Interaction with Forking Proxies
 - SIP Intra-Gateway Hairpinning

Feature History for SIP Register Support, SIP Redirect Processing Enhancement, and SIP 300 Multiple Choice Messages

Release	Modification
12.2(15)ZJ	This feature was introduced.
12.3(4)T	This feature was integrated into the release.

Feature History for SIP Implementation Enhancements: Interaction with Forking Proxies and SIP Intra-Gateway Hairpinning

Release	Modification
12.2(2)XB	These features were introduced.
12.2(8)T	This feature were integrated into the release.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.



Contents

- [Prerequisites for Basic SIP Configuration, page 2](#)
- [Information About Basic SIP Configuration, page 2](#)
- [How to Perform Basic SIP Configuration, page 4](#)
- [Configuration Examples for Basic SIP Configuration, page 20](#)
- [Toll Fraud Prevention, page 28](#)
- [Additional References, page 29](#)

Prerequisites for Basic SIP Configuration

SIP Redirect Processing Enhancement Feature

- Ensure that your SIP gateway supports 300 or 302 Redirect messages.

Information About Basic SIP Configuration

To perform basic SIP configuration tasks, you should understand the following concepts:

- [SIP Register Support, page 2](#)
- [SIP Redirect Processing Enhancement, page 3](#)
- [Sending SIP 300 Multiple Choice Messages, page 4](#)

SIP Register Support

With H.323, Cisco IOS gateways can register E.164 numbers of a POTS dial peer with a gatekeeper, which informs the gatekeeper of a user's contact information. Session Initiation Protocol (SIP) gateways allow the same functionality, but with the registration taking place with a SIP proxy or registrar. SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.

When registering dial peers with an external registrar, you can also register with a secondary SIP proxy or registrar to provide redundancy. The secondary registration can be used if the primary registrar fails.

SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar server on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones. By default, SIP gateways do not generate SIP Register messages. The following tasks set up the gateway to register E.164 telephone numbers with an external SIP registrar.



Note

There are no commands that allow registration between the H.323 and SIP protocols.

SIP Redirect Processing Enhancement

SIP Redirect Processing allows flexibility in the handling of incoming redirect or 3xx class of responses. Redirect responses can be enabled or disabled through the command-line interface, providing a benefit to service providers who deploy Cisco SIP gateways. Redirect processing is active by default, which means that SIP gateways handle incoming 3xx messages in compliance with RFC 2543. RFC 2543 states that redirect response messages are used by SIP user agents to initiate a new Invite when a user agent learns that a user has moved from a previously known location.

In accordance with RFC 2543-bis-04, the processing of 3xx redirection is as follows:

- The uniform resource identifier (URI) of the redirected INVITE is updated to contain the new contact information provided by the 3xx redirect message.
- The transmitted CSeq number found in the CSeq header is increased by one. The new INVITE includes the updated CSeq.
- The To, From, and Call ID headers that identify the call leg remain the same. The same Call ID gives consistency when capturing billing history.
- The UAC retries the request at the new address given by the 3xx Contact header field.

Redirect handling can be disabled by using the **no redirection** command in SIP user-agent configuration mode. In this case, the user agent treats incoming 3xx responses as 4xx error class responses. The call is not redirected, and is instead released with the appropriate PSTN cause-code message. [Table 1](#) shows the mapping of 3xx responses to 4xx responses.

Table 1 Mapping of 3xx Responses to 4xx Responses

Redirection (3xx) Response Message	Mapping to 4xx (Client Error) Response
300 Multiple choices	410 Gone
301 Moved Permanently	410 Gone
302 Moved Temporarily	480 Temporarily Unavailable
305 Use Proxy	410 Gone
380 Alternative Service	410 Gone
<any other 3xx response>	410 Gone

SIP Redirect Processing generates call history information with appropriate release cause codes that maybe used for accounting or statistics purposes. When a 3xx response is mapped to 4xx class of response, the cause code stored in call history is based on the mapped 4xx response code.

Call redirection must be enabled on the gateway for SIP call transfer involving redirect servers to be successful.

The Cisco IOS voice gateway can also use call redirection if an incoming VoIP call matches an outbound VoIP dial peer. The gateway sends a 300 or 302 Redirect message to the call originator, allowing the originator to reestablish the call. Two commands allow you to enable the redirect functionality, globally or on a specific inbound dial peer: **redirect ip2ip (dial-peer)** and **redirect ip2ip (voice service)**.

Sending SIP 300 Multiple Choice Messages

Originally, when a call was redirected, the SIP gateway would send a 302 Moved Temporarily message. The first longest match route on a gateway (dial-peer destination pattern) was used in the Contact header of the 302 message. Now, if multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a 300 Multiple Choice message, and the multiple routes in the Contact header are listed.

The **redirect contact order** command gives you the flexibility to choose the order in which routes appear in the Contact header.

How to Perform Basic SIP Configuration

This section contains the following procedures:

- [Configuring SIP VoIP Services on a Cisco Gateway, page 4](#)
- [Configuring SIP Register Support, page 6](#)
- [Configuring SIP Redirect Processing Enhancement, page 8](#)
- [Configuring SIP 300 Multiple Choice Messages, page 11](#)
- [Configuring SIP Implementation Enhancements, page 12](#)
 - [Interaction with Forking Proxies, page 13](#)
 - [SIP Intra-Gateway Hairpinning, page 13](#)
- [Verifying SIP Gateway Status, page 14](#)

**Note**

For help with a procedure, see the verification and troubleshooting sections listed above.

Configuring SIP VoIP Services on a Cisco Gateway

This section contains the following procedures:

- [Shut Down or Enable VoIP Service on Cisco Gateways, page 4](#)
- [Shut Down or Enable VoIP Submodes on Cisco Gateways, page 5](#)

Shut Down or Enable VoIP Service on Cisco Gateways

To shut down or enable VoIP service on Cisco gateways, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **[no] shutdown**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service VoIP configuration mode.
Step 4	[no] shutdown [forced] Example: Router(config-voi-serv)# shutdown forced	Shuts down or enables VoIP call services.
Step 5	exit Example: Router(config-voi-serv)# exit	Exits the current mode.

Shut Down or Enable VoIP Submodes on Cisco Gateways

To shut down or enable VoIP submodes on Cisco gateways, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **[no] call service stop**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service VoIP configuration mode.
Step 4	sip Example: Router(config-voi-serv)# sip	Enters SIP configuration mode.
Step 5	[no] call service stop [forced] [maintain-registration] Example: Router(conf-serv-sip)# call service stop maintain-registration	Shuts down or enables VoIP call services for the selected submode.
Step 6	exit Example: Router(conf-serv-sip)# exit	Exits the current mode.

Configuring SIP Register Support

To configure SIP register support, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registrar**
5. **retry register**
6. **timers register**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	registrar {dns:address ipv4:destination-address} expires seconds [tcp] [secondary] Example: Router(config-sip-ua)# registrar ipv4:10.8.17.40 expires 3600 secondary	Registers E.164 numbers on behalf of analog telephone voice ports (FXS) and IP phone virtual voice ports (EFXS) with an external SIP proxy or SIP registrar server. Keywords and arguments are as follows: <ul style="list-style-type: none"> • dns:address—Domain-name server that resolves the name of the dial peer to receive calls. • ipv4:destination-address—IP address of the dial peer to receive calls. • expires seconds—Default registration time, in seconds. • tcp—Sets transport layer protocol to TCP. UDP is the default. • secondary—Specifies registration with a secondary SIP proxy or registrar for redundancy purposes. Optional.
Step 5	retry register number Example: Router(config-sip-ua)# retry register 10	Use this command to set the total number of SIP Register messages that the gateway should send. The argument is as follows: <ul style="list-style-type: none"> • number—Number of Register message retries. Range: 1 to 10. Default: 10.
Step 6	timers register milliseconds Example: Router(config-sip-ua)# timers register 500	Use this command to set how long the SIP user agent waits before sending register requests. The argument is as follows: <ul style="list-style-type: none"> • milliseconds—Waiting time, in ms. Range: 100 to 1000. Default: 500.
Step 7	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP Redirect Processing Enhancement

This section provides the following information:

- [Configure Call-Redirect Processing Enhancement, page 8](#)
- [Configuring SIP 300 Multiple Choice Messages, page 11](#)

Configure Call-Redirect Processing Enhancement

Redirect processing using the **redirection** command is enabled by default. To disable and then reset redirect processing, perform the steps listed in this section:

- [Configuring Call-Redirect Processing Enhancement, page 8](#)

IP-to-IP call redirection can be enabled globally or on a dial-peer basis. To configure, perform the steps listed in these sections:

- [Configuring Call Redirect to Support Calls Globally, page 9](#)
- [Configuring Call Redirect to Support Calls on a Specific VoIP Dial Peer, page 10](#)

Configuring Call-Redirect Processing Enhancement

To configure call-redirect processing enhancement, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **no redirection**
5. **redirection**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
Step 4	no redirection Example: Router(config-sip-ua)# no redirection	Disables redirect handling—causes the gateway to treat incoming 3xx responses as 4xx error class responses.
Step 5	redirection Example: Router(config-sip-ua)# redirection	Resets call redirection to work as specified in RFC 2543. The command default redirection also resets call redirection to work as specified in RFC 2543.
Step 6	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring Call Redirect to Support Calls Globally

To configure call redirect to support calls globally, perform the following steps.



Note

To enable global IP-to-IP call redirection for all VoIP dial peers, use voice-service configuration mode. The default SIP application supports IP-to-IP redirection.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **redirect ip2ip**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service VoIP configuration mode.

	Command or Action	Purpose
Step 4	redirect ip2ip Example: Router(conf-voi-serv)# redirect ip2ip	Redirect SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS voice gateway.
Step 5	exit Example: Router(conf-voi-serv)# exit	Exits the current mode.

Configuring Call Redirect to Support Calls on a Specific VoIP Dial Peer

To configure call redirect to support calls on a specific VoIP dial peer, perform the following steps.



Note

- To specify IP-to-IP call redirection for a specific VoIP dial peer, configure it on an inbound dial peer in dial-peer configuration mode. The default application on SIP SRST supports IP-to-IP redirection.
- When IP-to-IP redirection is configured in dial-peer configuration mode, the configuration on the specific inbound dial peer takes precedence over the global configuration entered under voice service configuration.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **application**
5. **redirect ip2ip**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<code>dial-peer voice tag voip</code> Example: Router(config)# dial-peer voice 29 voip	Use this command to enter dial-peer configuration mode. The argument is as follows: <ul style="list-style-type: none"> <i>tag</i>—Digits that define a particular dial peer. Range: 1 to 2,147,483,647 (enter without commas).
Step 4	<code>application application-name</code> Example: Router(config-dial-peer)# application session	Enables a specific application on a dial peer. The argument is as follows: <ul style="list-style-type: none"> <i>application-name</i>—Name of the predefined application you wish to enable on the dial peer. For SIP, the default Tcl application (from the Cisco IOS image) is session and can be applied to both VoIP and POTS dial peers. The application must support IP-to-IP redirection
Step 5	<code>redirect ip2ip</code> Example: Router(conf-dial-peer)# redirect ip2ip	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS voice gateway.
Step 6	<code>exit</code> Example: Router(conf-dial-peer)# exit	Exits the current mode.

Configuring SIP 300 Multiple Choice Messages

This section contains the following information:

- [Sending SIP 300 Multiple Choice Messages, page 4](#)
- [Configuring Sending of SIP 300 Multiple Choice Messages, page 11](#)

Configuring Sending of SIP 300 Multiple Choice Messages

To send SIP 300 Multiple Choice messages, perform the following steps.



Note

If multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a 300 Multiple Choice message and the multiple routes in the Contact header are listed. This configuration allows users to choose the order in which the routes appear in the Contact header.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `redirect contact order`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: Router(config)# voice service voip	Enters voice-service VoIP configuration mode.
Step 4	<code>sip</code> Example: Router(config-voi-serv)# sip	Enters SIP configuration mode.
Step 5	<code>redirect contact order [best-match longest-match]</code> Example: Router(conf-serv-sip)# redirect contact order best-match	Sets the order of contacts in the 300 Multiple Choice Message. Keywords are as follows: <ul style="list-style-type: none"> • best-match—Use the current system configuration to set the order of contacts. • longest-match—Set the contact order by using the destination pattern longest match first, and then the second longest match, the third longest match, and so on. This is the default.
Step 6	<code>exit</code> Example: Router(conf-serv-sip)# exit	Exits the current mode.

Configuring SIP Implementation Enhancements

Minor underlying or minimally configurable features are described in the following sections:

- [Interaction with Forking Proxies, page 13](#)
- [SIP Intra-Gateway Hairpinning, page 13](#)

For additional information on SIP implementation enhancements, see [“Achieving SIP RFC Compliance.”](#)

Interaction with Forking Proxies

Call forking enables the terminating gateway to handle multiple requests and the originating gateway to handle multiple provisional responses for the same call. Call forking is required for the deployment of the *find me/follow me* type of services.

Support for call forking enables the terminating gateway to handle multiple requests and the originating gateway to handle multiple provisional responses for the same call. Interaction with forking proxies applies to gateways acting as a UAC, and takes place when a user is registered to several different locations. When the UAC sends an INVITE message to a proxy, the proxy forks the request and sends it to multiple user agents. The SIP gateway processes multiple 18X responses by treating them as independent transactions under the same call ID. When the relevant dial peers are configured for QoS, the gateway maintains state and initiates RSVP reservations for each of these independent transactions. When it receives an acknowledgment, such as a 200 OK, the gateway accepts the successful acknowledgment and destroys state for all other transactions.

The forking feature sets up RSVP for each transaction *only* if the dial peers are configured for QoS. If not, the calls proceed as best-effort.

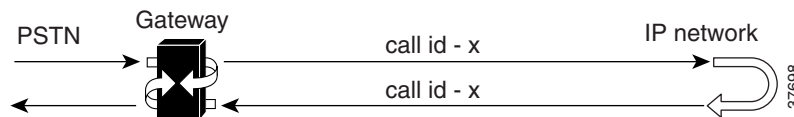
Support for interaction with forking proxies applies only to gateways acting as UACs. It does not apply when the gateway acts as a UAS. In that case, the proxy forks multiple INVITES with the same call ID to the same gateway but with different request URLs.

Also, the forking feature sets up RSVP for each transaction *only* if the dial peers are configured for QoS. If not, the calls proceed as best-effort.

SIP Intra-Gateway Hairpinning

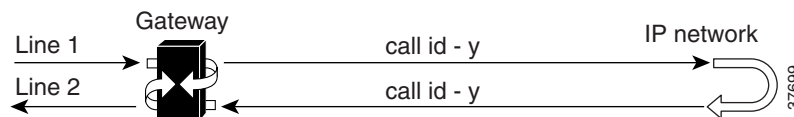
SIP hairpinning is a call routing capability in which an incoming call on a specific gateway is signaled through the IP network and back out the same gateway. This can be a PSTN call routed into the IP network and back out to the PSTN over the same gateway (see [Figure 11](#)).

Figure 11 PSTN Hairpinning Example



Similarly, SIP hairpinning can be a call signaled from a line (for example, a telephone line) to the IP network and back out to a line on the same access gateway (see [Figure 12](#)).

Figure 12 Telephone Line Hairpinning Example



With SIP hairpinning, unique gateways for ingress and egress are unnecessary.

SIP supports plain old telephone service (POTS)-to-POTS hairpinning (which means that the call comes in one voice port and is routed out another voice port). It also supports POTS-to-IP call legs and IP-to-POTS call legs. However, it does not support IP-to-IP hairpinning. This means that the SIP gateway cannot take an inbound SIP call and reroute it back to another SIP device using the VoIP dial peers.

Only minimal configuration is required for this feature. To enable hairpinning on the SIP gateway, see the following configuration example for dial peers. Note that:

- The POTS dial peer must have preference 2 defined, and the VoIP dial peer must have preference 1 defined. This ensures that the call is sent out over IP, not Plain Old Telephone Service (POTS).
- The session target is the same gateway because the call is being redirected to it.

```
!
dial-peer voice 53001 pots
preference 2
destination-pattern 5300001
prefix 5300001
!
dial-peer voice 53002 pots
preference 2
destination-pattern 5300002
prefix 5300002
!
dial-peer voice 530011 voip
preference 1
destination-pattern 5300001
session protocol sipv2
session target ipv4:10.1.1.41
playout-delay maximum 300
codec g711alaw
!
dial-peer voice 530022 voip
preference 1
destination-pattern 5300002
session protocol sipv2
session target ipv4:10.1.1.41
playout-delay maximum 300
codec g711alaw
```

Verifying SIP Gateway Status

To verify SIP gateway status and configuration, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show sip service**
2. **show sip-ua register status**
3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**

DETAILED STEPS

Step 1 **show sip service**

Use this command to display the status of SIP call service on a SIP gateway.

The following sample output shows that SIP call service is enabled:

```
Router# show sip service
```



```
SIP Service is up
```

The following sample output shows that SIP call service was shut down with the **shutdown** command:

```
Router# show sip service

SIP service is shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop** command:

```
Router# show sip service

SIP service is shut
under 'voice service voip', 'sip' submode
```

The following sample output shows that SIP call service was shut down with the **shutdown forced** command:

```
Router# show sip service

SIP service is forced shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop forced** command:

```
Router# show sip service

SIP service is forced shut
under 'voice service voip', 'sip' submode
```

Step 2 show sip-ua register status

Use this command to display the status of E.164 numbers that a SIP gateway has registered with an external primary SIP registrar.

```
Router# show sip-ua register status

Line peer expires(sec) registered
4001 20001 596 no
4002 20002 596 no
5100 1 596 no
9998 2 596 no
```

Step 3 show sip-ua statistics

Use this command to display response, traffic, and retry SIP statistics, including whether call redirection is disabled.

The following sample shows that four registers were sent:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
  Trying 0/0, Ringing 0/0,
  Forwarded 0/0, Queued 0/0,
  SessionProgress 0/0
Success:
  OkInvite 0/0, OkBye 0/0,
  OkCancel 0/0, OkOptions 0/0,
  OkPrack 0/0, OkPreconditionMet 0/0,
  OkSubscribe 0/0, OkNOTIFY 0/0,
```

```

OkInfo 0/0, 202Accepted 0/0
OkRegister 12/49
Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
MultipleChoice 0, MovedPermanently 0,
MovedTemporarily 0/0, UseProxy 0,
AlternateService 0
Client Error:
BadRequest 0/0, Unauthorized 0/0,
PaymentRequired 0/0, Forbidden 0/0,
NotFound 0/0, MethodNotAllowed 0/0,
NotAcceptable 0/0, ProxyAuthReqd 0/0,
ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
UnsupportedMediaType 0/0, BadExtension 0/0,
TempNotAvailable 0/0, CallLegNonExistent 0/0,
LoopDetected 0/0, TooManyHops 0/0,
AddrIncomplete 0/0, Ambiguous 0/0,
BusyHere 0/0, RequestCancel 0/0,
NotAcceptableMedia 0/0, BadEvent 0/0,
SETooSmall 0/0
Server Error:
InternalServerError 0/0, NotImplemented 0/0,
BadGateway 0/0, ServiceUnavail 0/0,
GatewayTimeout 0/0, BadSipVer 0/0,
PreCondFailure 0/0
Global Failure:
BusyEverywhere 0/0, Decline 0/0,
NotExistAnywhere 0/0, NotAcceptable 0/0
Miscellaneous counters:
RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
Invite 0/0, Ack 0/0, Bye 0/0,
Cancel 0/0, Options 0/0,
Prack 0/0, Comet 0/0,
Subscribe 0/0, NOTIFY 0/0,
Refer 0/0, Info 0/0
Register 49/16

Retry Statistics
Invite 0, Bye 0, Cancel 0, Response 0,
Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
Register 4

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0
Last time SIP Statistics were cleared: <never>

```

The following sample output shows the `RedirectResponseMappedToClientError` status message. An incremented number indicates that 3xx responses are to be treated as 4xx responses. When call redirection is enabled (default), the `RedirectResponseMappedToClientError` status message is not incremented.

```

Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
Trying 0/0, Ringing 0/0,
Forwarded 0/0, Queued 0/0,
SessionProgress 0/0
Success:

```

```

OkInvite 0/0, OkBye 0/0,
OkCancel 0/0, OkOptions 0/0,
OkPrack 0/0, OkPreconditionMet 0/0,
OKSubscribe 0/0, OkNotify 0/0,
202Accepted 0/0
Redirection (Inbound only):
MultipleChoice 0, MovedPermanently 0,
MovedTemporarily 0, UseProxy 0,
AlternateService 0
Client Error:
BadRequest 0/0, Unauthorized 0/0,
PaymentRequired 0/0, Forbidden 0/0,
NotFound 0/0, MethodNotAllowed 0/0,
NotAcceptable 0/0, ProxyAuthReqd 0/0,
ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
UnsupportedMediaType 0/0, BadExtension 0/0,
TempNotAvailable 0/0, CallLegNonExistent 0/0,
LoopDetected 0/0, TooManyHops 0/0,
AddrIncomplete 0/0, Ambiguous 0/0,
BusyHere 0/0, RequestCancel 0/0
NotAcceptableMedia 0/0, BadEvent 0/0
Server Error:
InternalError 0/0, NotImplemented 0/0,
BadGateway 0/0, ServiceUnavail 0/0,
GatewayTimeout 0/0, BadSipVer 0/0,
PreCondFailure 0/0
Global Failure:
BusyEverywhere 0/0, Decline 0/0,
NotExistAnywhere 0/0, NotAcceptable 0/0
Miscellaneous counters:
RedirectResponseMappedToClientError 1,
SIP Total Traffic Statistics (Inbound/Outbound)
Invite 0/0, Ack 0/0, Bye 0/0,
Cancel 0/0, Options 0/0,
Prack 0/0, Comet 0/0,
Subscribe 0/0, Notify 0/0,
Refer 0/0

Retry Statistics
Invite 0, Bye 0, Cancel 0, Response 0,
Prack 0, Comet 0, Reliable1xx 0, Notify 0

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0

```

Step 4 **show sip-ua status**

Use this command to display status for the SIP user agent (UA), including whether call redirection is enabled or disabled.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
Redirection (3xx) message handling: ENABLED

```

Step 5 **show sip-ua timers**

Use this command to display the current settings for the SIP user-agent (UA) timers.

The following sample output shows the waiting time before a register request is sent—that is, the value that is set with the **timers register** command:

```
Router# show sip-ua timers

SIP UA Timer Values (milliseconds)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 500, register 500
```

General Troubleshooting Tips

**Note**

For more information on troubleshooting, see the following references:

- *Cisco IOS Voice Troubleshooting and Monitoring Guide*
- Cisco Technical Support at <http://www.cisco.com/en/US/support/index.html>
- *Cisco IOS Debug Command Reference*
- *Cisco IOS Voice, Video, and Fax Configuration Guide*
- *Troubleshooting and Debugging VoIP Call Basics*
- *VoIP Debug Commands*

**Note**

Commands are listed in alphabetical order.

- Make sure that VoIP is working.
- Make sure that you can make a voice call.
- Verify that SIP-supported codecs are used. Support for codecs varies on different platforms; use the **codec ?** command to determine the codecs available on a specific platform.
- Use the **debug aaa authentication** command to display high-level diagnostics related to AAA logins.
- Use the **debug asnl events** command to verify that the SIP subscription server is up. The output displays a pending message if, for example, the client is unsuccessful in communicating with the server.
- Use the **debug call fallback** family of commands to display details of VoIP call fallback.
- Use the **debug cch323** family of commands to provide debugging output for various components within an H.323 subsystem.
- Use the **debug ccsip** family of commands for general SIP debugging, including viewing direction-attribute settings and port and network address-translation traces. Use any of the following related commands:
 - **debug ccsip all**—Enables all SIP-related debugging

- **debug ccsip calls**—Enables tracing of all SIP service-provider interface (SPI) calls
- **debug ccsip error**—Enables tracing of SIP SPI errors
- **debug ccsip events**—Enables tracing of all SIP SPI events
- **debug ccsip info**—Enables tracing of general SIP SPI information, including verification that call redirection is disabled
- **debug ccsip media**—Enables tracing of SIP media streams
- **debug ccsip messages**—Enables all SIP SPI message tracing, such as those that are exchanged between the SIP user-agent client (UAC) and the access server
- **debug ccsip preauth**—Enables diagnostic reporting of authentication, authorization, and accounting (AAA) preauthentication for SIP calls
- **debug ccsip states**—Enables tracing of all SIP SPI state tracing
- **debug ccsip transport**—Enables tracing of the SIP transport handler and the TCP or User Datagram Protocol (UDP) process
- Use the **debug isdn q931** command to display information about call setup and teardown of ISDN network connections (layer 3) between the local router (user side) and the network.
- Use the **debug kpml** command to enable debug tracing of KeyPad Markup Language (KPML) parser and builder errors.
- Use the **debug radius** command to enable debug tracing of RADIUS attributes.
- Use the **debug rpms-proc preauth** command to enable debug tracing on the RPMS process for H.323 calls, SIP calls, or both H.323 and SIP calls.
- Use the **debug rtr trace** command to trace the execution of an SAA operation.
- Use the **debug voip** family of commands, including the following:
 - **debug voip ccapi protoheaders**—Displays messages sent between the originating and terminating gateways. If no headers are being received by the terminating gateway, verify that the **header-passing** command is enabled on the originating gateway.
 - **debug voip ivr script**—Displays any errors that might occur when the Tel script is run
 - **debug voip rtp session named-event 101**—Displays information important to DTMF-relay debugging, if you are using codec types g726r16 or g726r24. Be sure to append the argument *101* to the command to prevent the console screen from flooding with messages and all calls from failing.

Sample output for some of these commands follows:

- [Sample Output for the debug ccsip events Command, page 19](#)
- [Sample Output for the debug ccsip info Command, page 20](#)

Sample Output for the debug ccsip events Command

- The example shows how the Proxy-Authorization header is broken down into a decoded username and password.

```
Router# debug ccsip events

CCSIP SPI: SIP Call Events tracing is enabled

21:03:21: sippmh_parse_proxy_auth: Challenge is 'Basic'.
21:03:21: sippmh_parse_proxy_auth: Base64 user-pass string is
'MTIzNDU2Nzg5MDEyMzQ1Njou'.
21:03:21: sip_process_proxy_auth: Decoded user-pass string is '1234567890123456:.'
```

```

21:03:21: sip_process_proxy_auth: Username is '1234567890123456'.
21:03:21: sip_process_proxy_auth: Pass is '.'.
21:03:21: sipSPIAddBillingInfoToCcb: sipCallId for billing records =
10872472-173611CC-81E9C73D-F836C2B6@172.18.192.19421:03:21: ****Adding to UAS Request
table

```

Sample Output for the debug ccsip info Command

This example shows only the portion of the debug output that shows that call redirection is disabled. When call redirection is enabled (default), there are no debug line changes.

```
Router# debug ccsip info
```

```

00:20:32: HandleUdpSocketReads :Msg enqueued for SPI with IPAddr: 172.18.207.10
:5060
00:20:32: CCSIP-SPI-CONTROL: act_sentininvite_new_message
00:20:32: CCSIP-SPI-CONTROL: sipSPICheckResponse
00:20:32: sip_stats_status_code
00:20:32: ccsip_get_code_class: !!Call Redirection feature is disabled on the GW
00:20:32: ccsip_map_call_redirect_responses: !!Mapping 302 response to 480
00:20:32: Roundtrip delay 4 milliseconds for method INVITE

```

Configuration Examples for Basic SIP Configuration

This section provides the following configuration examples:

- [SIP Register Support: Example, page 20](#)
- [SIP Redirect Processing Enhancement: Examples, page 22](#)
- [SIP 300 Multiple Choice Messages: Example, page 26](#)

SIP Register Support: Example

```

Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
  redirect ip2ip
sip
  redirect contact order best-match

ip dhcp pool vespa
  network 192.168.0.0 255.255.255.0
  option 150 ip 192.168.0.1
  default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1

```

```
    codec preference 2 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
  ip address 10.8.17.22 255.255.0.0
  half-duplex
!
interface FastEthernet0/0
  ip address 192.168.0.1 255.255.255.0
  speed auto
  no cdp enable
  h323-gateway voip interface
  h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
  network 10.0.0.0
  network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
  destination-pattern 5100
  port 1/0
!
dial-peer voice 2 pots
  destination-pattern 9998
  port 1/1
!
dial-peer voice 123 voip
  destination-pattern [12]...
  session protocol sipv2
  session target ipv4:10.8.17.42
  dtmf-relay sip-notify
!
gateway
!
sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar dns:myhost3.example.com expires 3600
  registrar ipv4:10.8.17.40 expires 3600 secondary
```

```

!
telephony-service
  max-dn 10
  max-conferences 4
!
ephone-dn 1
  number 4001
!
ephone-dn 2
  number 4002
!
line con 0
  exec-timeout 0 0
line aux 0
line vty 0 4
  login
line vty 5 15
  login
!
no scheduler allocate
end

```

SIP Redirect Processing Enhancement: Examples

This section provides configuration examples to match the identified configuration tasks in the previous sections.

- [Call Redirection Disabled, page 22](#)
- [Call Redirection Enabled, page 23](#)
- [Call Redirection Using IP-to-IP Redirection, page 24](#)
- [SIP 300 Multiple Choice Messages: Example, page 26](#)



Note

IP addresses and hostnames in examples are fictitious.

Call Redirection Disabled

This example shows that call redirection is disabled on the gateway.

```

Router# show running-config

Building configuration...

Current configuration : 2791 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut

```



```

speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8183821111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7173721111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
dial-peer voice 4 voip
application session
destination-pattern 6163621111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
no redirection
    retry invite 1
    retry bye 1
!
line con 0
line aux 0
line vty 0 4
login
!
end

```

Call Redirection Enabled

This example shows that call redirection is enabled on the gateway (the default). When call redirection is enabled, the output shows no redirection.

```

Router# show running-config

Building configuration...

Current configuration : 2791 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut

```

```

speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8183821111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7173721111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
dial-peer voice 4 voip
application session
destination-pattern 6163621111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
    retry invite 1
    retry bye 1
!
line con 0
line aux 0
line vty 0 4
login
!
end

```

Call Redirection Using IP-to-IP Redirection

This example shows that redirection was set globally on the router.

```

Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
    redirect ip2ip
sip
    redirect contact order best-match

ip dhcp pool vespa
network 192.168.0.0 255.255.255.0
option 150 ip 192.168.0.1
default-router 192.168.0.1

```

```
!
voice call carrier capacity active
!
voice class codec 1
  codec preference 2 g711ulaw
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
  ip address 10.8.17.22 255.255.0.0
  half-duplex
!
interface FastEthernet0/0
  ip address 192.168.0.1 255.255.255.0
  speed auto
  no cdp enable
  h323-gateway voip interface
  h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
  network 10.0.0.0
  network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
  destination-pattern 5100
  port 1/0
!
dial-peer voice 2 pots
  destination-pattern 9998
  port 1/1
!
dial-peer voice 123 voip
  destination-pattern [12]...
  session protocol sipv2
  session target ipv4:10.8.17.42
  dtmf-relay sip-notify
!
gateway
!
```

```

sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar dns:myhost3.example.com expires 3600
  registrar ipv4:10.8.17.40 expires 3600 secondary
!
!
telephony-service
  max-dn 10
  max-conferences 4
!
ephone-dn 1
  number 4001
!
ephone-dn 2
  number 4002
!
line con 0
  exec-timeout 0 0
line aux 0
line vty 0 4
  login
line vty 5 15
  login
!
no scheduler allocate
end

```

SIP 300 Multiple Choice Messages: Example

This section provides a configuration example showing redirect contact order set to best match.

```

Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
  redirect ip2ip
sip
  redirect contact order best-match

ip dhcp pool vespa
  network 192.168.0.0 255.255.255.0
  option 150 ip 192.168.0.1
  default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
  codec preference 2 g711ulaw
!
no voice hpi capture buffer

```

```
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
 ip address 10.8.17.22 255.255.0.0
 half-duplex
!
interface FastEthernet0/0
 ip address 192.168.0.1 255.255.255.0
 speed auto
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
 network 10.0.0.0
 network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
dial-peer voice 2 pots
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.example.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
telephony-service
 max-dn 10
```

```

max-conferences 4
!
ephone-dn 1
number 4001
!
ephone-dn 2
number 4002
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
login
line vty 5 15
login
!
no scheduler allocate
end

```

Toll Fraud Prevention

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

- Disable secondary dial tone on voice ports—By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.
- Cisco router access control lists (ACLs)—Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized SIP or H.323 calls from unknown parties to be processed and connected by the router or gateway.
- Close unused SIP and H.323 ports—If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplexing (TDM) trunks or IP, close the unused H.323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.
- Change SIP port 5060—If SIP is actively used, consider changing the port to something other than well-known port 5060.
- SIP registration—If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.
- SIP Digest Authentication—If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.

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

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

Additional References

- “[SIP Features Roadmap](#)”—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.

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Achieving SIP RFC Compliance

This chapter describes how to use or configure your Cisco SIP gateway so as to comply with published SIP standards. It discusses the following features:

- SIP: Core SIP Technology Enhancements (RFC 2543 and RFC 2543-bis-04)
- SIP - DNS SRV RFC 2782 Compliance (RFC 2782)
- SIP: RFC 3261 Enhancements (RFC 3261)
- SIP Gateway Compliance to RFC 3261, RFC 3262, and RFC 3264
- SIP Stack Portability



Note This feature is described in the [“Configuring SIP Message, Timer, and Response Features” on page 1.](#)

Feature History for SIP: Core SIP Technology Enhancements

Release	Modification
12.2(13)T	This feature was introduced to achieve compliance with SIP RFC 2543-bis-04, later published as RFC_3261.

Feature History for SIP - DNS SRV RFC 2782 Compliance

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(8)T	This feature was integrated into the release.

Feature History for SIP: RFC 3261 Enhancements

Release	Modification
12.3(4)T	This feature was introduced.



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Feature History for SIP Gateway Compliance to RFC 3261, RFC 3262, and RFC 3264

Release	Modification
12.3(8)T	This feature was introduced.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

**Note**

See “RFCs” section on page 28 for more detailed information about obsoleted, updated, and new RFCs.

Contents

- [Prerequisites for SIP RFC Compliance, page 2](#)
- [Restrictions for SIP RFC Compliance, page 2](#)
- [Information About SIP RFC Compliance, page 3](#)
- [How to Configure SIP RFC Compliance, page 35](#)
- [Configuration Examples for SIP RFC Compliance, page 46](#)
- [Additional References, page 49](#)

Prerequisites for SIP RFC Compliance

- Configure a basic VoIP network.
- Enable the Reliable Provisional Response feature.

**Note**

For information on reliable provisional responses, see *SIP Gateway Support of RSVP and TEL URL*.

Restrictions for SIP RFC Compliance

- As found in RFC 3261, the following are not supported:
 - Sending SIP UPDATE requests; the gateway is able to receive and process only UPDATE requests.
 - SIP with IPv6 host addresses.
 - Secure SIPs. Secure SIPs are secure Uniform Resource Identifiers (URIs). When a caller makes a call using SIPs, the message transport is secure to the called party.
 - Field characters 0x0 to 0x7E in quoted strings within SIP headers encoded in Unicode Transformation Format Version 8 (UTF-8).

- As found in RFC 3264, support for bandwidth (b=) SDP attribute equal to 0 is not supported.

Information About SIP RFC Compliance

This section contains the following information:

- [SIP RFC 2543 Compliance, page 3](#)
- [SIP RFC 2782 Compliance, page 3](#)
- [SIP RFC 3261 Compliance, page 3](#)
- [SIP RFC 3261, RFC 3262, and RFC 3264 Compliance, page 32](#)

SIP RFC 2543 Compliance

The Cisco SIP gateway complies with RFC 2543. However, RFC 3261 has now replaced (obsoleted) RFC 2543. See [“Restrictions for SIP RFC Compliance” section on page 2](#) and [“SIP RFC 3261 Compliance” section on page 3](#) for more information about what is and is not supported in the new RFCs.

SIP RFC 2782 Compliance

SIP on Cisco VoIP gateways uses Domain Name System Server (DNS SRV) query to determine the IP address of the user endpoint. The query string has a prefix in the form of “protocol.transport.” as defined by RFC 2052. This prefix is attached to the fully qualified domain name (FQDN) of the next-hop SIP server.

A second prefix style has been added to Cisco VoIP gateways and is now the default. This second style is defined by RFC 2782, which obsoleted RFC 2052 in February 2000. This new style is in compliance with RFC 2782 and appends the protocol label with an underscore “_” as in “_protocol._transport.” The addition of the underscore reduces the risk of the same name being used for unrelated purposes.

SIP RFC 3261 Compliance

RFC 3261, which obsoletes RFC 2543, defines the SIP signaling protocol for creating, modifying, and terminating sessions. Cisco’s implementation of RFC 3261 supports the following:

- Ability to receive and process SIP UPDATE requests
- Initial Offer and Answer exchanges
- Branch and Sent-by parameters in the Via header
- Merged request detection
- Loose-routing

Benefits of RFC 3261 include the following:

- Continued interoperability of Cisco IOS gateways in current SIP deployments
- Expanded interoperability of Cisco IOS gateways with new SIP products and applications

This section contains the following information about basic SIP functionality:

- [SIP Header Fields, Network Components, and Methods, page 4](#)

- [SIP Responses](#), page 7
- [SIP SDP Usage, Transport Layer Protocols, and DNS Records](#), page 11
- [SIP Extensions](#), page 12
- [SIP Security](#), page 13
- [SIP DTMF Relay](#), page 13
- [SIP Fax Relay and T.38](#), page 14
- [SIP URL Comparison](#), page 16
- [487 Sent for BYE Requests](#), page 16
- [3xx Redirection Responses](#), page 17
- [DNS SRV Query Procedure](#), page 17
- [CANCEL Request Route Header](#), page 17
- [Interpret User Parameters](#), page 17
- [user=phone Parameter](#), page 17
- [303 and 411 SIP Cause Codes](#), page 17
- [Flexibility of Content-Type Header](#), page 18
- [Optional SDP “s=” Line](#), page 18
- [Allow Header Addition to INVITES and 2xx Responses](#), page 18
- [Simultaneous Cancel and 2xx Class Response](#), page 18
- [UPDATE-Request Processing](#), page 18

SIP Header Fields, Network Components, and Methods

Table 2 through Table 4 show RFC 3261 SIP functions—including headers, components, and methods. They also show if the specific functionality is supported by Cisco SIP gateways.

Table 2 **SIP Header Fields**

Header Field	Supported by Cisco Gateways?
Accept	Yes. Used in OPTIONS response messages.
Accept-Encoding	No
Accept-Language	Yes
Alert-Info	No
Allow	Yes
Also	
Authentication-Info	No
Authorization	
Call-ID	Yes
Call-Info	No

Table 2 **SIP Header Fields (continued)**

Header Field	Supported by Cisco Gateways?	
CC-Diversion / Diversion	Yes	
Contact		
Content-Disposition		
Content-Encoding	No	
Content-Encoding	Yes	
Content-Language	No	
Content-Length	Yes	
Content-Type		
Cseq		
Date		
Encryption	No	
Error-Info		
Event	Yes	
Expires		
From		
In-Reply-To	No	
Max-Forwards	Yes	
MIME-Version		
Min-Expires		
Min-SE		
Organization	No	
Priority		
Proxy-Authenticate		
Proxy-Authenticate	Yes	
Proxy-Authorization		
Proxy-Require	No	
Rack	Yes	
Reason		
Record-Route		
Referred-By		
Referred-To		
Replaces		
Requested-By		
Require		
Response-Key		No
Retry-After		

Table 2 SIP Header Fields (continued)

Header Field	Supported by Cisco Gateways?
Retry-After	Yes
Route	
RSeq	
Server	
Session-Expires	
Subject	
Supported	Yes
Timestamp	
To	
Unsupported	
User-Agent	
Via	
Warning	
WWW-Authenticate	
WWW-Authenticate	Yes

Table 3 SIP Network Components

SIP Network Components	Supported by Cisco Gateways?
User Agent Client (UAC)	Yes
User Agent Server (UAS)	
Proxy Server	No
Redirect Server	Yes
Registrar Server	

Table 4 SIP Methods

Method	Supported by Cisco Gateways?
ACK	Yes
BYE	
CANCEL	
COMET	Deprecated. Conditions MET. Used in Quality of Service (QoS) implementations to indicate to the opposite endpoint whether or not the conditions have been met—that is, if the proper resources have been reserved.

Table 4 **SIP Methods (continued)**

Method	Supported by Cisco Gateways?
INVITE	Yes. SIP gateways support midcall Invite requests with the same call-ID but different Session Description Protocols (SDP) session parameters (to change the transport address). Midcall INVITE requests can also change the port number, codec, or refresh the session timer value.
INFO	Yes. SIP gateways can accept and generate INFO messages.
NOTIFY	Yes. Used in implementation of the Refer requests. Notify messages let the initiator of the Refer request know the outcome of the transfer. Notify messages also let a subscriber know of any changes occurring in selected events, such as dual tone multifrequency events (DTMF) or message waiting indication (MWI) events.
OPTIONS	Yes. SIP gateways receive this method only.
PRACK	Yes. Enable or Disable Provisional Reliable Acknowledgements (PRACK).
REFER	Yes. The SIP gateway responds to a Refer request and also generates a Refer request for attended and blind call transfers.
REGISTER	Yes. The SIP gateway can send and receive SIP REGISTER requests.
SUBSCRIBE	Yes. The SIP gateway can generate and accept SUBSCRIBE requests. The gateway processes SUBSCRIBE requests for selected applications such as DTMF telephony events and for MWI.
UPDATE	Yes. The SIP gateway can accept UPDATEs for media changes, target refreshes, and QoS scenarios. The gateway will send UPDATEs only for QoS scenarios.

SIP Responses

Table 5 through Table 10 show SIP responses that are supported by Cisco SIP gateways in compliance with RFC 3261.

Cisco SIP gateways do not initiate the use of keepalive messages for calls that they originate or terminate. If the remote gateway uses a keepalive message, the SIP gateway complies.

Table 5 **1xx Responses**

1xx Responses	Comments
100 Trying	Action is being taken on behalf of the caller, but that the called party has not yet been located. The SIP gateway generates this response for an incoming Invite request. Upon receiving this response, a gateway stops retransmitting Invite requests and waits for a 180 Ringing or 200 OK response.
180 Ringing	The called party has been located and is being notified of the call. The SIP gateway generates a 180 Ringing response when the called party has been located and is being alerted. Upon receiving this response, the gateway generates local ringback, then it waits for a 200 OK response.

Table 5 **1xx Responses (continued)**

1xx Responses	Comments
181 Call is being forwarded	The call is being rerouted to another destination. The SIP gateway does not generate these responses. Upon receiving these responses, the gateway processes the responses in the same way it processes a 100 Trying response.
182 Queued	The called party is not currently available, but has elected to queue the call rather than reject it. The SIP gateway does not generate these responses. Upon receiving these responses, the gateway processes the responses in the same way it processes a 100 Trying response.
183 Session progress	Performs inband alerting for the caller. The SIP gateway generates a 183 Session progress response when it receives an ISDN Progress message with an appropriate media indication from the PSTN.

Table 6 **2xx Responses**

2xx Responses	Comments
202 Accepted	The SIP gateway will send this response for incoming REFER and SUBSCRIBE requests. It will accept this response for outgoing REFER and SUBSCRIBE requests.
200 OK	The request has been successfully processed. The action taken depends on the request.

Table 7 **3xx Responses**

3xx Responses	Comments
	The SIP gateway does not generate this response. Upon receiving this response, the gateway contacts the new address in the Contact header field.
300 Multiple Choice	The address resolved to more than one location. All locations are provided and the user or user agent (UA) is allowed to select which location to use.
301 Moved permanently	The user is no longer available at the specified location. An alternate location is included in the header.
302 Moved temporarily	The user is temporarily unavailable at the specified location. An alternate location is included in the header.
305 Use proxy	The caller must use a proxy to contact the called party.
380 Alternative service	The call was unsuccessful, but that alternative services are available.

Table 8 **4xx Responses**

4xx Responses	Comments
Upon receiving a 4xx response, the SIP gateway initiates a graceful call disconnect and clears the call.	
423 Interval Too Brief	The SIP gateway generates this response.
400 Bad Request	The request could not be understood because of an illegal format. The SIP gateway generates this response for a badly formed request.
401 Unauthorized	The request requires user authentication. The SIP gateway does not generate this response.
402 Payment required	Payment is required to complete the call. The SIP gateway does not generate this response.
403 Forbidden	The server has received and understood the request but will not provide the service. The SIP gateway does not generate this response.
404 Not Found	The server has definite information that the user does not exist in the specified domain. The SIP gateway generates this response if it is unable to locate the called party.
405 Method Not Allowed	The method specified in the request is not allowed. The response contains a list of allowed methods. The SIP gateway generates this response if an invalid method is specified in the request.
406 Not Acceptable	The requested resource is capable of generating only responses that have content characteristics not acceptable as specified in the accept header of the request. The SIP gateway does not generate this response.
407 Proxy authentication required	Similar to a 401 Unauthorized response. However, the client must first authenticate itself with the proxy. The SIP gateway does not generate this response.
408 Request timeout	The server could not produce a response before the Expires time out. The SIP gateway does not generate this response.
410 Gone	A resource is no longer available at the server and no forwarding address is known. The SIP gateway generates this response if the PSTN returns a cause code of unallocated number.
413 Request entity too large	The server refuses to process the request because it is larger than the server is willing or able to process. The SIP gateway does not generate this response.
414 Request-URI too long	The server refuses to process the request because the Request-URI is too long for the server to interpret. The SIP gateway does not generate this response.
415 Unsupported media	The server refuses to process the request because the format of the body is not supported by the destination endpoint. The SIP gateway generates this response when it gets an Info message for an unsupported event-type. Supported event types are 0-9, A-D, # and *.
416 Unsupported Request URI scheme	The SIP gateway generates this response when it gets an unsupported URI scheme such as http: or sips: in a SIP request.
420 Bad extension	The server could not understand the protocol extension indicated in the Require header. The SIP gateway generates this response if it cannot understand the service requested.

Table 8 **4xx Responses (continued)**

4xx Responses	Comments
421 Extension Required	The SIP gateway does not generate this response.
422 Session Timer Too Small	Generated by the UAS when a request contains a Session-Expires header with a duration that is below the minimum timer for the gateway server. The 422 response must contain a Min-SE header with a minimum timer for that server.
480 Temporarily unavailable	The called party was contacted but is temporarily unavailable. The SIP gateway generates this response if the called party is unavailable. For example, the called party does not answer the phone within a certain amount of time, or the called number does not exist or is no longer in service.
481 Call leg/transaction does not exist	The server is ignoring the request because the request was either a Bye request for which there was no matching leg ID, or a Cancel request for which there was no matching transaction. The SIP gateway generates this response if the call leg ID or transaction cannot be identified.
482 Loop detected	The server received a request that included itself in the path. A SIP gateway generates this response when it detects the same request has arrived more than once in different paths (most likely due to forking).
483 Too many hops	The server received a request that required more hops than allowed by the Max-Forwards header. The SIP gateway does not generate this response.
484 Address incomplete	The server received a request containing an incomplete address. The SIP gateway does not generate this response.
485 Ambiguous	The server received a request in which the called party address was ambiguous. It can provide possible alternate addresses. The SIP gateway does not generate this response.
486 Busy here	The called party was contacted but that their system is unable to take additional calls. The SIP gateway generates this response if the called party was contacted but was busy.
487 Request cancelled	The request was terminated by a Bye or Cancel request. The SIP gateway generates this response to an unexpected Bye or Cancel received for a request.
488 Not Acceptable Media	Indicates an error in handling the request at this time. The SIP gateway generates this response if media negotiation fails.
491 Request Pending	The SIP gateway generates this response to reject an UPDATE message proposing a new offer, if it receives the new offer before it receives an answer to an offer it has previously requested.
493 Undecipherable	The SIP gateway does not generate this response.

Table 9 **5xx Responses**

5xx Responses	Comments
	The SIP gateway generates this response if it encountered an unexpected error that prevented it from processing the request. Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.
500 Server internal error	The server or gateway encountered an unexpected error that prevented it from processing the request.
501 Not implemented	The server or gateway does not support the functions required to complete the request.
502 Bad gateway	The server or gateway received an invalid response from a downstream server.
503 Service unavailable	The server or gateway is unable to process the request due to an overload or maintenance problem.
504 Gateway timeout	The server or gateway did not receive a timely response from another server (such as a location server).
505 Version not supported	The server or gateway does not support the version of the SIP protocol used in the request.
513 Message too large	The SIP gateway does not generate this response.
580 Precondition failed	A failure in having QoS preconditions met for a call.

Table 10 **6xx Responses**

6xx Responses	Comments
	The SIP gateway does not generate this response. Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.
600 Busy everywhere	The called party was contacted but that the called party is busy and cannot take the call at this time.
603 Decline	The called party was contacted but cannot or does not want to participate in the call.
604 Does not exist anywhere	The server has authoritative information that the called party does not exist in the network.
606 Not acceptable	The called party was contacted, but that some aspect of the session description was unacceptable.

SIP SDP Usage, Transport Layer Protocols, and DNS Records

Table 11 through Table 13 show SIP SDP usage, transport protocols, and DNS records that are supported in RFC 3261. They also show if the specific functionality is supported by Cisco SIP gateways.

Table 11 SIP Session Description Protocol (SDP) Usage Supported in RFC 3261

SIP Network Components	Supported by Cisco Gateways?
a (Media attribute line)	Yes. The primary means for extending SDP and tailoring it to a particular application or media.
c (Connection information)	Yes.
m (Media name and transport address)	
o (Owner/creator and session identifier)	
s (Session name)	
t (Time description)	
v (Protocol version)	

Table 12 SIP Transport Layer Protocols

Protocol	Supported by Cisco Gateways?
Multicast UDP	No
TCP	Yes
TLS	No
Unicast UDP	Yes

Table 13 SIP Domain Name System (DNS) Records

Authentication Encryption Mode	Supported by Cisco Gateways?
RFC 3263 Type A	Yes
RFC 3263 Type NAPTR	No
RFC 3263 Type SRV	Yes

SIP Extensions

Table 14 shows supported SIP extensions.

Table 14 SIP Extensions

SIP Extension	Comments
RFC 3262: Reliability of Provisional Responses in SIP	Supported.
RFC 3263: Locating SIP Servers	The gateway does not support DNS NAPTR lookups. It supports DNS SRV and A record lookups and has the provision to cycle through the multiple entries.
RFC 3265: SIP Specific Event Notification	The gateway supports the SUBSCRIBE-NOTIFY framework.

Table 14 *SIP Extensions (continued)*

SIP Extension	Comments
RFC 3311: SIP UPDATE Method	The gateway accepts UPDATE for media changes, target refreshes, and QoS Scenarios. It sends UPDATE for only QoS scenarios.
RFC 3312: Integration of Resource Management and SIP - RFC	Midcall QoS changes do not use the 183-PRACK model defined in this RFC.
RFC 3326: Reason Header field for SIP	The gateway uses this to relay the Q.850 cause code to the remote SIP device.
RFC 3515: SIP REFER Method	The gateway does not send or accept out-of-dialog REFER requests. Overlapping REFERs are not supported. REFER is supported only in the context of call transfer scenarios (that is, triggered INVITE cases only). The gateway supports relevant portions of RFC 3892 (Referred-By) and RFC 3891 (Replaces header) as needed for call-transfer scenarios.

SIP Security

Table 15 and Table 16 show SIP security encryption and responses supported in RFC 3261. They also show if the specific functionality is supported by Cisco SIP gateways.

Table 15 *SIP Encryption Modes*

Encryption Mode	Supported by Cisco Gateways?
End-to-end Encryption	No. IPSEC can be used for security.
Hop-by-Hop Encryption	
Privacy of SIP Responses	No.
Via Field Encryption	No. IPSEC can be used for security.

Table 16 *SIP Authentication Encryption Modes*

Authentication Encryption Mode	Supported by Cisco Gateways?
Digest Authentication	Yes
PGP	No
Proxy Authentication	No
Secure SIP or sips	URI scheme is not supported

SIP DTMF Relay

Cisco SIP gateways support DTMF relay in accordance with RFC 2833. The DTMF relay method is based on the transmission of Named Telephony Events (NTE) and DTMF digits over a Real-Time Transport Protocol (RTP) stream.

Cisco SIP gateways also support forwarding DTMF tones by means of cisco-rtp, which is a Cisco proprietary payload type.

Table 17 shows SIP DTMF relay methods. It also shows if the specific method is supported by Cisco SIP gateways.

Table 17 SIP DTMF Relay Supported in RFC 3261

Method	Supported by Cisco Gateways?
RFC 2833	Yes. The default RTP payload type for rtp-nte is 101. The default method of DTMF relay is inband voice.
Cisco RTP (Cisco proprietary)	Yes, except on Cisco AS5350 and Cisco AS5400.

SIP Fax Relay and T.38

Table 18 shows fax relay modes that are supported by Cisco SIP gateways in compliance with RFC 3261. It also shows if the specific method is supported by Cisco SIP gateways.

Table 18 Fax Relay Modes Supported in RFC 3261

Method	Supported by Cisco Gateways?
T.38 Fax Relay	Yes
Cisco Fax Relay	Yes, except on Cisco AS5350 and Cisco AS5400

Cisco SIP gateways support T.38 and T.37 fax relay, store, and forward mechanisms. Table 19 is based on Annex-D of the T.38 ITU recommendation, *Procedures for Real-Time Group 3 Facsimile Communication over IP Networks*, June 1998. The table indicates recommendations from the standard and if Cisco SIP gateways support the requirements.

Table 19 T.38 Fax Requirements

Requirement	Description	Mandatory or Optional	Supported?
SIPt38-01	T.38 over SIP must be implemented as described in ANNEX D of the T.38 ITU recommendation, <i>Procedures for Real-Time Group 3 Facsimile Communication over IP Networks</i> , June 1998.	Mandatory	Yes
SIPt38-02	SIP-enabled VoIP gateways detect calling tones (CNG), called station identifier (CED) fax tones, and/or the preamble flag sequence transmitted inside the audio RTP streams.	Mandatory	Yes — only the CED V.21 preamble and not the CNG tone is used to detect fax.
SIPt38-03	Fax transmission detection is performed by the receiving gateway by recognizing the CED tone.	Mandatory	Yes
SIPt38-04	If the CED tone is not present, the fax transmission is detected by the receiving gateway by recognizing the Preamble flag sequence.	Mandatory	Yes
SIPt38-05	Upon detection of the fax transmission, the receiving gateway initiates the switch over to T.38 fax mode by sending a reINVITE request with SDP.	Mandatory	Yes

Table 19 *T.38 Fax Requirements (continued)*

Requirement	Description	Mandatory or Optional	Supported?
SIPt38-06	To prevent glare, even if the transmitting gateway detects the fax transmission (CNG tone), the gateway does not initiate the switch over to T.38 fax mode.	Mandatory	Yes
SIPt38-07	If a SIP session starts with audio capabilities and then switches to fax, the session switches back to audio mode at the end of the fax transmission.	Mandatory	Yes
SIPt38-08	Support of SIP T.38 fax calls over TCP.	Desirable	UDP only
SIPt38-09	Facsimile UDP transport Layer (UDPTL) is supported.	Mandatory	Yes
SIPt38-10	The following SDP attributes support T.38 fax sessions: <ul style="list-style-type: none"> Registered SDP Protocol format, MIME media type image/t38: MIME media type name: image MIME subtype name: t38 	Mandatory	Yes
SIPt38-11	The following attributes support T.38 sessions. <ul style="list-style-type: none"> T38FaxVersion T38maxBitRate T38FaxFillBitRemoval T38FaxTranscodingMMR T38FaxTranscodingJBIG T38FaxRateManagement T38FaxMaxBuffer T38FaxMaxDatagram T38FaxUdpEC 	Mandatory	Yes
SIPt38-12	Cisco SIP-enabled gateways supporting T.38 interoperate with gateways from Cisco and other vendors.	Mandatory	Yes
SIPt38-13	Interoperability with gateways that support T.38 over H.323.	Optional	No
SIPt38-14	Configuration of SIP enabled gateways include management of SIP T.38 specific configurable choices.	Mandatory	Yes. The following are configurable: <ul style="list-style-type: none"> bitrate TCP/UDP (UDP only) hs and ls redundancy ECM

Table 19 T.38 Fax Requirements (continued)

Requirement	Description	Mandatory or Optional	Supported?
SIPt38-15	Tracking and reporting of SIP T.38 activity on the gateways is desired. This includes generation of Call Detail Records (CDR) for SIP T.38 fax calls.	Mandatory	Yes
SIPt38-16	RFC 3261 security mechanisms apply. Message authentication can be performed on SIP Invite request and Bye requests.	Optional	No

SIP URL Comparison

When Uniform Resource Locators (URLs) are received, they are compared for equality. URL comparison can be done between two From SIP URLs or between two To SIP URLs. The order of the parameters does not need to match precisely. However, for two URLs to be equal, the user, password, host, and port parameters must match.

With Cisco IOS Release 12.3, the *maddr* and *transport* parameters are no longer allowed in Cisco SIP gateway implementations. The *user-param* parameter is now an acceptable parameter for comparison.

If a compared parameter is omitted or not present, it is matched on the basis of its default value. [Table 20](#) shows a list of SIP URL compared parameters and their default values.

Table 20 SIP URL Compared Parameters and Default Values

SIP URL Compared Parameter	Default
User	—
Password	—
Host	Mandatory
Port	5060
User-param	IP

Assuming that a comparison is taking place, the following is an example of equivalent URLs:

Original URL:

```
sip:36602@172.18.193.120
```

Equivalent URLs:

```
sip:36602@172.18.193.120:
sip:36602@172.18.193.120;tag=499270-A62;pname=pvalue
sip:36602@172.18.193.120;user=ip
sip:36602@172.18.193.120:5060
```

487 Sent for BYE Requests

RFC 3261 requires that a UAS that receives a BYE request first send a response to any pending requests for that call before disconnecting. After receiving a BYE request, the UAS should respond with a 487 (Request Cancelled) status message.

3xx Redirection Responses

See the “Configuring SIP Redirect Processing Enhancement” section on page 8.

DNS SRV Query Procedure

In accordance with RFC 3261, when a Request URI or the session target in the dial peer contains a fully qualified domain name (FQDN), the UAC needs to determine the protocol, port, and IP address of the endpoint before it forwards the request. SIP on Cisco gateways uses Domain Name System Server (DNS SRV) query to determine the protocol, port, and IP address of the user endpoint.

Before Cisco IOS Release 12.2(13)T, the DNS query procedure did not take into account the destination port.

CANCEL Request Route Header

A CANCEL message sent by a UAC on an initial INVITE request cannot have a Route header. Route headers cannot appear in a CANCEL message because they take the same path as INVITE requests, and INVITE requests cannot contain Route headers.

Interpret User Parameters

There are instances when the telephone-subscriber or user parameters can contain escaped characters to incorporate space, control characters, quotation marks, hash marks, and other characters. After the receipt of an INVITE message, the telephone-subscriber or user parameter is interpreted before dial-peer matching is done. For example, the escaped telephone number in an incoming INVITE message may appear as:

```
-%32%32%32
```

Although 222 is a valid telephone number, it requires interpretation. If the interpretation is not done, the call attempt fails when the user parameter is matched with the dial-peer destination pattern.

user=phone Parameter

A SIP URL identifies a user’s address, which appears similar to an e-mail address. The form of the user’s address is *user@host* where *user* is the user identification and *host* is either a domain name or a numeric network address. For example, the request line of an outgoing INVITE request might appear as:

```
INVITE sip:5550100@example.com
```

The *user=phone* parameter formerly required in a SIP URL is no longer necessary. However, if an incoming SIP message has a SIP URL with *user=phone*, *user=phone* is parsed and used in the subsequent messages of the transaction.

303 and 411 SIP Cause Codes

RFC 3261 obsoletes the SIP cause codes 303 *Redirection: See Other* and 411 *Client Error: Length required*.

Flexibility of Content-Type Header

The Content-Type header, which specifies the media type of the message body, is permitted to have an empty Session Description Protocol (SDP) body.

Optional SDP “s=” Line

The “s=” line in SDP is accepted as optional. The “s=” line describes the reason or subject for SDP information. Cisco SIP gateways can create messages with an “s=” line in SDP bodies and can accept messages that have no “s=” line.

Allow Header Addition to INVITEs and 2xx Responses

The use of the Allow header in an initial or re-INVITE request or in any 2xx class response to an INVITE is permitted. The Allow header lists the set of methods supported by the user agent that is generating the message. Because it advertises what methods should be invoked on the user agent sending the message, it avoids congesting the message traffic unnecessarily. The Allow header can contain any or all of the following: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, NOTIFY, INFO, SUBSCRIBE.

Simultaneous Cancel and 2xx Class Response

According to RFC 3261, if the UAC desires to end the call before a response is received to an INVITE, the UAC sends a CANCEL. However, if the CANCEL and a 2xx class response to the INVITE “pass on the wire,” the UAC also receives a 2xx to the INVITE. When the two messages pass, the UAC terminates the call by sending a BYE request.

UPDATE-Request Processing

RFC 3261, which obsoletes RFC 2543, defines the SIP signaling protocol for creating, modifying and terminating sessions. The SIP Extensions for Caller Identity and Privacy feature provides support for the following SIP gateway implementations that are compliant with the RFC 3261 specification:

- [SIP UPDATE Requests, page 18](#)
- [Via Header Parameters and Merged Request Detection, page 23](#)
- [Loose-Routing and the Record-Route Header, page 23](#)
- [Multiple INVITE Requests Before a Final Response, page 23](#)
- [Mid-call Re-INVITE Request Failure, page 24](#)
- [PRACK Request with a New Offer, page 25](#)
- [Reliable Provisional Response Failure, page 25](#)

SIP UPDATE Requests

SIP accomplishes session management through a series of messages that are either requests from a server or client, or responses to a request. SIP uses an INVITE request to initiate and modify sessions between user agents (UAs), and uses the ACK method to acknowledge a final response to an INVITE request. In some cases a session needs to be modified before the INVITE request is answered. This scenario occurs, for example, in a call that sends early media, the information sent to convey call progress during an established session, and for which the INVITE request has not been accepted. In this scenario either the

caller or callee should be able to modify the characteristics of a session, for instance, by putting the early media on hold before the call is answered. Prior to the SIP UPDATE method, which allows a client to update session parameters, there was no mechanism to allow a caller or callee to provide updated session information before a final response to the initial INVITE request was generated. The SIP Extensions for Caller Identity and Privacy feature provides support for the UPDATE method and enables the gateway capability to receive and process, but not send, UPDATE requests. The gateway also updates the session timer value after the call is active.

A user agent client (UAC) initiates a session by sending an INVITE request to a user agent server (UAS). The UAS responds to the invitation by sending the following response codes:

- A 1xx provisional response indicating call progress. All 1xx responses are informational and are not final; all non-1xx responses are final.
- A 2xx response indicating successful completion or receipt of a request
- A 3xx, 4xx, 5xx, or 6xx response indicating rejection or failure.

A PRACK response is used to acknowledge receipt of a reliably transported provisional response, including a response with early media indication, while the ACK is used to acknowledge a final response to an INVITE request. A PRACK establishes an early dialog between UAC and UAS, a requirement to receive UPDATE requests with a new offer.

When a 2xx response is sent it establishes a session and also creates a dialog, or call leg. A dialog established by a 1xx response is considered an early dialog, whereas a final response establishes a confirmed dialog. The SIP UPDATE method allows a UAC to update session parameters, such as the set of media streams and their codecs, without affecting the dialog state. Unlike a re-INVITE request, a SIP UPDATE request may be sent to modify a session before the initial INVITE request is answered without impacting the dialog state itself. The UPDATE method is useful for updating session parameters within early dialogs before the initial INVITE request has been answered, for example, when early media is sent.

The SIP UPDATE method makes use of the offer and answer exchange using Session Description Protocol (SDP), as defined in the IETF specification, RFC 3264, *An Offer/Answer Model with the Session Description Protocol (SDP)*. One UA in the session generates an SDP message that constitutes the offer, that is, the set of media streams and codecs the UA wants to use, along with IP addresses and ports where the UA wants to receive the media. The other UA generates an answer, an SDP message responding to the offer.

In the Cisco SIP implementation, a UAS can receive an UPDATE request in both early and confirmed dialogs. The point at which the offer is generated, the UPDATE is received, the presence or absence of reliable provisional response and SDP, are all factors that determine how the gateway handles the UPDATE request. An UPDATE request generates a response indicating one of several possible outcomes:

- Success
- Pending response to outstanding offers
- Failure

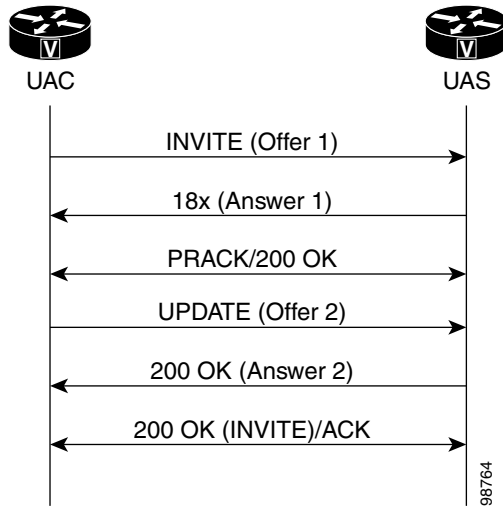
The following sections discuss how UPDATE requests are received and processed in various scenarios and call flows.

UPDATE Request Processing Before the Call Is Active

When the gateway sends a reliable provisional response with SDP, the response includes an Allow header that lists the UPDATE method and informs the caller of the gateway capability to support UPDATE processing.

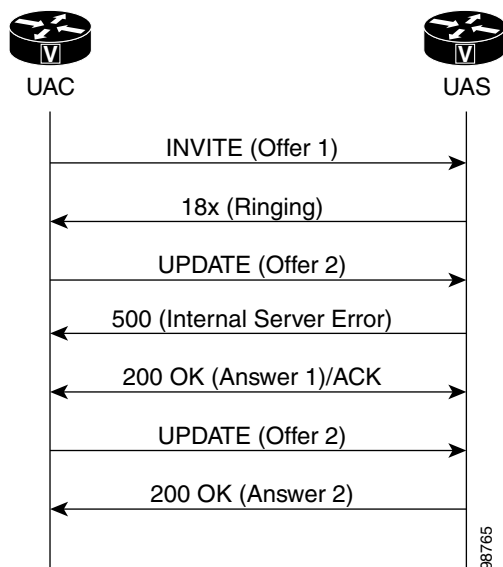
Figure 13 shows a call where the UAS sent a reliable provisional response (ANSWER 1) to an INVITE request (Offer 1). The 18x early media response indicated the gateway capability to support UPDATES. The UAC sent a provisional acknowledgement (PRACK) and received a 200 OK response to the PRACK request. The UAC requested the UAS modify the existing session media parameters of the early dialog by sending an UPDATE request (Offer 2). The UAS accepted Offer 2 by sending a 200 OK response. If media negotiation had failed, the UAS would have sent a 488 Unacceptable Media response instead. Later the UAS sent a 200 OK final response to the initial INVITE request. The UAS sent an ACK request acknowledging the final response to the INVITE request.

Figure 13 UPDATE for Early Media



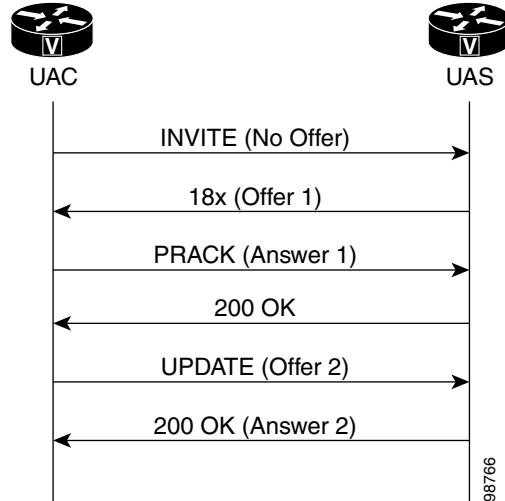
In Figure 14 the gateway received an UPDATE (Offer 2) before responding to the INVITE request (Offer 1), causing the gateway to reject the request by sending a 500 Internal Server Error with a Retry-After header field set to a randomly chosen value between zero and ten seconds.

Figure 14 Initial UPDATE Rejected



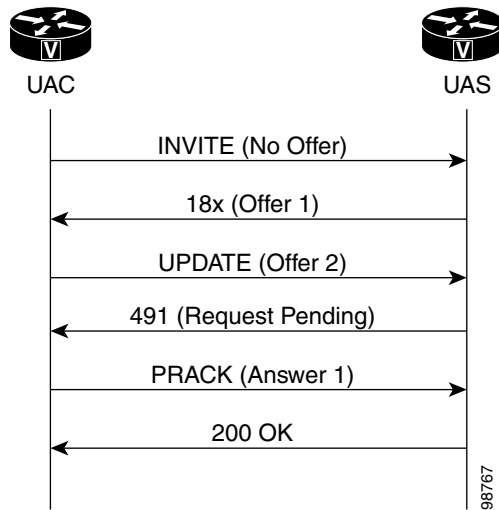
In [Figure 15](#) the initial INVITE request did not contain an offer, and the UAS gateway sent SDP with reliable provisional response (Offer 1) which was treated by the UAC as an offer.

Figure 15 UPDATE Request for Delayed Media



In [Figure 16](#) the UAS received an UPDATE request with an offer (Offer 2) before receiving a PRACK, that is, before the early dialog is established, causing the UAS (gateway) to generate a 491 Request Pending response.

Figure 16 UPDATE Request Failure for Delayed Media

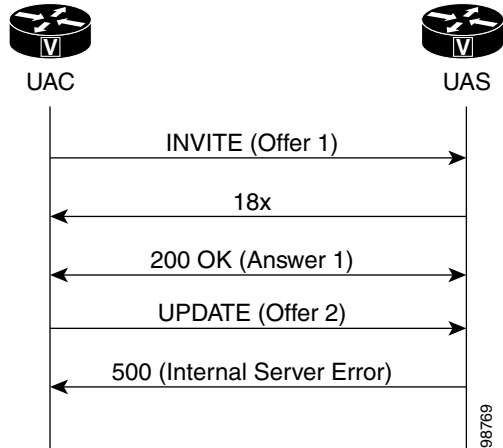


Error Responses to UPDATE Request Processing Before the Call Is Active

In other scenarios, additional rules apply to processing an UPDATE request with an offer when the gateway has sent a 200 OK response to an INVITE request but has not yet received an ACK. The following scenarios generate an error response and are shown in [Figure 17](#):

- If the initial INVITE request contains an offer but does not require provisional responses be sent reliably, then the SDP in the 200 OK is treated like an answer. If the UAS then receives an UPDATE request before an ACK response to the 200 OK, the UAS sends a 500 Server Internal error response with a Retry-After header.
- If the initial INVITE does not contain an offer and does not require provisional responses be sent reliably, then the SDP in the 200 OK is treated like an offer. If the UAS then receives an UPDATE request before receiving an ACK to the 200 OK, the UAS sends a 491 Request Pending response.

Figure 17 Error Cases for UPDATE Requests



UPDATE Request Processing in the Active State

RFC 3261 recommends using a re-INVITE request, the SIP message that changes session parameters of an existing or pending call, to update session parameters after a call is active. UPDATES received after a call is active are processed like a re-INVITE except that the 200 OK to update is not resent (see Figure 18).

Figure 18 UPDATE Request in the Active State

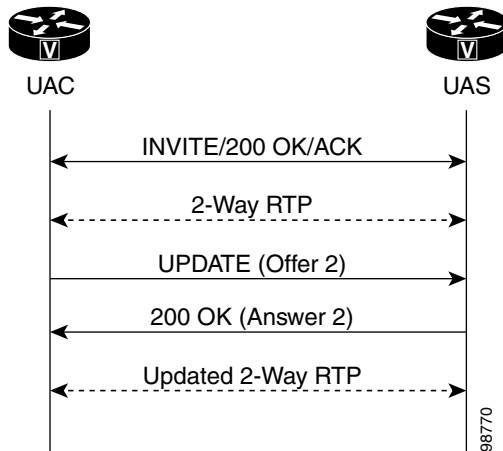
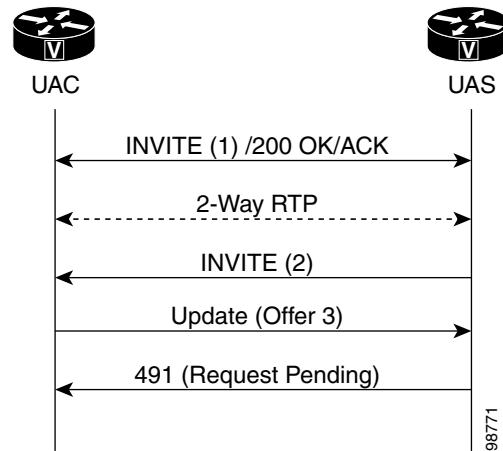


Figure 19 shows a UAC that sent a mid-call INVITE request which has not yet been answered. In this state, when the gateway receives an UPDATE request with a new offer, it sends a 491 Request Pending error.

Figure 19 Error Response to an UPDATE Request in the Active State



Via Header Parameters and Merged Request Detection

To meet specifications of RFC 3261, the SIP Extensions for Caller Identity and Privacy feature provides support for the branch parameter in the Via header of a request, the information used to identify the transaction created by that request. The branch parameter value begins with the value “z9hG4bK” indicating that the request was generated by a UAC that is RFC 3261 compliant. The SIP Extensions for Caller Identity and Privacy feature also adds support for generating the received parameter with the received address.

The SIP Extensions for Caller Identity and Privacy feature uses the branch and sent-by parameters to detect a merged request, that is, a request that has arrived at the UAS more than once by following different paths. If the request has no tag in the To header field, the UAS checks the request against ongoing transactions. If the From tag, Call-ID, and CSeq headers exactly match those headers associated with an ongoing transaction, but the topmost Via header, including the branch parameter, does not match, the UAS treats the request as merged. The UAS responds to a merged request with a 482 Loop Detected error.

Loose-Routing and the Record-Route Header

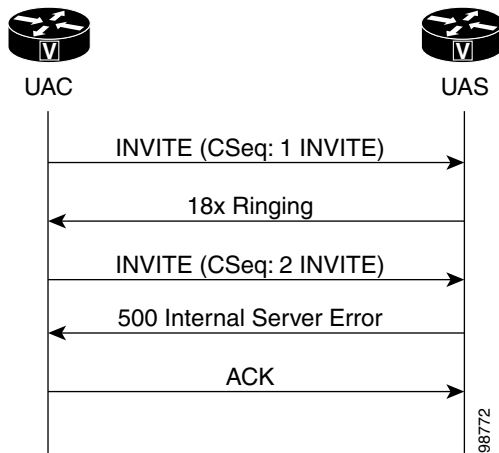
The SIP Extensions for Caller Identity and Privacy feature supports loose-routing, a mechanism that helps keep the request target and next route destination separate. The lr parameter, used in the uniform resource indicator (URI) that a proxy places in the Record-Route header, indicates proxy compatibility with RFC 3261. If the lr parameter is missing from a request, the UA assumes the next-hop proxy implements strict-routing in compliance with RFC 2543, and reformats the message to preserve information in the Request-URI.

Multiple INVITE Requests Before a Final Response

This feature implements support for processing multiple INVITE requests received by the UAS before it sends a final response to the initial INVITE request (see Figure 20). If the UAS gateway receives a second INVITE request before it sends the final response to the first INVITE request with a lower CSeq

sequence number on the same dialog, the UAS returns a 500 Server Internal Error response to the second INVITE request. The error response also includes a Retry-After header field with a random value between 0 and 10 seconds.

Figure 20 Re-INVITE Request Rejected With a 5xx Response

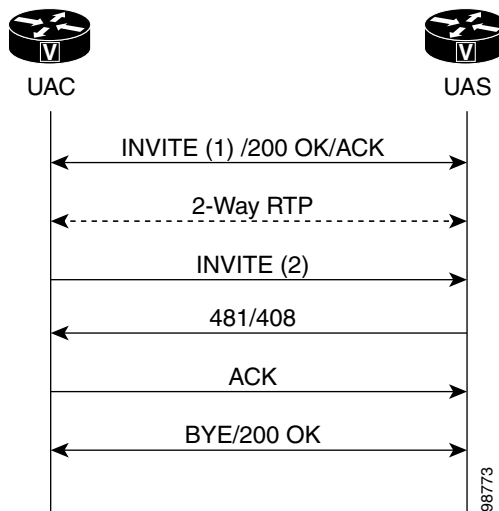


Mid-call Re-INVITE Request Failure

The SIP Extensions for Caller Identity and Privacy feature implements the mid-call re-INVITE request failure treatment shown in Figure 21. The UAC terminates a dialog when a non-2xx final response to a mid-call INVITE request is one of the following:

- A 481 Call/Transaction Does Not Exist failure response
- A 408 Request Timeout failure response

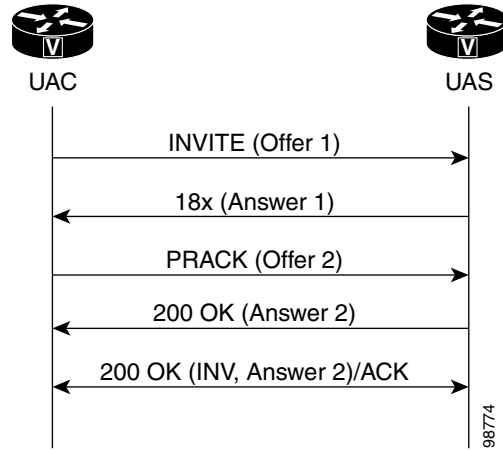
Figure 21 Dialog Termination After a 481 or 408 Response to Re-INVITE Request



PRACK Request with a New Offer

The SIP Extensions for Caller Identity and Privacy feature supports a PRACK request with a new offer (see Figure 22). If the UAC receives a reliable provisional response with an answer (Answer 1), it may generate an additional offer in the PRACK (Offer 2). If the UAS receives a PRACK with an updated offer, it generates a 200 OK with an answer (Answer 2) if negotiation is successful. Otherwise the UAS generates a 488 Unacceptable Media response.

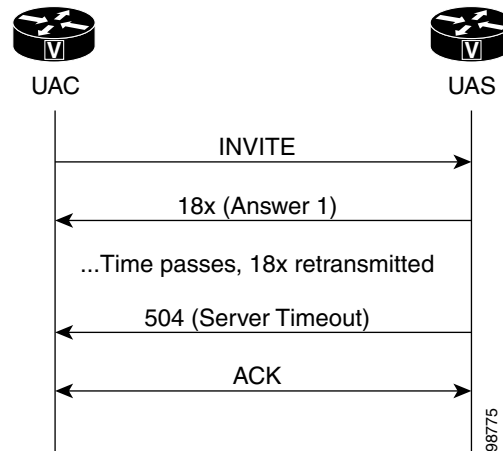
Figure 22 Offer in PRACK Accepted



Reliable Provisional Response Failure

The SIP Extensions for Caller Identity and Privacy feature provides the treatment shown in Figure 23 when the UAS does not receive a corresponding PRACK after resending a 18x reliable provisional response for the maximum number of retries allowed or for 32 seconds. The UAS generates a 5xx response to clear the call.

Figure 23 Reliable Provisional Response Failure



Sample Messages

This section contains sample SIP messages collected at the terminating SIP gateway.

SIP UPDATE Request Call Flow Example

The following example shows an exchange of SIP requests and responses, including an UPDATE request before the call is active:

```
1w0d:SIP Msg:ccsipDisplayMsg:Received:
INVITE sip:222@192.0.2.12:5060 SIP/2.0
Record-Route:<sip:222@192.0.2.4:5060;maddr=192.0.2.4>
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=5,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK1D38
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>
Date:Mon, 08 Apr 2002 16:58:08 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Supported:timer
```

The next line shows the UAC requires the provisional response be reliably transported.

```
Require:100rel
Min-SE: 1800
Cisco-Guid:2729535908-1246237142-2148443152-4064420637
User-Agent:Cisco-SIPGateway/IOS-12.x
```

The Allow header shows that the UPDATE method is supported.

```
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq:101 INVITE
Max-Forwards:70
Remote-Party-ID:<sip:111@192.0.2.14>;party=calling;screen=no;privacy=off
Timestamp:1018285088
Contact:<sip:111@192.0.2.14:5060>
Expires:180
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:262
```

The following SDP constitutes the initial offer, including media streams and codecs, along with IP addresses and ports to receive media.

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 6579 1987 IN IP4 192.0.2.14
s=SIP Call
c=IN IP4 192.0.2.14
t=0 0
m=audio 17782 RTP/AVP 8 0 18 19
c=IN IP4 192.0.2.14
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
```

```
1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=5,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK1D38
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Sat, 07 Oct 2000 02:56:34 GMT
```

```

Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Timestamp:1018285088
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE

```

```

Allow-Events:telephone-event
Content-Length:0

```

In the following lines, the gateway responds by sending early media in answer to the initial offer.

```

1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 183 Session Progress
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=5,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK1D38
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Sat, 07 Oct 2000 02:56:34 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Timestamp:1018285088
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Require:100rel
RSeq:5785
Allow:UPDATE
Allow-Events:telephone-event
Contact:<sip:222@192.0.2.12:5060>
Record-Route:<sip:222@192.0.2.4:5060;maddr=192.0.2.4>
Content-Disposition:session;handling=required
Content-Type:application/sdp
Content-Length:191

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 5565 7580 IN IP4 192.0.2.12
s=SIP Call
c=IN IP4 192.0.2.12
t=0 0
m=audio 18020 RTP/AVP 8 19
c=IN IP4 192.0.2.12
a=rtpmap:8 PCMA/8000
a=rtpmap:19 CN/8000

```

The following lines show the UAS receiving a PRACK for the 183 response.

```

1w0d:SIP Msg:ccsipDisplayMsg:Received:
PRACK sip:222@192.0.2.12:5060 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=6,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK40A
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Mon, 08 Apr 2002 16:58:08 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
CSeq:102 PRACK
RAck:5785 101 INVITE
Content-Length:0

```

```

1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=6,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK40A
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Sat, 07 Oct 2000 02:56:34 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Server:Cisco-SIPGateway/IOS-12.x

```

```
CSeq:102 PRACK
Content-Length:0
```

The next lines show the UAS receiving an updated offer with different media streams and codecs.

```
1w0d:SIP Msg:ccsipDisplayMsg:Received:
UPDATE sip:222@192.0.2.12:5060 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=z9hG4bK10
Via:SIP/2.0/UDP 192.0.2.14:5060
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
CSeq:103 UPDATE
Contact:sip:111@192.0.2.14:5060
Content-Length:262
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 6579 1987 IN IP4 192.0.2.14
s=SIP Call
c=IN IP4 192.0.2.14
t=0 0
m=audio 17782 RTP/AVP 8 0 18 19
c=IN IP4 192.0.2.14
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
```

The new offer in the UPDATE request is acceptable to the server, so it responds with the corresponding answer in the 200 OK message.

```
1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=z9hG4bK10,SIP/2.0/UDP 192.0.2.14:5060
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Sat, 07 Oct 2000 02:56:34 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Server:Cisco-SIPGateway/IOS-12.x
CSeq:103 UPDATE
Content-Type:application/sdp
Content-Length:191
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5565 7580 IN IP4 192.0.2.12
s=SIP Call
c=IN IP4 192.0.2.12
t=0 0
m=audio 18020 RTP/AVP 8 19
c=IN IP4 192.0.2.12
a=rtpmap:8 PCMA/8000
a=rtpmap:19 CN/8000
```

```
1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=5,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK1D38
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Sat, 07 Oct 2000 02:56:34 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Timestamp:1018285088
Server:Cisco-SIPGateway/IOS-12.x
```

```

CSeq:101 INVITE
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
Allow-Events:telephone-event
Contact:<sip:222@192.0.2.12:5060>
Record-Route:<sip:222@192.0.2.4:5060;maddr=192.0.2.4>
Content-Type:application/sdp
Content-Length:191

v=0
o=CiscoSystemsSIP-GW-UserAgent 5565 7580 IN IP4 192.0.2.12
s=SIP Call
c=IN IP4 192.0.2.12
t=0 0
m=audio 18020 RTP/AVP 8 19
c=IN IP4 192.0.2.12
a=rtpmap:8 PCMA/8000
a=rtpmap:19 CN/8000

1w0d:SIP Msg:ccsipDisplayMsg:Received:
ACK sip:222@192.0.2.12:5060 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=7,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK230
From:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
To:<sip:222@192.0.2.4>;tag=24D435A8-C29
Date:Mon, 08 Apr 2002 16:58:08 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Max-Forwards:70
CSeq:101 ACK
Content-Length:0

1w0d:SIP Msg:ccsipDisplayMsg:Sent:
BYE sip:222@192.0.2.4:50605060;maddr=192.0.2.4 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.12:5060;branch=z9hG4bKCA
From:<sip:222@192.0.2.4>;tag=24D435A8-C29
To:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
Date:Sat, 07 Oct 2000 02:56:35 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:70
Route:<sip:111@192.0.2.14:5060>
Timestamp:970887414
CSeq:101 BYE
Content-Length:0

1w0d:SIP Msg:ccsipDisplayMsg:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.12:5060;branch=z9hG4bKCA
From:<sip:222@192.0.2.4>;tag=24D435A8-C29
To:<sip:111@192.0.2.14>;tag=3DD33DE4-10DF
Date:Mon, 08 Apr 2002 16:58:29 GMT
Call-ID:A2B205CC-4A4811D6-8010A410-F242231D@192.0.2.14
Server:Cisco-SIPGateway/IOS-12.x
Timestamp:970887414
Content-Length:0
CSeq:101 BYE

```

Loose-Routing Call Flow Example

The following sample message shows a loose-routing request:

```

1w0d:SIP Msg:ccsipDisplayMsg:Received:
INVITE sip:222@192.0.2.12:5060 SIP/2.0

```

The SIP messages in the following call flow have the Request-URI set to the SIP URI of the destination UA instead of the SIP URI of the next-hop destination, that is, the SIP proxy server.

```
Record-Route:<sip:222@192.0.2.4:5060;lr;maddr=192.0.2.4>
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=9,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK2394
From:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
To:<sip:222@192.0.2.4>
Date:Mon, 08 Apr 2002 16:58:34 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Supported:timer
Min-SE: 1800
Cisco-Guid:2991015782-1246237142-2148770832-4064420637
User-Agent:Cisco-SIPGateway/IOS-12.x
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq:101 INVITE
Max-Forwards:70
Remote-Party-ID:<sip:111@192.0.2.14>;party=calling;screen=no;privacy=off
Timestamp:1018285114
Contact:<sip:111@192.0.2.14:5060>
Expires:180
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:262
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 1981 1761 IN IP4 192.0.2.14
s=SIP Call
c=IN IP4 192.0.2.14
t=0 0
m=audio 18354 RTP/AVP 8 0 18 19
c=IN IP4 192.0.2.14
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
```

```
1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=9,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK2394
From:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
To:<sip:222@192.0.2.4>;tag=24D49BE8-2346
Date:Sat, 07 Oct 2000 02:57:00 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Timestamp:1018285114
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Content-Length:0
```

```
1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 180 Ringing
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=9,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK2394
From:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
To:<sip:222@192.0.2.4>;tag=24D49BE8-2346
Date:Sat, 07 Oct 2000 02:57:00 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Timestamp:1018285114
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
```

```
Allow:UPDATE
Allow-Events:telephone-event
Contact:<sip:222@192.0.2.12:5060>
Record-Route:<sip:222@192.0.2.4:5060;lr;maddr=192.0.2.4>
Content-Length:0

1w0d:SIP Msg:ccsipDisplayMsg:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=9,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK2394
From:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
To:<sip:222@192.0.2.4>;tag=24D49BE8-2346
Date:Sat, 07 Oct 2000 02:57:00 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Timestamp:1018285114
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
Allow-Events:telephone-event
Contact:<sip:222@192.0.2.12:5060>
Record-Route:<sip:222@192.0.2.4:5060;lr;maddr=192.0.2.4>
Content-Type:application/sdp
Content-Length:191

v=0
o=CiscoSystemsSIP-GW-UserAgent 5181 4737 IN IP4 192.0.2.12
s=SIP Call
c=IN IP4 192.0.2.12
t=0 0
m=audio 16720 RTP/AVP 8 19
c=IN IP4 192.0.2.12
a=rtpmap:8 PCMA/8000
a=rtpmap:19 CN/8000

1w0d:SIP Msg:ccsipDisplayMsg:Received:
ACK sip:222@192.0.2.12:5060 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.4:5060;branch=10,SIP/2.0/UDP
192.0.2.14:5060;branch=z9hG4bK103D
From:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
To:<sip:222@192.0.2.4>;tag=24D49BE8-2346
Date:Mon, 08 Apr 2002 16:58:34 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Max-Forwards:70
CSeq:101 ACK
Content-Length:0

1w0d:SIP Msg:ccsipDisplayMsg:Sent:
BYE sip:111@192.0.2.14:5060 SIP/2.0
Via:SIP/2.0/UDP 192.0.2.12:5060;branch=z9hG4bK18B6
From:<sip:222@192.0.2.4>;tag=24D49BE8-2346
To:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
Date:Sat, 07 Oct 2000 02:57:01 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:70
Route:<sip:222@192.0.2.4:5060;lr;maddr=192.0.2.4>
Timestamp:970887440
CSeq:101 BYE
Content-Length:0

1w0d:SIP Msg:ccsipDisplayMsg:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 192.0.2.12:5060;branch=z9hG4bK18B6
```

```
From:<sip:222@192.0.2.4>;tag=24D49BE8-2346
To:<sip:111@192.0.2.14>;tag=3DD3A404-12A3
Date:Mon, 08 Apr 2002 16:58:54 GMT
Call-ID:B2474766-4A4811D6-8015A410-F242231D@192.0.2.14
Server:Cisco-SIPGateway/IOS-12.x
Timestamp:970887440
Content-Length:0
CSeq:101 BYE
```

SIP RFC 3261, RFC 3262, and RFC 3264 Compliance

The Internet Engineering Task Force (IETF) continually updates SIP standards. This feature describes the specific updates or optimizations that were made on Cisco SIP gateways to remain in compliance with the IETF. The following standards have been updated:

- RFC 3261: Core Standard for SIP (obsoleting RFC 2543)
- RFC 3262: Standard for Reliability of Provisional Responses in SIP
- RFC 3264: Standard for Offer/Answer Model with Session Description Protocol (SDP)

To provide quality service to our SIP customers, Cisco optimizes its SIP gateways to comply with the latest SIP-related RFCs. In addition, backward compatibility is maintained, providing customers interoperability with gateways that do not yet support the current RFCs.

This section contains the following information:

- [SIP Messaging Enhancements, page 32](#)
- [SIP TCP and UDP Connection Enhancements, page 33](#)
- [Dynamic Transport Switching \(UDP to TCP\) for Large SIP Requests, page 34](#)
- [Call-Hold Enhancement, page 34](#)
- [Expanded Range of the max-forwards Command, page 35](#)

SIP Messaging Enhancements

The following changes or additions were made to SIP messaging:

- This feature is in compliance with RFC 3261. If a user agent server (UAS) generates a 2xx request and is waiting for an acknowledgement (ACK), and the call disconnects at the server side, the UAS does not send a BYE message immediately. The UAS sends a BYE message when the retry timer times out or when the ACK response is received. The BYE message terminates the call to prevent hung networks.
- In compliance with RFC 3261, the user agent (UA) cannot send a BYE message until it receives an ACK response from the originating gateway. This enhancement prevents a race condition, which is when a BYE response arrives at the terminating gateway before the 200 OK response. This enhancement applies to normal disconnects and not to disconnects due to timeouts or errors.
- In compliance with RFC 3262, the user agent client (UAC) now waits for a 1xx provisional response (PRACK) from the terminating gateway before sending a Cancel request to an Invite request. Waiting for a 1xx response prevents resources from being held up, which can happen if the Cancel request arrives at the terminating gateway before the Invite message.

- In compliance with RFC 3261, a Cisco SIP gateway returns a 491 Request Pending response when it receives an Invite requesting session modification on a dialog while an Invite request is still in progress. The gateway that sent the re-Invite and that receives the 491 response starts a timer with a randomly chosen value. When the timer expires, the gateway attempts the Invite request again if it still desires the session modification to take place.

If the UAC generated the request, the timer has a randomly chosen value between 2.1 and 4 seconds, in units of 10 ms. If the UAC did not generate the request, the timer has a randomly chosen value between 0 and 2 seconds, in units of 10 ms.

SIP TCP and UDP Connection Enhancements

Prior to RFC 3261, TCP support was optional for SIP user agents. RFC 3261 now requires support for both UDP and TCP. While Cisco SIP gateways already supported TCP, there have been several optimizations that are described below:

- [Failed Transmissions of 2xx Responses, page 33](#)
- [Reuse of TCP and UDP Connections, page 33](#)
- [Transaction-Based Transport Switching and Usage, page 33](#)
- [Detection of Remote End Connection Closures, page 34](#)
- [Creation of New Connections for Sending Responses in Case the Original Connection Dropped, page 34](#)

Failed Transmissions of 2xx Responses

The transmission of 2xx responses is in compliance with RFC 3261. If the transport is TCP and a gateway does not receive an acknowledgement to a 2xx response it sent to an INVITE message, the gateway retries the 2xx response over TCP. The retry ensures that a gateway receives a 200 OK message, eliminating the possibility that the 2xx response is lost when hops over the network use an unreliable transport such as UDP.

Reuse of TCP and UDP Connections

Prior to RFC 3261, a remote gateway could not initiate two requests over the same TCP connection. In addition, the gateway created a new connection for each new transaction, and after the completion of a transaction, the gateway closed the connection. Closing the connection, even if a subsequent request was destined for the same location as the previous transaction, resulted in potentially lower performance due to the large number of unnecessary open/close connections. With Cisco IOS Release 12.3(8)T, the gateway opens one TCP connection per remote IP address and port. The gateway opens a new connection only if a connection to the particular destination IP address and port is not already present. The gateway closes the connection when all requests that use that connection have terminated and no activity is detected for a given time period.

The **timers connection** command allows you to time out a TCP or UDP connection because of inactivity.

Transaction-Based Transport Switching and Usage

With Cisco IOS Release 12.3(8)T, if a new transaction request is larger than the threshold switchable value, it is sent over TCP. The threshold switchable value is a value that is 200 bytes or more than the interface or path's MTU. If the message size is smaller than the threshold switchable value, the original configured transport is used. The original transport means the transport configured under the dial peer

for the initial Invite request or the transport specified in the incoming response's Contact or Record-Route headers in subsequent requests. In other words, the transport usage is now transaction-based instead of call-based.

Detection of Remote End Connection Closures

Remote gateway closures that go undetected can result in hung TCP connections. If a closed connection remains undetected, the corresponding connection entry is never removed from the connection table. Continuous occurrences of undetected closures can lead to the connection table being filled with invalid entries and valid SIP requests being rejected, requiring a router reboot. With Cisco IOS Release 12.3(8)T, the SIP gateway uses internal mechanisms to detect remote closures and to clean up the connection table. No user input is required to initiate the cleanup.

Creation of New Connections for Sending Responses in Case the Original Connection Dropped

With Cisco IOS Release 12.3(8)T, if a gateway tears down the connection of an incoming request before a response is sent, the receiving gateway creates a new connection to send out a response. The new connection is based on the port specified in the sent-by parameter of the Via header. Prior to Cisco IOS Release 12.3(8)T, a dropped connection resulted in failure of the call.

Dynamic Transport Switching (UDP to TCP) for Large SIP Requests

RFC 3261 states that large SIP requests, requests within 200 bytes of the maximum transmission unit (MTU), should be transmitted over TCP. Transport over TCP avoids UDP fragmentation, and the switch to TCP can occur even if the gateway is configured to use UDP. If the TCP transmission fails (for example if the terminating gateway does not support TCP), the message is then retried over UDP.

The capability to configure the MTU size on an Ethernet or Fast Ethernet interface already exists on the Cisco SIP gateways. If the MTU is not configured, the default MTU value is 1500 bytes. Assuming an MTU of 1500 bytes, requests larger than 1300 bytes are considered the threshold value for dynamic transport switching.

Two commands allow the user to enable or disable support for dynamic switching. Use the commands to avoid interoperability issues with gateways that do not support TCP and to maintain backward compatibility. The **transport switch** command can be configured at the global level, and the **voice-class sip transport switch** command can be configured at the dial peer level. The global configuration is considered only when there is no matching VoIP dial peer.

This feature is disabled by default.

Call-Hold Enhancement

RFC 3264 recommends that call-hold be initiated using the direction attribute (a=sendonly) in SDP. Cisco SIP gateways follow the new guideline, and SIP gateways can now initiate call-hold using either one of the two ways. The **offer call-hold** command allows the user to globally specify the format to initiate call-hold. That is, the gateway should use a=sendonly or conn addr=0.0.0.0; it cannot set usage to both. The default configuration is a=sendonly, because this is the RFC recommended method. Specifying a call-hold format is not available at the dial peer level.



Note

Cisco SIP gateways support receiving call-hold requests in either of the two formats, but use of the direction attribute is recommended.

Expanded Range of the max-forwards Command

In compliance with RFC 3261, the **max-forwards** command was enhanced with a greater configurable range (1 to 70) and a higher default value (70).

How to Configure SIP RFC Compliance

This section contains the following procedures:

- [Configuring Compliance to RFC 2543, page 35](#)
- [Configuring Compliance to RFC 2782, page 35](#)
- [Configuring Compliance to RFC 3261, page 36](#)
- [Configuring Compliance to RFC 3261, RFC 3262, and RFC 3264, page 36](#)
- [Verifying SIP RFC Compliance, page 42](#)
- [Troubleshooting Tips, page 45](#)



Note

- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP RFC Compliance” section on page 2](#)
 - [“Restrictions for SIP RFC Compliance” section on page 2](#)
- For help with a procedure, see the verification and troubleshooting sections listed above.

Configuring Compliance to RFC 2543

No configuration tasks are required to enable RFC 2543. It is enabled by default.

Configuring Compliance to RFC 2782

To configure compliance with RFC 2782, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **srv version**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	srv version {1 2} Example: Router(config-sip-ua)# srv version 2	Generates DNS SRV queries with either RFC 2052 or RFC 2782 format. Keywords are as follows: <ul style="list-style-type: none"> • 1—Domain-name prefix of format protocol.transport. (RFC 2052 style) • 2—Domain-name prefix of format _protocol._transport. (RFC 2782 style) Default: 2.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring Compliance to RFC 3261

No configuration tasks are required to enable RFC 3261. It is enabled by default.

Configuring Compliance to RFC 3261, RFC 3262, and RFC 3264

This section contains the following procedures:

- [Configure SIP Messaging, page 37](#)
- [Configure TCP and UDP Connection Enhancements, page 37](#)
- [Configure Dynamic Transport Switching \(UDP to TCP\) for Large SIP Requests, page 38](#)
- [Configure Call-Hold, page 40](#)
- [Configure Max Forwards, page 41](#)

Configure SIP Messaging

No configuration is necessary.

Configure TCP and UDP Connection Enhancements

To set the time before the SIP UA ages out a TCP or UDP connection because of inactivity, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers connection aging**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers connection aging <i>timer-value</i> Example: Router(config-sip-ua)# timers connection aging 5	Sets the time before the SIP UA ages out a TCP or UDP connection because of inactivity. The argument is as follows: <ul style="list-style-type: none"> • <i>timer-value</i>—Time, in minutes, to wait. Range: 5 to 30. Default: 5.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configure Dynamic Transport Switching (UDP to TCP) for Large SIP Requests

RFC 3261 states that large SIP requests, within 200 bytes of the maximum transmission unit (MTU), should be transmitted over TCP. Transport over TCP avoids UDP fragmentation, and the switch to TCP can occur even if the gateway is configured to use UDP.

The configurations below describe setting the gateway to switch from UDP to TCP. The default MTU configuration of 1500 bytes on the interface is assumed. After configuration, the threshold value is 1300 bytes—that is, for all SIP requests over 1300 bytes, TCP is the transport mechanism.

You can configure dynamic transport switching on a dial-peer or global basis.

- [Configuring Dynamic Transport Switching for Large SIP Requests on a Dial-Peer Basis, page 38](#)
- [Configuring Dynamic Transport Switching for Large SIP Requests on a Global Basis, page 39](#)

Configuring Dynamic Transport Switching for Large SIP Requests on a Dial-Peer Basis

To configure switching between UDP and TCP transport mechanisms for a specific dial peer, perform the following steps.



Note

- Dynamic transport switching from UDP to TCP is disabled by default.
- When the dynamic transport switching mechanism is enabled in dial-peer voice configuration mode, it takes precedence over the global configuration.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **voice-class sip transport switch udp tcp**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<code>dial-peer voice tag voip</code> Example: Router(config)# dial-peer voice 25 voip	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	<code>voice-class sip transport switch udp tcp</code> Example: Router(config-dial-peer)# voice-class sip transport switch udp tcp	Enables switching between UDP and TCP transport mechanisms for large SIP messages for a specific dial peer. Keywords are as follows: <ul style="list-style-type: none"> • udp—Switching transport from UDP on the basis of the size of the SIP request being greater than the MTU size. • tcp—Switching transport to TCP.
Step 5	<code>exit</code> Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring Dynamic Transport Switching for Large SIP Requests on a Global Basis

To configure switching between UDP and TCP transport mechanisms on all the connections of a Cisco SIP gateway, perform the following steps.



Note

- Dynamic transport switching from UDP to TCP is disabled by default.
- When the dynamic transport switching mechanism is enabled in dial-peer voice configuration mode, it takes precedence over the global configuration. Consider the global configuration described below only when there is no matching VoIP dial peer.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `transport switch udp tcp`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: <code>Router> enable</code>	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: <code>Router# configure terminal</code>	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: <code>Router(config)# voice service voip</code>	Enters voice-service configuration mode.
Step 4	<code>sip</code> Example: <code>Router(config-voi-srv)# sip</code>	Enters SIP configuration mode.
Step 5	<code>transport switch udp tcp</code> Example: <code>Router(conf-serv-sip)# transport switch udp tcp</code>	Enables switching between UDP and TCP transport mechanisms globally for large SIP messages. Keywords are as follows: <ul style="list-style-type: none"> • udp— Switching transport from UDP based on the size of the SIP request being greater than the MTU size. • tcp— Switching transport to TCP.
Step 6	<code>exit</code> Example: <code>Router(conf-serv-sip)# exit</code>	Exits the current mode.

**Note**

Use the following commands to aid in verifying and troubleshooting the SIP transport and connection configurations:

- **debug ccsip transport**
- **show sip-ua connections**

To learn more about these commands as well as other verification and troubleshooting commands, see the [“Verifying SIP RFC Compliance”](#) section on page 42 and [“Troubleshooting Tips”](#) section on page 45.

Configure Call-Hold

To specify how the SIP gateway should initiate call-hold requests, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **offer call-hold**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	offer call-hold {conn-addr direction-attr} Example: Router(config-sip-ua)# offer call-hold direction-attr	Specifies how the SIP gateway should initiate call-hold requests. Keywords are as follows: <ul style="list-style-type: none"> • conn-addr—RFC 2543/RFC 3261 method of using the connection address for initiating call-hold requests. Uses 0.0.0.0. • direction-attr—RFC 3264 method of using the direction attribute for initiating call-hold requests. Uses the direction attribute in SDP.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configure Max Forwards

To set the maximum number of proxy or redirect servers that can forward the SIP request, perform the following steps.

SUMMARY STEPS

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

3. **sip-ua**
4. **max-forwards**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	max-forwards <i>number</i> Example: Router(config-sip-ua)# max-forwards 65	Sets the maximum number of hops—that is, proxy or redirect servers that can forward the SIP request. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Number of forwards. Range: 1 to 70. Default: 70.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Verifying SIP RFC Compliance

To verify SIP RFC compliance, perform the following steps as appropriate (commands are listed in alphabetical order).



Note

A typical verification sequence involves use of one of the **show sip-ua connections** commands to view call statistics, followed by judicious use of the **clear sip-ua tcp connection** or **clear sip-ua udp connection** command to clear those statistics.

SUMMARY STEPS

1. **show sip-ua connections**
2. **show sip-ua statistics**

DETAILED STEPS

Step 1 show sip-ua connections

Use this command, after a call is made, to learn connection details.

The following sample output shows multiple calls to multiple destinations. This example shows UDP details, but the command output looks identical for TCP calls.

```
Router# show sip-ua connections udp detail

Total active connections : 2
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
-----Printing Detailed Connection Report-----
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=====
5060 1 Established 0
Remote-Agent:172.19.154.18, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=====
5060 2 Established 0
```

The following sample output shows sequential display and clearing of call statistics for connection to a particular target (in this case, 172.18.194.183, port 5060).



Caution

Take care when you use the **clear** commands. Inappropriate usage without understanding the issue or the implications can lead to erroneous call behavior, inappropriate usage of connections, and call failures.

1. Output for the **show sip-ua connections** command displays call statistics:

```
Router# show sip-ua connections tcp detail

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
-----Printing Detailed Connection Report-----
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
=====
5060 1 Established 0
```

2. Output for the **clear sip-ua tcp connection** command shows that statistics are being cleared:

```
Router# clear sip-ua tcp connection id 1 target ipv4:172.18.194.183:5060
```

```
Purging the entry from sip tcp process
Purging the entry from reusable global connection table
```

3. Output for the **show sip-ua connections** command verifies that all connections are cleared as expected:

```
Router# show sip-ua connections tcp detail
```

```
Total active connections : 0
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Max. tcp send msg queue size of 1, recorded for 172.18.194.183:5060
-----Printing Detailed Connection Report-----
Note:
** Tuples with no matching socket entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
- Do 'clear sip <tcp/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition
Remote-Agent:172.18.194.183, Connections-Count:0
```

Step 2 show sip-ua statistics

Use this command to display SIP statistics, including UPDATE requests.

```
Router# show sip-ua statistics
```

```
SIP Response Statistics (Inbound/Outbound)
Informational
  Trying 1/4, Ringing 0/0,
  Forwarded 0/0, Queued 0/0,
  SessionProgress 1/4
Success:
  OkInvite 1/2, OkBye 1/2,
  OkCancel 0/2, OkOptions 0/0,
  OkPrack 1/4, OkPreconditionMet 0/0,
  OkSubscribe 0/0, OkNotify 0/0,
  OkInfo 0/0, 202Accepted 0/0,
OkUpdate 0/0
Redirection (Inbound only):
  MultipleChoice 0, MovedPermanently 0,
  MovedTemporarily 0, UseProxy 0,
  AlternateService 0
Client Error:
  BadRequest 0/0, Unauthorized 0/0,
  PaymentRequired 0/0, Forbidden 0/0,
  NotFound 0/0, MethodNotAllowed 0/0,
  NotAcceptable 0/0, ProxyAuthReqd 0/0,
  ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
  ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
  UnsupportedMediaType 0/0, BadExtension 0/0,
  TempNotAvailable 0/0, CallLegNonExistent 0/0,
  LoopDetected 0/0, TooManyHops 0/0,
  AddrIncomplete 0/0, Ambiguous 0/0,
  BusyHere 0/0, RequestCancel 0/2,
  NotAcceptableMedia 0/0, BadEvent 0/0,
  SETooSmall 0/0, RequestPending 0/0
```

```

Server Error:
  InternalError 0/0, NotImplemented 0/0,
  BadGateway 0/0, ServiceUnavail 2/0,
  GatewayTimeout 0/0, BadSipVer 0/0,
  PreCondFailure 0/0
Global Failure:
  BusyEverywhere 0/0, Decline 0/0,
  NotExistAnywhere 0/0, NotAcceptable 0/0
Miscellaneous counters:
  RedirectRspMappedToClientErr 0
SIP Total Traffic Statistics (Inbound/Outbound)
  Invite 4/4, Ack 4/3, Bye 2/1,
  Cancel 2/0, Options 0/0,
  Prack 4/1, Comet 0/0,
  Subscribe 0/0, Notify 0/0,
  Refer 0/0, Info 0/0,
  Update 0/0
Retry Statistics
  Invite 1, Bye 0, Cancel 0, Response 0,
  Prack 0, Comet 0, Reliable1xx 0, Notify 0

SDP application statistics:
  Parses: 6, Builds 10
  Invalid token order: 0, Invalid param: 0
  Not SDP desc: 0, No resource: 0
Last time SIP Statistics were cleared: <never>

```

Troubleshooting Tips



Note

For general troubleshooting tips and a list of important **debug** commands, see the [“General Troubleshooting Tips” section on page 18](#).

- Use the **debug ccsip all** command to enable SIP-related debugging.
- Use the **debug ccsip transport** command to debug transport and connection related operations while sending out an Invite Message.

Sample output of some of these commands is shown below:

Sample Output for the debug ccsip transport Command

The operations captured here show the following:

- That the connection is established and the Invite was sent.
- That UDP is the transport of the initial Invite message.
- Remote target details; that is where the request is to be sent.
- That the size of the message exceeded the threshold size of the MTU. Therefore transport switching (from UDP to TCP) is enabled.
- That the connection algorithm is started; that is, the counter starts to age out the TCP or UDP connection if inactivity occurs.

```

Router# debug ccsip transport
.
.

```

```

.
lwd: //18/8E16980D800A/SIP/Transport/sipSPISendInvite: Sending Invite to the transport
layer
lwd: //18/8E16980D800A/SIP/Transport/sipSPIGetSwitchTransportFlag: Return the Global
configuration, Switch Transport is TRUE
lwd: //18/8E16980D800A/SIP/Transport/sipSPITransportSendMessage: msg=0x64082D50,
addr=172.18.194.183, port=5060, sentBy_port=0, is_req=1, transport=1, switch=1,
callBack=0x614FAB58
lwd: //18/8E16980D800A/SIP/Transport/sipSPITransportSendMessage: Proceedable for sending
msg immediately
lwd: //18/8E16980D800A/SIP/Transport/sipTransportLogicSendMsg: switch transport is 1
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportGetInterfaceMtuSize: MTU size for remote
address 172.18.194.183 is 500
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportVerifyMsgForMTUThreshold: Interface MTU
Size 500, Msg Size 1096
lwd: //18/8E16980D800A/SIP/Transport/sipTransportLogicSendMsg: Switching msg=0x64082D50
transport UDP->TCP
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportSetAgeingTimer: Aging timer initiated
for holder=0x64084058, addr=172.18.194.183
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipCreateConnHolder: Created new holder=0x64084058,
addr=172.18.194.183
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportPostRequestConnection: Posting TCP conn
create request for addr=172.18.194.183, port=5060, context=0x64128D5C
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportSetConnWaitTimer: Wait timer set for
connection=0x64129BF4, addr=172.18.194.183, port=5060
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipCreateConnInstance: Created new initiated
conn=0x64129BF4, connid=-1, addr=172.18.194.183, port=5060, transport=tcp
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipConnectionManagerProcessConnCreated:
gConnTab=0x64128D5C, addr=172.18.194.183, port=5060, connid=1, transport=tcp
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipInstanceHandleConnectionCreated: Moving
connection=0x64129BF4, connid=1state to pending
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportProcessNWConnectionCreated:
context=0x64128D5C
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipConnectionManagerProcessConnCreated:
gConnTab=0x64128D5C, addr=172.18.194.183, port=5060, connid=1, transport=tcp
lwd: //-1/xxxxxxxxxxxx/SIP/Transport/sipTransportPostSendMessage: Posting send for
msg=0x64082D50, addr=172.18.194.183, port=5060, connId=1 for TCP
.
.
.

```

Configuration Examples for SIP RFC Compliance

This section provides the following configuration example:

- [SIP Gateway Compliance to RFC 3261, RFC 3262, and RFC 3264: Example, page 46](#)



Note

IP addresses and hostnames in examples are fictitious.

SIP Gateway Compliance to RFC 3261, RFC 3262, and RFC 3264: Example

This section provides a configuration example to match the identified configuration tasks in the previous sections.

```

lwd: %SYS-5-CONFIG_I: Configured from console by console
Building configuration...

```

```
Current configuration : 3326 bytes
!
!Last configuration change at 18:09:20 EDT Fri Apr 23 2004
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
boot-start-marker
boot system tftp mantis/c3640-is-mz.disc_w_pi 172.18.207.10
boot-end-marker
!
clock timezone EST -5
clock summer-time EDT recurring
voice-card 3
!
aaa new-model
!
aaa accounting connection h323 start-stop group radius
aaa nas port extended
aaa session-id common
ip subnet-zero
!
ip cef
ip host example.com 172.18.194.183
ip host CALLGEN-SECURITY-V2 10.36.54.81 10.1.0.0
ip name-server 172.18.192.48
!
isdn switch-type primary-ni
!
trunk group 1
!
voice service voip
    sip
    rellxx require "100rel"
    transport switch udp tcp
!
voice class uri 800 sip
pattern test@example.com
!
controller T1 3/0
    framing sf
    linecode ami
    pri-group timeslots 1-24
!
controller T1 3/1
    framing sf
    linecode ami
    pri-group timeslots 1-24
    gw-accounting aaa
!
interface Ethernet0/0
    description CentreComm Hub port 9 in PP070
    ip address 172.18.194.170 255.255.255.0
    no ip proxy-arp
    ip mtu 500
    half-duplex
    no cdp enable
    ip rsvp bandwidth 100 100
!
interface Serial3/0:23
    no ip address
    no logging event link-status
    isdn switch-type primary-ni
```

```

    isdn incoming-voice voice
    no cdp enable
    !
interface Serial3/1:23
    no ip address
    no logging event link-status
    isdn switch-type primary-ni
    isdn protocol-emulate network
    isdn incoming-voice voice
    no cdp enable
    !
no ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 172.18.194.1
ip route 0.0.0.0 0.0.0.0 Ethernet0/0
ip route 10.0.0.0 255.0.0.0 172.18.194.1
ip route 172.16.0.0 255.0.0.0 Ethernet0/0
    !
dialer-list 1 protocol ip permit
no cdp run
    !
radius-server host 10.13.84.133 auth-port 1645 acct-port 1646
radius-server timeout 2
radius-server key cisco
radius-server vsa send accounting
radius-server vsa send authentication
    !
control-plane
    !
call application voice testapp79 tftp://172.18.207.10/mantis/my_app.tcl
call application voice testapp888 tftp://172.18.207.10/mantis/AL_FEAT_SIP_URL_O_RV_79.tcl
call application voice testapp888 mcid-dtmf 9876
call application voice testapp888 test 5444
    !
voice-port 1/1/0
    !
voice-port 1/1/1
    !
voice-port 3/0:23
    !
voice-port 3/1:23
    !
dial-peer cor custom
    !
dial-peer voice 9876 voip
    destination-pattern 9876
    voice-class sip transport switch udp tcp
    session protocol sipv2
    session target ipv4:172.18.194.183
    session transport udp
    !
dial-peer voice 222 pots
    incoming called-number .
    direct-inward-dial
    !
sip-ua
    max-forwards 65
    retry invite 4
    retry bye 4
    retry cancel 4
    retry comet 4
    retry notify 4
    timers connection aging 15
    offer call-hold conn-addr

```



```
!  
line con 0  
  exec-timeout 0 0  
line vty 0 4  
  password password1  
  
ntp clock-period 17179695  
ntp server 172.18.194.178  
ntp server 10.81.254.131  
!  
end
```

Additional References

General SIP References

- “[SIP Features Roadmap](#)” on page 1—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- “[Overview of SIP](#)” on page 1—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter

- *SIP Gateway Support of RSVP and TEL URL* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122limit/122x/122xb/122xb_2/vvfresrv.htm#1027153

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Configuring SIP Call-Transfer Features

This chapter describes how to configure SIP call-transfer features. It describes the following features:

- SIP - Call Transfer Using Refer Method
- SIP - Call Transfer Enhancements Using Refer Method
- SIP Transfer Using the Refer Method and Call Forwarding
- SIP Stack Portability



Note The SIP Stack Portability feature is described in the “[Configuring SIP, Timer, and Response Features](http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-msg_tmr_rspns.html)” chapter (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-msg_tmr_rspns.html).

Feature History for SIP - Call Transfer Using Refer Method

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB2	The feature was implemented on an additional platform.
12.2(11)T	The feature was integrated into this release and support was added for additional platforms.

Feature History for SIP - Call Transfer Enhancements Using Refer Method

Release	Modification
12.2(13)T	This feature was introduced.

Feature History for SIP Transfer Using the Refer Method and Call Forwarding

Release	Modification
12.2(13)T	This feature was introduced.



Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Prerequisites for SIP Call Transfer, page 2](#)
- [Restrictions for SIP Call Transfer, page 3](#)
- [Information About SIP Call Transfer, page 4](#)
- [How to Configure SIP Call-Transfer Features, page 19](#)
- [Configuration Examples for SIP Call-Transfer Features, page 39](#)
- [Additional References, page 45](#)

Prerequisites for SIP Call Transfer

All SIP Call-Transfer Features

- Establish a working IP network and configure VoIP.



Note

For information about configuring VoIP, see *Enhancements to the Session Initiation Protocol for VoIP on Cisco Access Platforms* (http://www.cisco.com/en/US/docs/ios/12_2t/12_2t11/feature/guide/ftsipgv1.html).

- Ensure that the gateway has voice functionality configured for SIP.
- Ensure that your Cisco router has minimum memory requirements.
- With all SIP call-transfer methods, configure dial peers for correct functioning of the Refer method.



Note

For dial-peer configuration steps, see the “[Configure SIP Call Transfer on a POTS Dial Peer](#)” section on page 20.

- As necessary, configure the router to use Greenwich Mean Time (GMT). SIP requires that all times be sent in GMT. The INVITE is sent in GMT. However, the default for routers is to use Coordinated Universal Time (UTC). To configure the router to use GMT, issue the **clock timezone** command in global configuration mode and specify GMT.

SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications Feature

- Load Cisco IOS Release 12.2(15)T or a later release.
- Configure hookflash signaling.
- Write a Tool Command Language (Tcl) Interactive Voice Response (IVR) 2.0 script that implements Cisco IOS call-transfer and call-forward functionality.

Restrictions for SIP Call Transfer

All SIP Call-Transfer Features

- The SIP gateway does not support codecs other than those listed in the table titled “SIP Codec Support by Platform and Cisco IOS Release” in the “[Enhanced Codec Support for SIP Using Dynamic Payloads](#)” section of the “[Configuring SIP QoS Features](#)” document (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-qos.html).
- SIP requires that all times be sent in GMT.
- Although SIP Cisco IOS gateways currently support SIP URLs and TEL URLs, the Refer-To header and the Also header must be in SIP URL format to be valid. The TEL URL is only supported in the Refer-To header for blind transfer. The TEL URL format cannot be used because it does not provide a host portion, and without one, the triggered Invite request cannot be routed.
- Only three overloaded headers in the Refer-to header are accepted: Accept-Contact, Proxy-Authorization, and Replaces. All other headers present in the Refer-To are ignored.
- The Refer-To and Contact headers are required in the Refer request. The absence of either header results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Refer-To header. Multiple Refer-To headers result in a 4xx class response.
- The Referred-By header is required in a Refer request. The absence of this header results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Referred-By header. Multiple Referred-By headers result in a 4xx class response.
- With all SIP call-transfer methods, dial peers must be configured for correct functioning of the Refer method.



Note For dial-peer configuration steps, see the “[Configure SIP Call Transfer on a POTS Dial Peer](#)” section on page 20.

- With call transfer using the Bye method, the Requested-By header identifies the party initiating the transfer. The Requested-By header is included in the INVITE request that is sent to the transferred-to party only if a Requested-By header was also included in the Bye request.
- With call transfer using the Also method, the Also header identifies the transferred-to party. To invoke a transfer, the user portion of the Also header must be defined explicitly or with wildcards as a destination pattern on a VoIP dial peer. The transferred call is routed using the session target parameter on the dial peer instead of the host portion of the Also header. Therefore, the Also header can contain *user@host*, but the *host* portion is ignored for call routing purposes.
- The grammar for the Also and Requested-By headers is not fully supported. Only the name-addr is supported. This implies that the crypto-param, which might be present in the Bye request, is not populated in the ensuing Invite to the transferred-to party.
- Cisco SIP gateways do not support the “user=np-queried” parameter in a Request URI.
- If a Cisco SIP gateway receives an ISDN Progress message, it generates a 183 Session progress message. If the gateway receives an ISDN ALERT, it generates a 180 Ringing message.
- The SIP gateway requires each INVITE to include a Session Description Protocol (SDP) header.
- The contents of the SDP header cannot change between the 180 Ringing message and the 200 OK message.

- VoIP dial peers allow a user to configure the **bytes** parameter associated with a codec. Cisco SIP gateways present or respond to the **a=ptime** parameter in the SDP body of a SIP message. However, only one **a=ptime** attribute is allowed per m-line block.
- If early transfer is attempted, and the call between the originator and final-recipient involves QoS or RSVP, the triggered Invite from the recipient with the Replaces header is not processed and the transfer fails. The session between the originator and the final-recipient remains unchanged.

SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications Feature

- SIP call transfer and call forwarding using Tcl IVR 2.0 and VoiceXML applications feature is supported only through Tcl IVR 2.0 and VoiceXML applications; the feature is not supported for Tcl IVR 1.0 applications or the DEFAULT session application.
- Only Cisco 1700 series, Cisco 2600 series, and Cisco 3600 series routers support the initiating of call transfer and call redirection.
- Cisco SIP customer premise equipment (CPE) such as 79xx and Analog Telephone Adaptors (ATAs) do not currently support TEL URLs.
- RLT on CAS or analog (FXS) ports are necessary to initiate SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- SIP call forwarding is supported only on ephones—IP phones that are not configured on the gateway. FXS, FXO, T1, E1, and CAS phones are not supported.
- In Cisco IOS Release 12.2(15)T, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

Information About SIP Call Transfer

To configure SIP call-transfer features, you should understand the following concepts:

- [SIP Call-Transfer Basics, page 4](#)
- [SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications, page 14](#)

SIP Call-Transfer Basics

This section contains the following information:

- [Basic Terminology of SIP Call Transfer, page 4](#)
- [Types of SIP Call Transfer Using the Refer Method, page 7](#)

Basic Terminology of SIP Call Transfer

The Refer method provides call-transfer capabilities to supplement the Bye and Also methods already implemented on Cisco IOS SIP gateways.

Call transfer allows a wide variety of decentralized multiparty call operations. These decentralized call operations form the basis for third-party call control, and thus are important features for 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.

Refer Method

The SIP Refer method provides call-transfer capabilities to supplement the Bye and Also methods already implemented on Cisco IOS SIP gateways. 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.



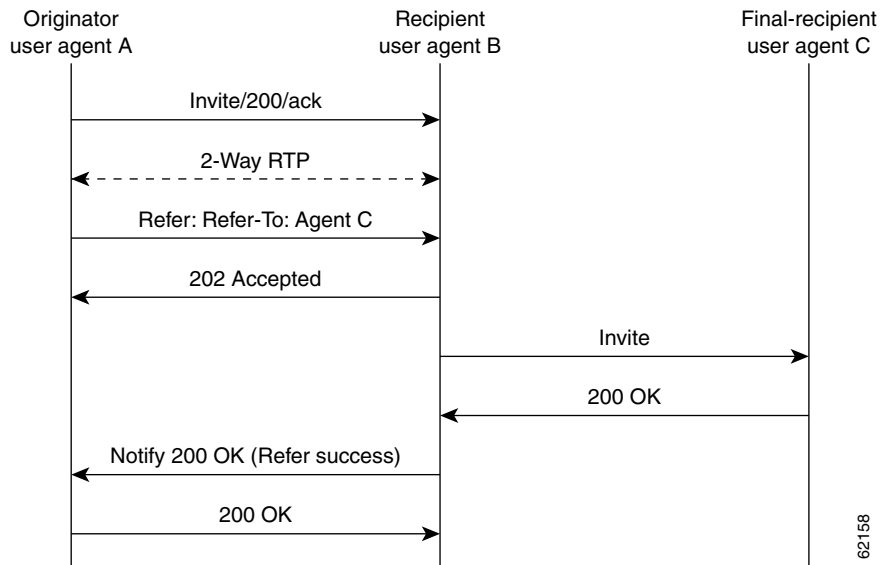
Note

A gateway can be a recipient or final-recipient; but not an originator.

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.

Figure 1 shows the call flow of a successful Refer transaction initiated within the context of an existing call.

Figure 1 Successful Refer Transaction



Refer-To Header

The recipient receives from the originator a Refer request that always contains a single Refer-to header. The Refer-to header includes a SIP URL that indicates the party to invite and must be in SIP URL format.

**Note**

The TEL URL format cannot be used in a Refer-to header, because it does not provide a host portion, and without one, the triggered Invite request cannot be routed.

The Refer-To header may contain three additional overloaded headers to form the triggered Invite request. If any of these three headers are present, they are included in the triggered Invite request. The three headers are:

- **Accept-Contact**—Optional in a Refer request. A SIP IOS gateway that receives an Invite request with an Accept-Contact does not act upon this header. This header is defined in draft-ietf-sip-callerprefs-03.txt and may be used by user agents that support caller preferences.
- **Proxy-Authorization**—A nonstandard header that SIP gateways do not act on. It is echoed in the triggered Invite request because proxies occasionally require it for billing purposes.
- **Replaces**—The Replaces header is used by SIP gateways to indicate whether the originator of the Refer request is requesting a blind or attended transfer. It is required if the originator is performing an attended transfer, and not required for a blind transfer.

All other headers present in the Refer-To are ignored, and are not sent in the triggered invite.

**Note**

The Refer-To and Contact headers are required in the Refer request. The absence of these headers results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Refer-To header. Multiple Refer-To headers result in a 4xx class response.

Referred-By Header

The Referred-By header is required in a Refer request. It identifies the originator and may also contain a signature (included for security purposes). SIP gateways echo the contents of the Referred-By header in the triggered Invite request, but on receiving an Invite request with this header, gateways do not act on it.

**Note**

The Referred-By header is required in a Refer request. The absence of this header results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Referred-By header. Multiple Referred-By headers result in a 4xx class response.

Notify Method

Once 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.

The Notify message must:

- Reflect the same To, From, and Call-ID headers that were received in the Refer request.
- Contain an Event header refer.
- Contain a message body with a SIP response line. For example: SIP/2.0 200 OK to report a successful Refer transaction, or SIP/2.0 503 Service Unavailable to report a failure. To report that the recipient disconnected before the transfer finished, it must use SIP/2.0 487 Request Canceled.

Two Cisco IOS commands pertain to the Notify method.

- The **timers notify** command sets the amount of time that the recipient should wait before retransmitting a Notify message to the originator.
- The **retry notify** command configures the number of times a Notify message is retransmitted to the originator.



Note For information on these commands, see the *Cisco IOS Voice Command Reference* (http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html).

Types of SIP Call Transfer Using the Refer Method

This section discusses how the Refer method facilitates call transfer.

There are two types of call transfer: blind and attended. The primary difference between the two is that the Replaces header is used in attended call transfers. The Replaces header is interpreted by the final-recipient and contains a Call-ID header, indicating that the initial call leg is to be replaced with the incoming Invite request.

As outlined in the Refer method, there are 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.

A gateway can be a recipient or final-recipient but not an originator.

Blind Call-Transfer Process

A blind, or unattended, transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. This is different from a consultative, or attended, transfer in which one of the transferring parties either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. Blind transfers are often preferred by automated devices that do not have the capability to make consultation calls.

The basic process of blind transfers works as described in the “Refer Method” section on page 5. In blind transfer, the originator (user agent that initiates the transfer or Refer request) sets up a call with the recipient (user agent that receives the Refer request). After the originator issues a Refer request to the recipient, the recipient, triggered by the Refer request, sends an Invite request to the final-recipient (user agent introduced into a call with the recipient). The recipient returns a SIP 202 (Accepted) response to the originator, and notifies the originator of the outcome of the Refer transaction—if the final-recipient was successfully (SIP 200 OK) or unsuccessfully (SIP 503 Service Unavailable) contacted.

If successful, a call is established between the recipient and the final-recipient. The original signaling relationship between the originator and recipient is terminated when a Bye request is sent by one of the parties. On a successful transfer, if the originator does not send a Bye request after receiving an acknowledgement for the Notify message, the recipient initiates a Bye request. [Figure 2](#) shows a successful blind or unattended call transfer in which the originator initiates a Bye request to terminate signaling with the recipient.

Figure 2 Successful Blind or Unattended Transfer—Originator Initiating a Bye Request

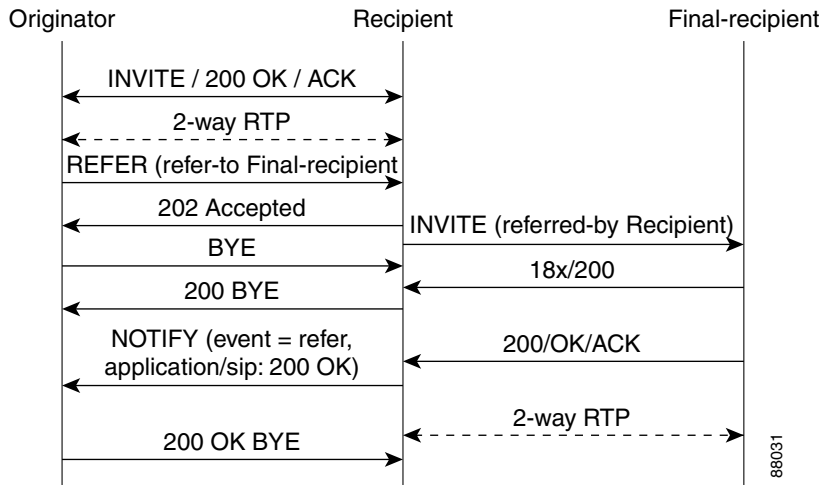
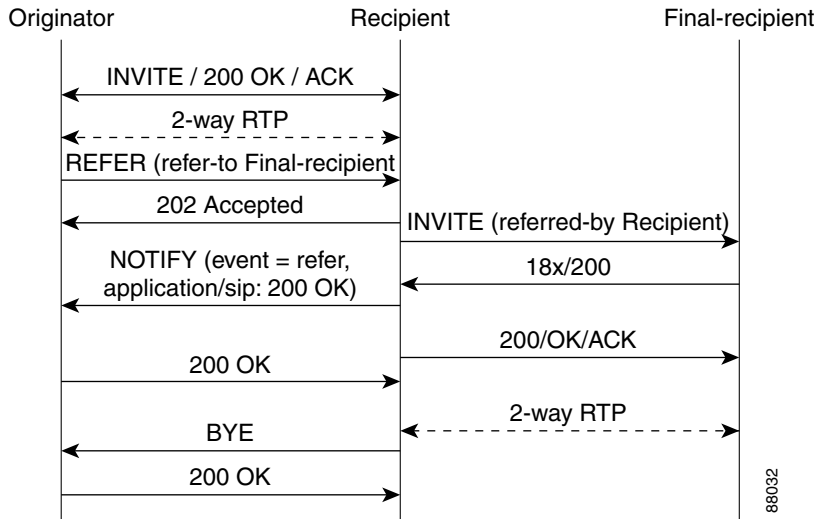


Figure 3 shows a successful blind or unattended call transfer in which the recipient initiates a Bye request to terminate signaling with the originator. A Notify message is always sent by the recipient to the originator after the final outcome of the call is known.

Figure 3 Successful Blind or Unattended Transfer –Recipient Initiating a Bye Request

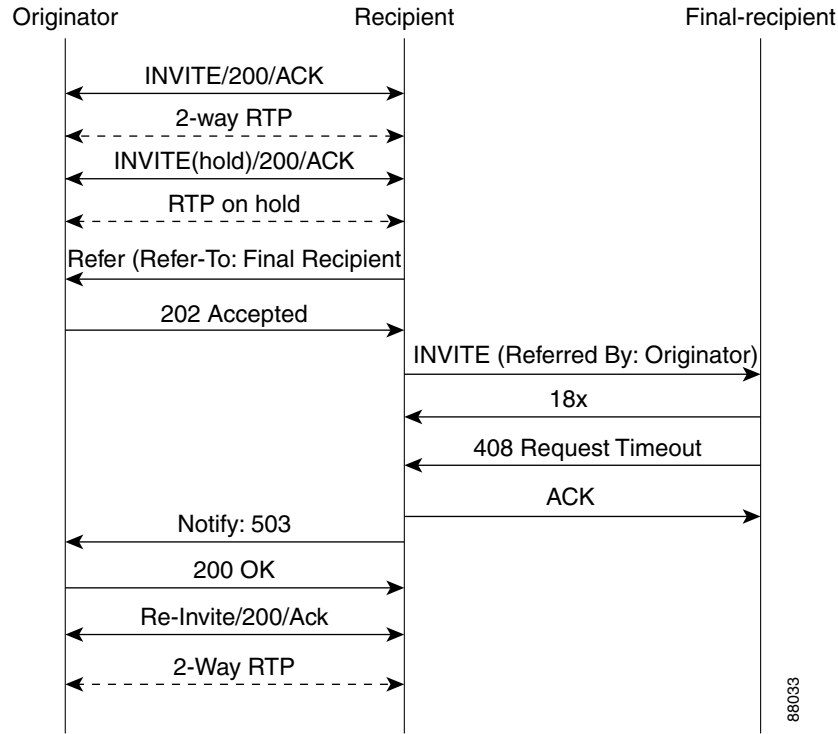


If a failure occurs with the triggered Invite to the final-recipient, the call between the originator and the recipient is not disconnected. The originator sends a re-Invite which takes the call off hold and returns to the original call with the recipient. With prior blind transfer functionality, if the recipient receives an 18x informational response from the final-recipient and then the call fails, the originator can not recover the call with the recipient.

A failure can be caused by an error condition or timeout.

Figure 4 shows that the call leg between the originator and the recipient remains active. Thus, if the Invite to the final-recipient fails (408 Request Timeout), the recipient notifies the originator of the failure with a Notify message. The originator sends a re-Invite and returns to the original call with the recipient.

Figure 4 Failed Blind Transfer—Originator Returns to Original Call with Recipient



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Attended Transfer

In attended transfers, the Replaces header is inserted by the initiator of the Refer request as an overloaded header in the Refer-To and is copied into the triggered Invite request sent to the final-recipient. The header has no affect on the recipient, but is interpreted by the final-recipient as a way to distinguish between blind transfer and attended transfer. The attended transfer process is described in [Table 1](#).

Table 1 Attended Transfer Process

Process	Description or Detail
1. Originator sets up a call with the recipient.	After the call is set up, originator places recipient on hold.
2. Originator establishes a call to the final-recipient.	—
3. Originator sends recipient a Refer request with an overloaded Replaces header in the Refer-To header.	—
4. Upon receipt of the Refer request, recipient sends a triggered Invite request to the final-recipient.	The Invite request received by final-recipient includes the Replaces header, identifying the call leg between the originator and final-recipient.
5. Recipient returns a SIP 202 (Accepted) response to the originator.	The SIP 202 (Accepted) acknowledges that the Invite has been sent.

Table 1 **Attended Transfer Process (continued)**

6. Final-recipient establishes a direct signaling relationship with recipient.	Receipt of the Replaces header is what indicates that the initial call leg is to be shut down and replaced by the incoming Invite request.
7. Recipient notifies originator of the outcome of the Refer transaction.	Recipient notifies the originator if the final-recipient was successfully or unsuccessfully contacted.
8. Recipient terminates the session with originator by sending a Bye request.	—

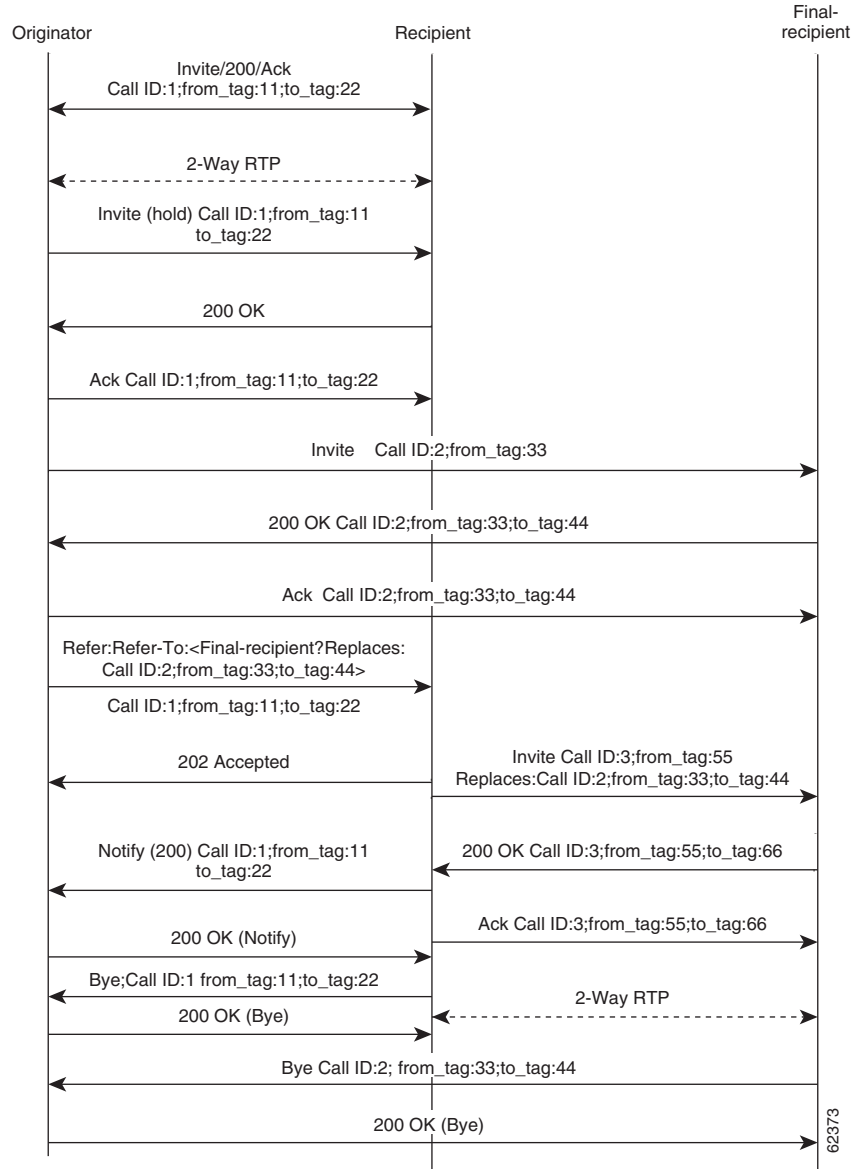
Replaces Header

The Replaces header is required in attended transfers. It indicates to the final-recipient that the initial call leg (identified by the Call-ID header and tags) is to be shut down and replaced by the incoming Invite request. The final-recipient sends a Bye request to the originator to terminate its session.

If the information provided by the Replaces header does not match an existing call leg, or if the information provided by the Replaces header matches a call leg but the call leg is not active (a Connect, 200 OK to the Invite request has not been sent by the final-recipient), the triggered Invite does not replace the initial call leg and the triggered Invite request is processed normally.

Any failure resulting from the triggered Invite request from the recipient to final-recipient does not destroy the call between the originator and the final-recipient. In these scenarios, all calls that are active (originator to recipient and originator to final-recipient) remain active after the failed attended transfer attempt. [Figure 5](#) shows a call flow for a successful attended transfer.

Figure 5 Successful Attended Transfer



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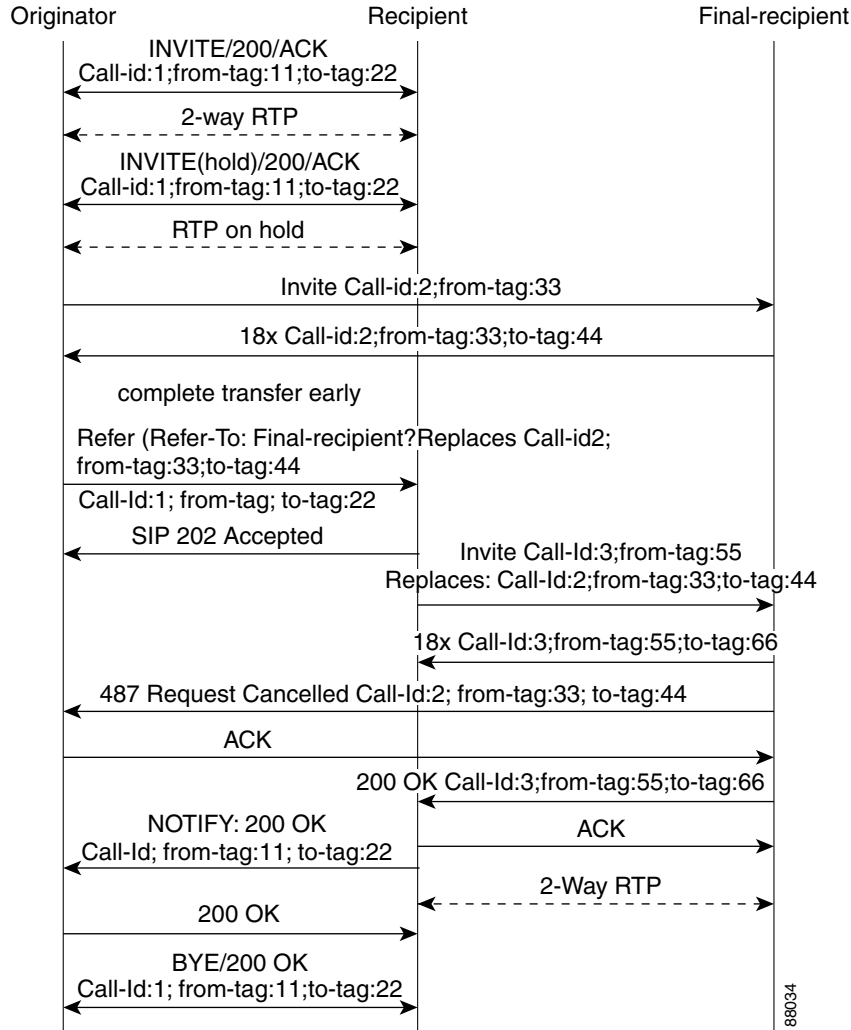
Attended Transfer with Early Completion

Attended transfers allow the originator to have a call established between both the recipient and the final-recipient. With attended transfer with early completion, the call between the originator and the final-recipient does not have to be active, or in the talking state, before the originator can transfer it to the recipient. The originator establishes a call with the recipient and only needs to be in the process of setting up a call with the final-recipient. The final-recipient may be ringing, but has not answered the call from the originator when it receives a re-Invite to replace the call with the originator and the recipient. [Figure 6](#) shows the process of attended transfer with early completion, and the detailed actions involved are described in [Table 2](#).

Table 2 *Attended Transfer with Early Completion Process*

Process	Description or Detail
1. Originator sets up a call with recipient.	After the call is set up, originator places recipient on hold.
2. Originator contacts final-recipient.	—
3. When originator gets an indication that final-recipient is ringing, it sends recipient a Refer request with an overloaded Replaces header in the Refer-to header.	The Replaces header is required in attended transfers and distinguishes between blind transfer and attended transfers.
4. Recipient returns a SIP 202 (Accepted) response to originator.	The SIP 202 (Accepted) acknowledges that the Invite has been sent.
5. Upon receipt of the Refer request, recipient sends a triggered Invite request to final-recipient.	The Invite request received by final-recipient includes the Replaces header, which indicates that the initial call leg (identified by the Call-ID header and tags) is to be shut down and replaced by the incoming Invite request.
6. Final-recipient establishes a direct signaling relationship with recipient.	Final-recipient tries to match the Call-ID header and the To or From tags in the Replaces header of the incoming Invite with an active call leg in its call control block. If a matching active call leg is found, final-recipient replies with exactly the same status as the found call leg. However it then terminates the found call leg with a 487 Request Cancelled response.
Note	If early transfer is attempted and the call involves quality of service (QoS) or Resource Reservation Protocol (RSVP), the triggered Invite from the recipient with the Replaces header is not processed and the transfer fails. The session between originator and final-recipient remains unchanged.
7. Recipient notifies originator of the outcome of the Refer transaction—that is, whether final-recipient was successfully or unsuccessfully contacted.	—
8. Recipient or originator terminates the session by sending a Bye request.	—

Figure 6 *Attended Transfer with Early Completion*



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VSA for Call Transfer

You can use a vendor-specific attribute (VSA) for SIP call transfer.

Referred-By Header

For consistency with existing billing models, the Referred-By and Requested-By headers are populated in call history tables as a VSA. Cisco VSAs are used for VoIP call authorization. The new VSA tag **supp-svc-xfer-by** helps to associate the call-legs for Call Detail Records (CDR) generation. The call-legs could be originator to recipient or recipient to final-recipient.

The new VSA tag **supp-svc-xfer-by** contains the user@host portion of the SIP URL of the Referred-By header for transfers performed with the Refer method. For transfers performed with the Bye/Also method, the tag contains the user@host portion of the SIP URL of the Requested-By header. For each call on the gateway, there are two RADIUS records that are generated: start and stop. The **supp-svc-xfer-by** VSA is only generated for stop records and is only generated on the recipient gateway—the gateway receiving the Refer or Bye/Also message.

The VSA is generated when a gateway that acts as a recipient receives a Refer or Bye/Also message with the Referred-By or Requested-By headers. There are usually two pairs of start and stop records. There is a start and stop record between the recipient and the originator and also between the recipient to final-recipient. In the latter case, the VSA is generated between the recipient to final-recipient only.

Business Group Field

A new business group VSA field has also been added that assists service providers with billing. The field allows service providers to add a proprietary header to call records. The VSA tag for business group ID is **cust-biz-grp-id** and is only generated for stop records. It is generated when the gateway receives an initial Invite with a vendor dial-plan header to be used in call records. In cases when the gateway acts as a recipient, the VSA is populated in the stop records between the recipient and originator and the recipient final-recipient.



Note

For more information about VSAs and CDRs, see the *CDR Accounting for Cisco IOS Voice Gateways* guide at <http://www.cisco.com/en/US/docs/ios/voice/cdr/developer/guide/cdrdev.html>.

SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications

Information about SIP Call Transfer and Call Forwarding with a Tcl IVR or VoiceXML (VMXL) script is provided in the following subsections:

- [SIP Call Transfer and Call Forwarding with a Tcl IVR Script, page 14](#)
- [Release Link Trunking on SIP Gateways, page 15](#)
- [SIP Gateway Initiation of Call Transfers, page 17](#)
- [SIP Call Forwarding, page 19](#)

SIP Call Transfer and Call Forwarding with a Tcl IVR Script

When using a Tcl IVR 2.0 application, you can implement SIP support of blind or attended call-transfer and call-forwarding requests from a Cisco IOS gateway. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. An attended transfer is one that is consultative—one of the transferring parties either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller. Blind transfers are often preferred by automated devices that do not have the capability to make consultative calls.

Before implementing blind transfer and call forwarding, you must write a custom Tcl IVR 2.0 script that implements call transfer and call forwarding. The script is responsible for receiving the hookflash event, providing dial tone, matching against the dial plan, initiating call transfer, and reestablishing the original call if the transfer attempt fails.



Note

For information on writing a Tcl IVR script, see the *Tcl IVR API Version 2.0 Programmer's Guide* (<http://www.cisco.com/en/US/docs/ios/voice/tcl/developer/guide/tclivr2.html>).

When the Tcl IVR script runs on the Cisco gateway, it can respond to requests to initiate blind call transfer (transfer without consultation) on a SIP call leg. SIP call forwarding on ephones (IP phones that are not configured on the gateway) is also supported.



Note

SIP Call Transfer and Call Forwarding is compliant with Voice Extensible Markup Language (VXML). VXML scripts can also be used to implement call transfer and call forwarding.

Release Link Trunking on SIP Gateways

RLT functionality has been added to Cisco IOS SIP gateways. With RLT functionality, SIP call transfer can now be triggered by CAS trunk signaling, which the custom Tcl IVR application can monitor. After a SIP call transfer has transpired and the CAS interface is no longer required, the CAS interface can be released.

The RLT functionality can be used to initiate blind transfers on SIP gateways. Blind call transfer uses the Refer method. A full description of blind transfer and the refer Method can be found in *Call Transfer Capabilities Using the Refer Method* documentation (http://www.cisco.com/en/US/docs/ios/12_2t/12_2t11/feature/guide/ftrefer.html).

RLT and SIP Call Transfers

Call transfer can be triggered by CAS trunk signaling and then captured by the custom Tcl IVR script on a gateway. The process begins with the originator (the SIP user agent that initiates the transfer or Refer request) responding with a dial tone once the originator receives the signal or hookflash from the PSTN call leg. The originator then prepares to receive dual-tone multifrequency (DTMF) digits that identify the final-recipient (the user agent introduced into a call with the recipient).

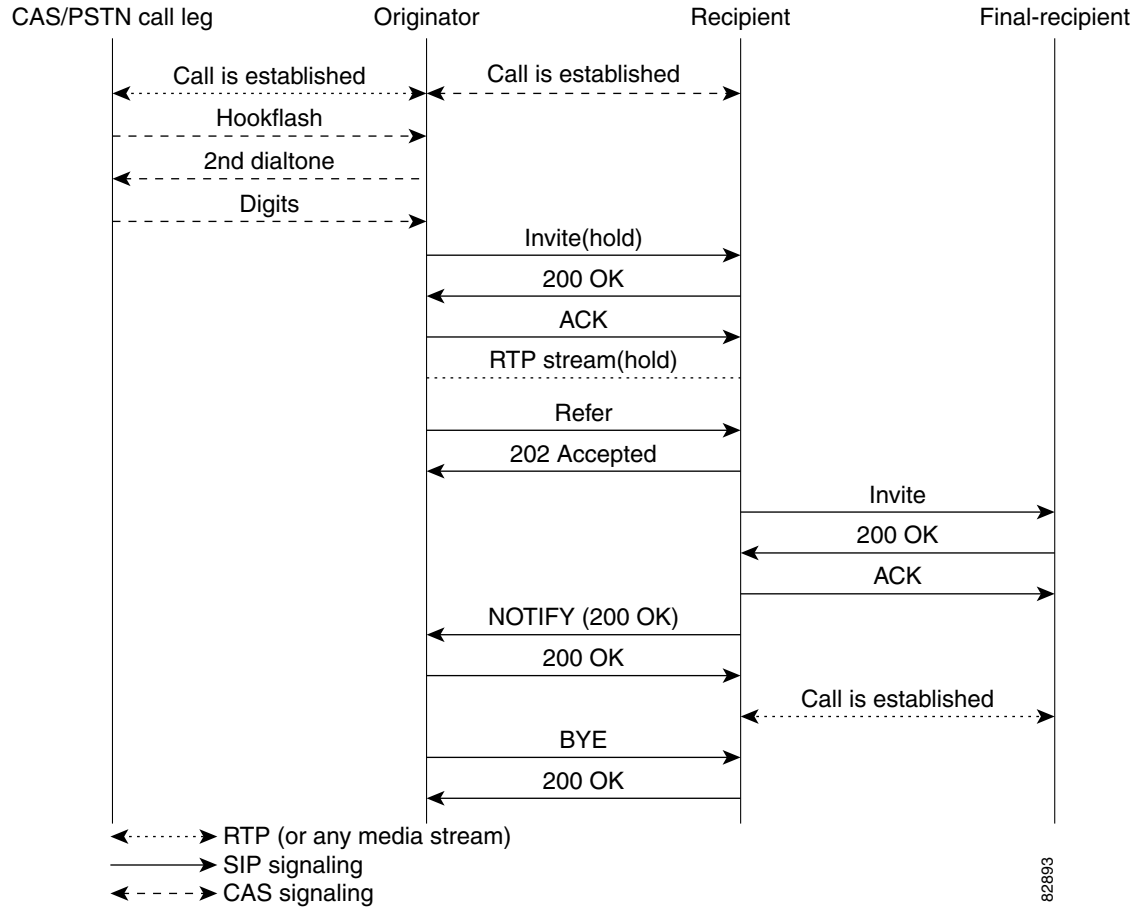
Once the first DTMF digit is received, the dial tone is discontinued. DTMF-digit collection is not completed until a 4-second interdigit timeout occurs, or an on-hook is received on that specific CAS time slot. Call transfer starts when DTMF-digit collection is successful. If digit collection fails, for example if not enough DTMF digits or invalid digits are collected, the initial call is reestablished.

Once the DTMF digits are successfully collected, the custom Tcl IVR script can initiate call transfer. SIP messaging begins when the transfer is initiated with the Refer method. The originator sends an Invite to the recipient (the user agent that receives the Refer request and is transferred to the final-recipient) to hold the call and request that the recipient not return Real-Time Transport Protocol (RTP) packets to the originator. The originator then sends a SIP Refer request to the recipient to start the transfer process. When the recipient receives the request, the recipient returns a 202 *Accepted* acknowledgement to the originator. The Tcl IVR script run by the originator can then release the CAS trunk and close the primary call (see [Figure 7 on page 16](#)).

If the recipient does not support the Refer method, a 501 *Not implemented* message is returned. However, for backward compatibility purposes, the call transfer is automatically continued with the Bye/Also method. The originator sends a Bye/Also request to the recipient and releases the CAS trunk with the PSTN call leg. The primary call between the originator and the recipient is closed when a 200 OK response is received.

In all other cases of call-transfer failures, the primary call between the originator and the recipient is immediately shut down.

Figure 7 Call Transfer Using the Refer Method



SIP and TEL URLs in Call Transfers

When the SIP call-transfer originator collects DTMF digits from the CAS trunk, it attempts to find a dial peer. If a dial peer is found, the session target in the dial peer is used to formulate a Session Initiation Protocol Uniform Resource Locator (SIP URL). This URL can be used with both the Refer method and the Bye/Also method. A SIP URL is in the following form:

```
sip:JohnSmith@example.com
```

If a valid dial peer is not found, a Telephone Uniform Resource Locator (TEL URL) is formulated in the Refer-To header. A TEL URL is in the following form:

```
tel:+11231234567
```

The choice of which URL to use is critical when correctly routing SIP calls. For example, the originating gateway can send out a Bye with an Also header, but the Also header can carry only a SIP URL. The Also header cannot carry a TEL URL. That is, if the gateway decides to send a Bye/Also but cannot find a matched dial peer, the gateway reports an error on the transfer gateway and sends a Bye without the Also header.

If the recipient of a SIP call transfer is a SIP phone, the phone must have the capability to interpret either the Refer method or the Bye/Also method for the call transfer to work. If the recipient is a Cisco IOS gateway, there needs to be a matching dial peer for the Refer-To *user*. *User*, looking at the previous example, can be either *JohnSmith* or *11231234567*. The dial peer also needs to have an application

session defined, where session can be the name of a Tcl IVR application. If there's no match, a 4xx error is sent back and no transfer occurs. If there's a POTS dial peer match, a call is made to that POTS phone. Before the 12.2(15)T release, if there's a VoIP match, the Refer-To URL is used to initiate a SIP call. In release 12.2(15)T and later releases, the application session target in the dial peer is used for the SIP call.

**Note**

For information on the application session target, see the [“Configure SIP Call Transfer and Call Forwarding on a POTS Dial Peer”](#) section on page 29.

SIP Gateway Initiation of Call Transfers

SIP gateways can also initiate, or originate, attended call transfers. The process begins when the originator establishes a call with the recipient. When the user on the PSTN call leg wants to transfer the call, the user uses hookflash to get a second dial tone and then enters the final-recipient's number. The Tcl IVR script can then put the original call on hold and set up the call to the final-recipient, making the originator active with the final-recipient. The Refer request is sent out when the user hangs up to transfer the call. The Refer request contains a Replaces header that contains three tags: *SIP CallID*, *from*, and *to*. The tags are passed along in the Invite from the recipient to the final-recipient, giving the final-recipient adequate information to replace the call leg. The host portion of the Refer request is built from the established initial call. The following is an example of a Refer request that contains a Replaces header:

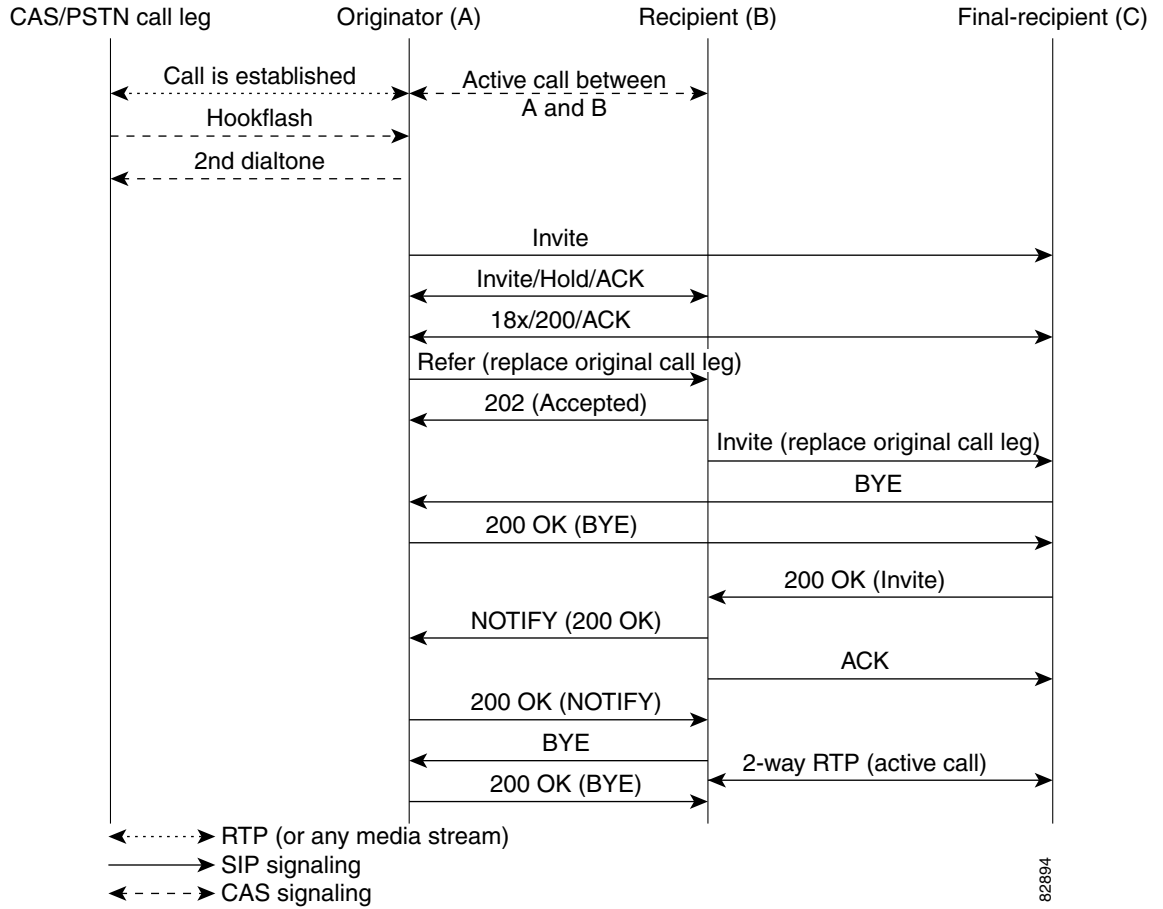
**Note**

IP addresses and hostnames in examples are fictitious.

```
Refer sip:3100801@172.16.190.100:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.16.190.99:5060
From: "5550100" <sip:5550100@172.16.190.187>
To: <sip:3100801@172.16.190.187>;tag=A7C2C-1E8C
Date: Sat, 01 Jan 2000 05:15:06 GMT
Call-ID: c2943000-106ae5-1c5f-3428@172.16.197.182
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 946685709
CSeq: 103 Refer
Refer-To:
sip:3100802@10.102.17.217?Replaces=DD713380-339C11CC-80BCF308-92BA812C@172.16.195.77;to-tag=A5438-23E4;from-tag=C9122EDB-2408
Referred-By: <sip:3100802@172.16.190.99>
Content-Length: 0
```

After the NOTIFY is received by the originator, the Tcl IVR script can disconnect the call between the originator and the recipient. The call between the originator and the final-recipient is disconnected by the recipient sending a BYE to the originator. [Figure 8](#) shows a call flow of a successful call transfer.

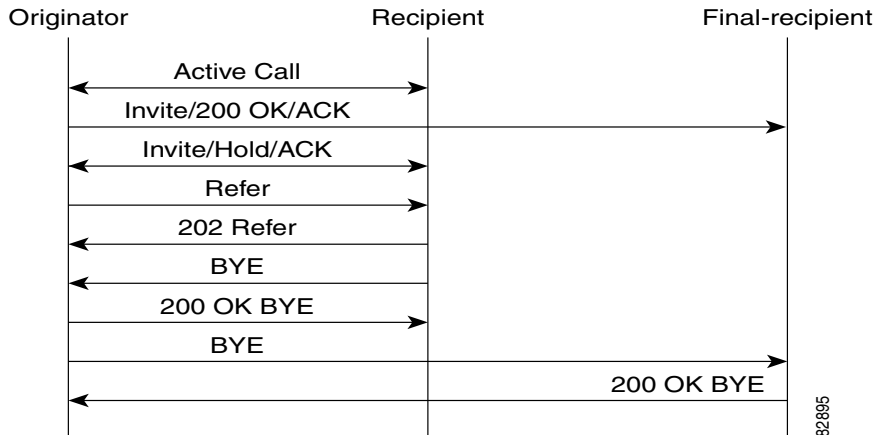
Figure 8 Successful Attended Call Transfer Initiated by the Originator



If the recipient does not support the Refer method, a 501 *Not implemented* message is returned.

In all other cases of call-transfer failures, the primary call between the originator and the recipient is immediately shut down. Figure 9 shows the recipient hanging up the call before the transfer completes. The item to notice is that the NOTIFY message is never sent.

Figure 9 Unsuccessful Call Transfer—Recipient Hangs Up Before Transfer Completes



SIP Call Forwarding

SIP call forwarding is supported only on ephones—IP phones that are not configured on the gateway. FXS, FXO, T1, E1, and CAS phones are not supported.

With ephones, there are four different types of SIP call forwarding supported:

- Call Forward Unavailable
- Call Forward No Answer
- Call Forward Busy
- Call Forward Unconditional

In all four of these call forwarding types, a 302 *Moved Temporarily* response is sent to the user agent client. A Diversion header included in the 302 response indicates the type of forward.

The 302 response also includes a Contact header, which is generated by the calling number that is provided by the custom Tcl IVR script. The 302 response also includes the host portion found in the dial peer for that calling number. If the calling number cannot match a VoIP dial-peer or POTS dial-peer number, a 503 *Service Unavailable* message is sent, except in the case of the Call Forward No Answer. With Call Forward No Answer, call forwarding is ignored, the phone rings, and the expires timer clears the call if there is no answer.



Note By default, SIP credentials for forwarded calls on Cisco IOS voice gateways are based on the calling number. To globally enable a gateway to use the redirecting number, instead, use the **authenticate redirecting-number** command. To configure this behavior for a specific dial peer on a gateway, use the **voice-class sip authenticate redirecting-number** command. For detailed information, see these commands in the *Cisco IOS Voice Command Reference* (http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html).



Note

In Cisco IOS Release 12.2(15)T and later releases, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

How to Configure SIP Call-Transfer Features

This section contains the following procedures:

- [Configuring SIP Call Transfer Using the Refer Method, page 20](#)
- [Configuring SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications, page 26](#)
- [Verifying SIP Call Transfer, page 36](#)
- [Troubleshooting Tips, page 39](#)



Note

- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP Call Transfer” section on page 2](#)
 - [“Restrictions for SIP Call Transfer” section on page 3](#)
- For help with a procedure, see the verification and troubleshooting sections listed above.

Configuring SIP Call Transfer Using the Refer Method

This section contains the following procedures:

- [Configure SIP Call Transfer on a POTS Dial Peer, page 20](#)
- [Configure SIP Call Transfer on a VoIP Dial Peer, page 21](#)
- [Configure the SIP Call-Transfer Session Target, page 23](#) (optional)
- [Configure SIP Refer and Notify Message Settings, page 25](#)

Configure SIP Call Transfer on a POTS Dial Peer

To configure SIP call transfer on a POTS dial peer, perform the following steps.



Note

To handle all call-transfer situations, configure both POTS and VoIP dial peers. This task configures SIP call transfer for a POTS dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice pots**
4. **application**
5. **destination-pattern**
6. **port**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag pots Example: Router(config)# dial-peer voice 25 pots	Enters dial-peer configuration mode for the specified POTS dial peer.

	Command or Action	Purpose
Step 4	<p>application <i>application-name</i></p> <p>Example: Router(config-dial-peer)# application session</p>	<p>Enables a specific application on a dial peer. The argument is as follows:</p> <ul style="list-style-type: none"> <i>application-name</i>—Name of the predefined application that you wish to enable on the dial peer. For SIP, the default Tcl application (from the Cisco IOS image) is session and can be applied to both VoIP and POTS dial peers.
Step 5	<p>destination-pattern [+]<i>string</i>[T]</p> <p>Example: Router(config-dial-peer)# destination-pattern 7777</p>	<p>Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer: Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> +—(Optional) Character indicating an E.164 standard number. <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. T—(Optional) Control character indicating that the destination-pattern value is a variable length dial string.
Step 6	<p>port <i>slot/port</i></p> <p>Example: Router(config-dial-peer)# port 1/1</p>	<p>Associates a dial peer with a voice slot number and a specific local voice port through which incoming VoIP calls are received.</p> <p>Note To find the correct port argument for your router, see the Cisco IOS Voice Command Reference.</p>
Step 7	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configure SIP Call Transfer on a VoIP Dial Peer

To configure SIP call transfer on a VoIP dial peer, perform the following steps.



Note

To handle all call-transfer situations, configure both POTS and VoIP dial peers. This task configures SIP call transfer for a VoIP dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **application**
5. **destination-pattern**
6. **session target**

7. `session protocol sipv2`
8. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>enable</code></p> <p>Example: Router> enable</p>	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	<p><code>configure terminal</code></p> <p>Example: Router# configure terminal</p>	Enters global configuration mode.
Step 3	<p><code>dial-peer voice tag voip</code></p> <p>Example: Router(config)# dial-peer voice 29 voip</p>	Enters dial-peer configuration mode for the specified dial peer.
Step 4	<p><code>application application-name</code></p> <p>Example: Router(config-dial-peer)# application session</p>	<p>Enables a specific application on a dial peer. The argument is as follows:</p> <ul style="list-style-type: none"> • <i>application-name</i>—Name of the predefined application that you wish to enable on the dial peer. For SIP, the default Tcl application (from the Cisco IOS image) is session and can be applied to both VoIP and POTS dial peers.
Step 5	<p><code>destination-pattern [+]string[T]</code></p> <p>Example: Router(config-dial-peer)# destination-pattern 7777</p>	<p>Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • +—(Optional) Character that indicates an E.164 standard number. • <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • T—(Optional) Control character indicating that the destination-pattern value is a variable-length dial string.
Step 6	<p><code>session target ipv4:destination-address</code></p> <p>Example: Router(config-dial-peer)# session target ipv4:10.10.1.3</p>	<p>Specifies a network-specific address for a dial peer. Keyword and argument are as follows:</p> <ul style="list-style-type: none"> • ipv4:destination address—IP address of the dial peer, in this format: <i>xxx.xxx.xxx.xxx</i>

	Command or Action	Purpose
Step 7	<code>session protocol sipv2</code> Example: Router(config-dial-peer)# <code>session protocol sipv2</code>	Configures the VoIP dial peer to use IETF SIP.
Step 8	<code>exit</code> Example: Router(config-dial-peer)# <code>exit</code>	Exits the current mode.

Configure the SIP Call-Transfer Session Target

To configure the SIP call-transfer session target, perform the following steps.



Note

This task configures a SIP server as a session target. Although it is not required, configuring a SIP server as a session target is useful if there is a Cisco SIP proxy server (CSPS) present in the network. With a CSPS, you can configure the SIP server option and have the interested dial peers use the CSPS by default.

To determine the call-transfer destination on the originator, check if there is a matching dial peer:

- If yes, check the session target for the dial peer. If the session target is a SIP server, configure the SIP server as described in the task below. If the session target is not a SIP server, the session target configured in the VoIP dial peer is used.
- If no, a TEL URL is sent.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `sip-ua`
4. `sip-server`
5. `exit`
6. `dial-peer voice voip`
7. `destination-pattern`
8. `session target sip-server`
9. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	sip-server dns: host-name Example: Router(config-sip-ua)# sip-server dns:example.sip.com	Sets the global SIP server interface to a Domain Name System (DNS) hostname. If you do not specify a hostname, the default DNS defined by the ip name-server command is used.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.
Step 6	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the specified dial peer.
Step 7	destination-pattern [+]string[T] Example: Router(config-dial-peer)# destination-pattern 7777	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. Keywords and arguments are as follows: <ul style="list-style-type: none"> • +—(Optional) Character that indicates an E.164 standard number. • <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • T—(Optional) Control character indicating that the destination-pattern value is a variable-length dial string.

	Command or Action	Purpose
Step 8	<pre>session target sip-server</pre> <p>Example: Router(config-dial-peer)# session target sip-server</p>	Instructs the dial-peer session target to use the global SIP server. Doing so saves you from having to repeatedly enter the SIP server interface address for each dial peer.
Step 9	<pre>exit</pre> <p>Example: Router(config-dial-peer)# exit</p>	Exits the current mode.

Configure SIP Refer and Notify Message Settings

To configure SIP Refer and Notify message settings, perform the following steps.



Note

The Refer request is initiated by the originating gateway and signals the start of call transfer. Once the outcome of the SIP Refer transaction is known, the recipient of the Refer request notifies the originating gateway of the outcome of the Refer transaction—whether the final-recipient was successfully or unsuccessfully contacted. Notification is accomplished using the Notify method.

Prerequisites

- Configure dial peers for correct functioning of the Refer method.



Note

For dial-peer configuration steps, see the [“Configure SIP Call Transfer on a POTS Dial Peer” section on page 20](#).

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `sip-ua`
4. `timers notify`
5. `retry notify`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers notify <i>milliseconds</i> Example: Router(config-sip-ua)# timers notify 500	Sets the amount time that the user agent waits before retransmitting the Notify message. The argument is as follows: <ul style="list-style-type: none"> <i>milliseconds</i>—Time, in ms. Range: 100 to 1000. Default: 500.
Step 5	retry notify <i>number</i> Example: Router(config-sip-ua)# retry notify 10	Sets the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or refer request. The argument is as follows: <ul style="list-style-type: none"> <i>number</i>—Number of notify message retries. Range: 1 to 10.
Step 6	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications

To configure SIP call transfer and call forwarding using Tcl IVR 2.0 and VoiceXML applications, perform the following steps.

- [Load the Tcl IVR Application on the Gateway, page 27](#) (required)
- [Configure SIP Call Transfer and Call Forwarding on a POTS Dial Peer, page 29](#) (required)
- [Configure SIP Call Transfer and Call Forwarding on a VoIP Dial Peer, page 30](#) (required)
- [Configure the SIP Call-Transfer and Call-Forwarding Session Target, page 32](#) (optional)
- [Configure SIP Refer and Notify Message Settings, page 34](#) (required)

Load the Tcl IVR Application on the Gateway

Prerequisites

- Before you implement SIP support of blind or attended call-transfer and call-forwarding requests from a Cisco IOS gateway, you must load a custom Tcl IVR 2.0 or VXML script on the gateway. Write a Tcl IVR 2.0 script that implements Cisco IOS call-transfer and call-forwarding services. The Tcl IVR script is responsible for receiving the hookflash event, providing dial tone, matching against the dial plan, initiating the call transfer, and reestablishing the original call if the transfer attempt fails.



Note For information on writing a Tcl IVR script, see the *Tcl IVR API Version 2.0 Programmer's Guide* (<http://www.cisco.com/en/US/docs/ios/voice/tcl/developer/guide/tclivr2.html>).

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call application voice** *application-name location*
4. **call application voice** *application-name language number language*
5. **call application voice** *application-name set-location language category location*
6. **exit**
7. **call application voice load**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	call application voice <i>application-name location</i> Example: Router(config)# call application voice transfer_app flash:app_h450_transfer.tcl	Loads the Tcl IVR script and specifies its application name. Arguments are as follows: <ul style="list-style-type: none"> • <i>application-name</i>—Name used to reference the call application. This is a user-defined name and does not have to match the document name. • <i>location</i>—Location of the Tcl IVR file in URL format. For example, flash memory (flash:filename), TFTP (tftp://./filename) or HTTP server paths (http://./filename) are valid locations.

	Command or Action	Purpose
Step 4	<p>call application voice <i>application-name</i> language <i>number language</i></p> <p>Example: Router(config)# call application voice transfer_app language 1 en</p>	<p>(Optional) Sets the language for dynamic prompts used by the application. Arguments are as follows:</p> <ul style="list-style-type: none"> • <i>application-name</i>—Name of the Tcl IVR application to which the language parameters pass. • <i>number</i>—Number that identifies the language used by the audio files for the IVR application. • <i>language</i>—Language of the associated audio file. Valid values are as follows: <ul style="list-style-type: none"> – en—English – sp—Spanish – ch—Mandarin – aa—All
Step 5	<p>call application voice <i>application-name</i> set-location <i>language category location</i></p> <p>Example: Router(config)# call application voice transfer_app set-location en 0 flash:/prompts</p>	<p>(Optional) Defines the location and category of the audio files that are used by the application for dynamic prompts. Arguments are as follows:</p> <ul style="list-style-type: none"> • <i>application-name</i>—Name of the Tcl IVR application. • <i>language</i>—Language of the associated audio file. Valid values are as above. • <i>category</i>—Category group (0 to 4) for the audio files from this location. For example, audio files for the days and months could be category 1, audio files for units of currency could be category 2, and audio files for units of time (seconds, minutes, and hours) could be category 3. Range is from 0 to 4. The value 0 means all categories. • <i>location</i>—URL of the directory that contains the language audio files used by the application, without filenames. For example, flash memory (flash) or a directory on a server (TFTP, HTTP, or RTSP) are valid locations.
Step 6	<p>exit</p> <p>Example: Router(config)# exit</p>	Exits the current mode.
Step 7	<p>all application voice load <i>application-name</i></p> <p>Example: Router# call application voice load transfer.app</p>	<p>(Optional) Reloads the Tcl script after it has been modified. The argument is as follows:</p> <ul style="list-style-type: none"> • <i>application-name</i>—Name of the Tcl IVR application to reload.

Configure SIP Call Transfer and Call Forwarding on a POTS Dial Peer

To configure SIP call transfer and call forwarding on a POTS dial peer, perform the following steps.



Note

- To handle all call-transfer and call-forwarding situations, configure both POTS and VoIP dial peers. This task configures SIP call transfer and call forwarding for a POTS dial peer.
- To configure SIP call transfer and forwarding on a Cisco IOS gateway by using the CAS trunk, see the *Cisco IOS Dial Technologies Configuration Guide*.



Note

To locate a release-specific configuration guide for your Cisco IOS software release, select the **Cisco IOS and NX-OS Software** category at the following Product Support page and navigate accordingly: <http://www.cisco.com/web/psa/products/index.html>.

Restrictions

In Cisco IOS Release 12.2(15)T, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice pots**
4. **application**
5. **destination-pattern**
6. **port**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag pots Example: Router(config)# dial-peer voice 25 pots	Enters dial-peer configuration mode and for the specified POTS dial peer.

	Command or Action	Purpose
Step 4	<p>application <i>application-name</i></p> <p>Example: Router(config-dial-peer)# application transfer_app</p>	<p>Enables a specific application on a dial peer. The argument is as follows:</p> <ul style="list-style-type: none"> <i>application-name</i>—Name of the predefined application that you wish to enable on the dial peer. In this case, you need to load the Tcl IVR script specified in the “Load the Tcl IVR Application on the Gateway” section on page 27.
Step 5	<p>destination-pattern [+]<i>string</i>[T]</p> <p>Example: Router(config-dial-peer)# destination-pattern 7777</p>	<p>Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> +—(Optional) Character that indicates an E.164 standard number. <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. T—(Optional) Control character indicating that the destination-pattern value is a variable-length dial string.
Step 6	<p>port <i>slot/port</i></p> <p>Example: Router(config-dial-peer)# port 1/1</p>	<p>Associates a dial peer with a voice slot number and a specific local voice port through which incoming VoIP calls are received.</p> <p>Note To find the correct port argument for your router, see the <i>Cisco IOS Voice Command Reference</i>.</p>
Step 7	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configure SIP Call Transfer and Call Forwarding on a VoIP Dial Peer

To configure SIP call transfer and call forwarding on a VoIP dial peer, perform the following steps.



Note

- To handle all call-transfer and call-forwarding situations, configure both POTS and VoIP dial peers. This task configures SIP call transfer and call forwarding for a VoIP dial peer.
- To configure SIP call transfer and forwarding on a Cisco IOS gateway by using the CAS trunk, see the *Cisco IOS Dial Technologies Configuration Guide*.



Note

To locate a release-specific configuration guide for your Cisco IOS software release, select the **Cisco IOS and NX-OS Software** category at the following Product Support page and navigate accordingly: <http://www.cisco.com/web/psa/products/index.html>.

Restrictions

- RLT on CAS or analog (FXS) ports is necessary for initiating IP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- In Cisco IOS Release 12.2(15)T, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **application**
5. **destination-pattern**
6. **session target**
7. **session protocol sipv2**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the specified dial peer.
Step 4	application application-name Example: Router(config-dial-peer)# application transfer_app	Enables a specific application on a dial peer. The argument is as follows: <ul style="list-style-type: none"> • <i>application-name</i>—Name of the predefined application you wish to enable on the dial peer. Loads the Tcl IVR script specified in the “Load the Tcl IVR Application on the Gateway” section on page 27.

	Command or Action	Purpose
Step 5	<p>destination-pattern [+]<i>string</i>[T]</p> <p>Example: Router(config-dial-peer)# destination-pattern 7777</p>	<p>Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • +—(Optional) Character that indicates an E.164 standard number. • <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • T—(Optional) Control character indicating that the destination-pattern value is a variable-length dial string.
Step 6	<p>session target ipv4:<i>destination-address</i></p> <p>Example: Router(config-dial-peer)# session target ipv4:10.10.1.3</p>	<p>Specifies a network-specific address for a dial peer. The argument is as follows:</p> <ul style="list-style-type: none"> • <i>destination address</i>—IP address of the dial peer, in this format: <i>xxx.xxx.xxx.xxx</i>
Step 7	<p>session protocol sipv2</p> <p>Example: Router(config-dial-peer)# session protocol sipv2</p>	<p>Configures the VoIP dial peer to use IETF SIP. The keyword is as follows:</p> <ul style="list-style-type: none"> • sipv2—Causes the VoIP dial peer to use IETF SIP. Use this keyword with the SIP option.
Step 8	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configure the SIP Call-Transfer and Call-Forwarding Session Target

To configure the SIP call-transfer and call-forwarding session target, perform these steps.



Note

- To configure a SIP server as a session target, follow this task. Although configuring a SIP server as a session target is not required, it is useful if there is a Cisco SIP proxy server (CSPS) present in the network. With a CSPS, you can configure the SIP server option and have the interested dial peers use the CSPS by default.
- To determine the call-transfer destination on the originator, check if there is a matching dial peer:
 - If yes, check the session target for the dial peer. If the session target is a SIP server, configure the SIP server as described in the task below. If the session target is not a SIP server, the session target configured in the VoIP dial peer is used.
 - If no, a TEL URL is sent.
- To configure SIP call transfer and forwarding on a Cisco IOS gateway by using the CAS trunk, see the *Cisco IOS Dial Technologies Configuration Guide*.



Note

To locate a release-specific configuration guide for your Cisco IOS software release, select the **Cisco IOS and NX-OS Software** category at the following Product Support page and navigate accordingly: <http://www.cisco.com/web/psa/products/index.html>.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **sip-server**
5. **exit**
6. **dial-peer voice voip**
7. **destination-pattern**
8. **session target sip-server**
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	sip-server dns:host-name Example: Router(config-sip-ua)# sip-server dns:example.sip.com	Sets the global SIP server interface to a DNS hostname. The argument is as follows: <ul style="list-style-type: none"> • <i>host-name</i>—DNS hostname. If you do not specify a hostname, the default DNS defined by the ip name-server command is used.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.
Step 6	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the specified dial peer.

	Command or Action	Purpose
Step 7	<p>destination-pattern [+]<i>string</i>[T]</p> <p>Example: Router(config-dial-peer)# destination-pattern 7777</p>	<p>Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • +—(Optional) Character that indicates an E.164 standard number. • <i>string</i>—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • T—(Optional) Control character indicating that the destination-pattern value is a variable-length dial string.
Step 8	<p>session target sip-server</p> <p>Example: Router(config-dial-peer)# session target sip-server</p>	<p>Instruct the dial-peer session target to use the global SIP server. Doing so saves you from having to repeatedly enter the SIP server interface address for each dial peer.</p>
Step 9	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configure SIP Refer and Notify Message Settings

To configure SIP Refer and Notify message settings, perform the following steps.



Note

The Refer request is initiated by the originating gateway and signals the start of call transfer. Once the outcome of the SIP Refer transaction is known, the recipient of the Refer request notifies the originating gateway of the outcome of the Refer transaction—whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the Notify method.

Prerequisites

- Custom scripting is necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.
- Configure the dial peers for correct functioning of the Refer method.



Note

Dial-peer configuration steps are in the [“Configure SIP Call Transfer and Call Forwarding on a POTS Dial Peer”](#) section on page 29.

Restrictions

- Only RLT on CAS or analog (FXS) ports is supported with SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.

- SIP call forwarding is supported only on ephones—IP phones that are not configured on the gateway. FXS and CAS phones are not supported.
- In Cisco IOS Release 12.2(15)T, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

**Note**

Custom scripting is necessary for ephones to initiate call forwarding. The standard configurations listed in this document work only when an ephone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers refer**
5. **retry refer**
6. **timers notify**
7. **retry notify**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers refer <i>milliseconds</i> Example: Router(config-sip-ua)# timers refer 500	Sets the amount time that the user agent waits before retransmitting the Refer request. The argument is as follows: <ul style="list-style-type: none"> • <i>milliseconds</i>—Time, in ms. Range: 100 to 1000. Default: 500.

	Command or Action	Purpose
Step 5	<pre>retry refer number</pre> <p>Example: Router(config-sip-ua)# retry refer 10</p>	<p>Sets the number of times that the Refer request is retransmitted to the user agent that initiated the transfer or refer request. The argument is as follows:</p> <ul style="list-style-type: none"> <i>number</i>—Number of Notify message retries. Range: 1 to 10. Default: 10.
Step 6	<pre>timers notify milliseconds</pre> <p>Example: Router(config-sip-ua)# timers notify 500</p>	<p>Sets the amount time that the user agent waits before retransmitting the Notify message. The argument is as follows:</p> <ul style="list-style-type: none"> <i>milliseconds</i>—Time, in ms. Range: 100 to 1000. Default: 500.
Step 7	<pre>retry notify number</pre> <p>Example: Router(config-sip-ua)# retry notify 10</p>	<p>Sets the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or refer request. The argument is as follows:</p> <ul style="list-style-type: none"> <i>number</i>—Number of notify message retries. Range: 1 to 10.
Step 8	<pre>exit</pre> <p>Example: Router(config-sip-ua)# exit</p>	<p>Exits the current mode.</p>

Verifying SIP Call Transfer

To verify SIP configurations, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. `show dial-peer voice`
2. `show ephone`
3. `show ephone-dn`
4. `show running-configuration`
5. `show sip-ua retry`
6. `show sip-ua statistics`
7. `show sip-ua timers`
8. `show telephony-service ephone-dn`
9. `show telephony-service voice-port`
10. `show voice port`

DETAILED STEPS

Step 1 **show dial-peer voice**

Use this command to display configuration information about voice dial peers. Use with the **summary** keyword to display a summary of all voice dial peers.

Step 2 **show ephone**

Use this command to display IP-phone output. Use with the **summary** keyword to display a summary of all IP phones.

Step 3 **show ephone-dn**

Use this command to display the IP-phone destination number. Use with the **summary** keyword to display a summary of all IP-phone destination numbers.

Step 4 **show running-configuration**

Use this command to verify your configuration.

Step 5 **show sip-ua retry**

Use this command to display SIP retry statistics including Notify responses.

```
Router# show sip-ua retry
```

```
SIP UA Retry Values
invite retry count = 6 response retry count = 1
bye retry count = 1 cancel retry count = 1
prack retry count = 10 comet retry count = 10
reliable 1xx count = 6 notify retry count = 10
```

Step 6 **show sip-ua statistics**

Use this command to display response, traffic, and retry statistics for the SIP user agent.

The following applies to the example below.

Field	Meaning
OkNotify1/0	Successful response to the Notify request.
202Accepted 0/1	Successful response to the Refer request.
Notify 0/1	Status.
Refer 1/0	Status.
Notify 0/1	No Notify requests were received from the gateway. One request was sent.
Refer 1/0	One request was received. No requests were sent.
Notify 0 under Retry Statistics	The Notify request was not retransmitted.

```
Router# show sip-ua statistics
```

```
SIP Response Statistics (Inbound/Outbound)
Informational:
  Trying 0/0, Ringing 0/0,
  Forwarded 0/0, Queued 0/0,
  SessionProgress 0/0
Success:
  OkInvite 0/0, OkBye 0/0,
  OkCancel 0/0, OkOptions 0/0,
  OkPrack 0/0, OkPreconditionMet 0/0,
```

```

    OKSubscribe 0/0, OkNotify 0/0,
    202Accepted 0/0
  Redirection (Inbound only):
    MultipleChoice 0, MovedPermanently 0,
    MovedTemporarily 0, UseProxy 0,
    AlternateService 0
  Client Error:
    BadRequest 0/0, Unauthorized 0/0,
    PaymentRequired 0/0, Forbidden 0/0,
    NotFound 0/0, MethodNotAllowed 0/0,
    NotAcceptable 0/0, ProxyAuthReqd 0/0,
    ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
  ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
  UnsupportedMediaType 0/0, BadExtension 0/0,
  TempNotAvailable 0/0, CallLegNonExistent 0/0,
  LoopDetected 0/0, TooManyHops 0/0,
  AddrIncomplete 0/0, Ambiguous 0/0,
  BusyHere 0/0, RequestCancel 0/0
  NotAcceptableMedia 0/0, BadEvent 0/0
  Server Error:
    InternalError 0/0, NotImplemented 0/0,
    BadGateway 0/0, ServiceUnavail 0/0,
    GatewayTimeout 0/0, BadSipVer 0/0,
    PreCondFailure 0/0
  Global Failure:
    BusyEverywhere 0/0, Decline 0/0,
    NotExistAnywhere 0/0, NotAcceptable 0/0
  Miscellaneous counters:
    RedirectResponseMappedToClientError 1,
  SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0,
    Subscribe 0/0, Notify 0/0,
    Refer 0/0

  Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliablelxx 0, Notify 0

  SDP application statistics:
    Parses: 0, Builds 0
    Invalid token order: 0, Invalid param: 0
    Not SDP desc: 0, No resource: 0

```



Tip To reset counters for the **show sip-ua statistics** command, use the **clear sip-ua statistics** command.

Step 7 show sip-ua timers

Use this command to display the current settings for SIP user-agent timers, including Notify responses.

```

Router# show sip-ua timers

SIP UA Timer Values (milliseconds)
trying 500, expires 150000, connect 500, disconnect 500
comet 500, prack 500, rellxx 500, notify 500

```

Step 8 show telephony-service ephone-dn

Use this command to display the Cisco-IP-phone destination number of the Cisco IOS telephony-service router.

Step 9 show telephony-service voice-port

Use this command to display the virtual voice-port configuration.

Step 10 show voice port

Use this command to display configuration information about a specific voice port.

Troubleshooting Tips

For general troubleshooting tips and a list of important **debug** commands, see the “General Troubleshooting Tips” section of the “Basic SIP Configuration” document (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-basic_cfg.html).

Configuration Examples for SIP Call-Transfer Features

This section provides the following configuration examples:

- [SIP Call Transfer Using the Refer Method: Examples, page 39](#)
- [SIP Call Transfer and Call Forwarding Using Tel IVR 2.0 and VoiceXML Applications: Examples, page 40](#)

SIP Call Transfer Using the Refer Method: Examples

**Note**

Note that the **application session** command is set on all involved gateway dial peers. You must set the correct Cisco IOS session for call transfer.

```
Router# show running-config

Building configuration...

Current configuration : 4192 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
voice-card 2
!
ip subnet-zero
!
controller T1 2/0
framing esf
linecode b8zs
ds0-group 0 timeslots 1-24 type e&m-wink-start
!
```

```

interface FastEthernet3/0
ip address 172.18.200.36 255.255.255.0
speed 10
half-duplex
no shut
ip rsvp bandwidth 7500 7500
!
voice-port 2/0:0
timing hookflash-in 1500
!
dial-peer voice 3660110 voip
application session
incoming called-number 3660110
destination-pattern 3660110
session protocol sipv2
session target ipv4:172.18.200.24
codec g711ulaw
!
dial-peer voice 3640110 pots
application session
destination-pattern 3640110
direct-inward-dial
port 2/0:0
!
sip-ua
retry bye 1
retry refer 3
timers notify 400
timers refer 567
no oli
sip-server ipv4:172.18.200.21
!
line con 0
line aux 0
line vty 0 4
login
!
end

```

SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications: Examples

This section provides an end-to-end call-transfer configuration example.

- [Blind Call Transfer, page 40](#)
- [Originating Gateway, page 41](#)
- [Recipient Gateway, page 42](#)
- [Final-Recipient, page 43](#)

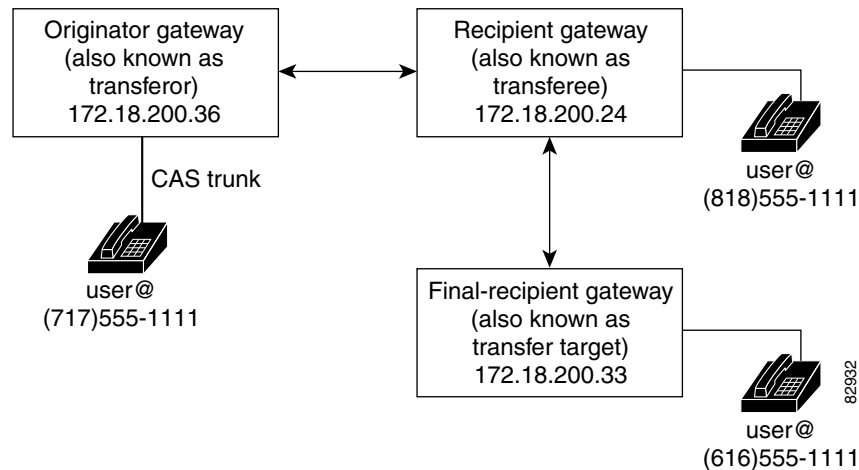


Note

IP addresses and hostnames in examples are fictitious.

Blind Call Transfer

[Figure 10](#) shows the relationship of the gateways in the blind call transfer.

Figure 10 **Blind Call Transfer**

The following scenario is an example of a blind call transfer.

1. User at (818) 555-0111 calls user at (717) 555-0111, and they are in a conversation.
2. User at (717) 555-0111 decides to transfer user at (818) 555-0111 to user at (616) 555-0111.
Transfer takes place by the user at (717) 555-0111 going on-hook over the CAS trunk and dialing (616) 555-0111.
3. Call transfer is initiated from the originating gateway to the recipient gateway, and the originator releases the CAS trunk to (717) 555-0111.
4. Recipient gateway releases the call leg to the originator and initiates a new call to the final-recipient—user at (616) 555-0111.
5. Call transfer is complete, and user at (818) 555-0111 and user at (616) 555-0111 are in a conversation.

Originating Gateway

The following example shows a configuration of the originating gateway.

```
Router# show running-config

Building configuration...

Current configuration : 4192 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
voice-card 2
!
ip subnet-zero
!
controller T1 2/0
framing esf
linecode b8zs
```

```

ds0-group 0 timeslots 1-24 type e&m-wink-start
!
interface FastEthernet3/0
ip address 172.18.200.36 255.255.255.0
speed 10
half-duplex
no shut
ip rsvp bandwidth 7500 7500
!
voice-port 2/0:0
timing hookflash-in 1500
!
call application voice sample_RLT tftp://sample_RLT.tcl
call application voice sample_RLT uid-len 4
call application voice sample_RLT language 1 en
call application voice sample_RLT set-location en 0 tftp://prompts/en/
!
dial-peer voice 2 voip
application sample_rlt
destination-pattern 8183821111
session protocol sipv2
session target ipv4:172.18.200.24
codec g711ulaw
!
dial-peer voice 3 pots
destination-pattern 7173721111
direct-inward-dial
port 2/0:0
prefix 7173721111
!
dial-peer voice 3621111 voip
application sample_rlt
destination-pattern 6163621111
session protocol sipv2
session target sip-server
codec g711ulaw
!
sip-ua
retry bye 1
retry refer 3
timers notify 400
timers refer 567
no oli
sip-server ipv4:172.18.200.21
!
line con 0
line aux 0
line vty 0 4
login
!
end

```

Recipient Gateway

The following example shows a configuration of the recipient gateway.

```

Router# show running-config

Building configuration...

Current configuration : 2791 bytes
!
version 12.2
service config

```

```

no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut
speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8183821111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7173721111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
dial-peer voice 4 voip
application session
destination-pattern 6163621111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
!
line con 0
line aux 0
line vty 0 4
login
!
end

```

Final-Recipient

The following example shows a configuration of the final-recipient gateway.

```

Router# show running-config

!
version 12.2
no parser cache
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
no logging buffered
!

```

```
clock timezone GMT 0
aaa new-model
!
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
!
ip domain name example.com
ip dhcp smart-relay
!
!
voice class codec 1
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 5 g726r16
codec preference 6 g726r24
codec preference 7 g726r32
codec preference 8 g723ar53
codec preference 9 g723ar63
codec preference 10 g729r8
!
interface Ethernet0/0
ip address 172.18.200.33 255.255.255.0
no shut
half-duplex
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
dial-peer voice 1 pots
application session
destination-pattern 6163621111
port 1/1/1
!
ip classless
no ip http server
ip pim bidir-enable
!
gateway
!
sip-ua
!
rtr responder
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
password password1
line vty 5 15
!
end
```

Additional References

General SIP References

- “SIP Features Roadmap” (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-roadmap.html)
—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- “Overview of SIP” (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-overview.html)
—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter (listed alphabetically)

- *Call Transfer Capabilities Using the Refer Method* at http://www.cisco.com/en/US/docs/ios/12_2t/12_2t11/feature/guide/ftrefer.html.
- *Cisco IOS Dial Technologies Configuration Guide*.



Note To locate a release-specific configuration guide for your Cisco IOS software release, select the **Cisco IOS and NX-OS Software** category at the following Product Support page and navigate accordingly: <http://www.cisco.com/web/psa/products/index.html>.

- *Cisco IOS Voice Command Reference* at http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html.
- *Enhancements to the Session Initiation Protocol for VoIP on Cisco Access Platforms* at http://www.cisco.com/en/US/docs/ios/12_2t/12_2t11/feature/guide/ftsipgv1.html.
- *CDR Accounting for Cisco IOS Voice Gateways* guide at <http://www.cisco.com/en/US/docs/ios/voice/cdr/developer/guide/cdrdev.html>.
- *Tcl IVR API Version 2.0 Programmer's Guide* at <http://www.cisco.com/en/US/docs/ios/voice/tcl/developer/guide/tclivr2.html>.

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Configuring SIP Message, Timer, and Response Features

First Published: March 1992

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This chapter describes how to configure Session Initiation Protocol (SIP) message components, session timers, and responses. It describes the following features:

- [Internal Cause Code Consistency Between SIP and H.323, page 6](#)
- [SIP - Configurable PSTN Cause Code Mapping, page 8](#)
- [SIP: Accept-Language Header Support, page 11](#)
- [SIP Enhanced 180 Provisional Response Handling, page 13](#)
- [SIP Extensions for Caller Identity and Privacy, page 14](#)
- [SIP INVITE Request with Malformed Via Header, page 27](#)
- [SIP Session Timer Support, page 27](#)
- [SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion, page 29](#)
- [SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers, page 32](#)
- [SIP Stack Portability, page 41](#)
- [SIP: Domain Name Support in SIP Headers, page 63](#)
- [SIP Gateway Support for SDP Session Information and Permit Hostname CLI, page 66](#)
- [Outbound Proxy Support for the SIP Gateway, page 68](#)
- [SIP: SIP Support for PAI, page 68](#)
- [SIP: History-info Header Support, page 68](#)

History for the Internal Cause Code Consistency Between SIP and H.323 Feature

Release	Modification
12.2(11)T	These features were introduced.

History for the SIP - Configurable PSTN Cause Code Mapping Feature

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB2	This feature was implemented on an additional platform.



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12.2(8)T	This feature was integrated into this release.
12.2(11)T	This feature was implemented on additional platforms.

History for the SIP Accept-Language Header Support Feature

Release	Modification
12.3(1)	The SIP Accept-Language Header Support feature was introduced.

History for the SIP Enhanced 180 Provisional Response Handling Feature

Release	Modification
12.2(13)T	The features were introduced.

History for the SIP Extensions for Caller Identity and Privacy Feature

Release	Modification
12.2(13)T	The features were introduced.

History for the SIP INVITE Request with Malformed Via Header Feature

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(8)T	This feature was integrated into this release.
12.2(11)T	Support was added for additional platforms.

History for the SIP Session Timer Support Feature

Release	Modification
12.2(11)T	These features were introduced.
12.4(9)T	This feature was updated to support RFC 4028.

History for the SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion Feature

Release	Modification
12.3(8)T	This feature was introduced.

History for the SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers Feature

Release	Modification
12.3(4)T	This feature was introduced.

History for the SIP Stack Portability Feature

Release	Modification
12.4(1)T	This feature was introduced.

History for the SIP: Domain Name Support in SIP Headers Feature

Release	Modification
12.4(2)T	This feature was introduced.

History for the SIP Gateway Support for SDP Session Information and Permit Hostname CLI Feature

Release	Modification
12.4(9)T	This feature was introduced.

History for the Outbound Proxy Support for the SIP Gateway

Release	Modification
12.4(15)T	This feature was introduced.
12.4(20)T	Support was added for disabling outbound proxy support for SIP on a per dial peer basis

History for the SIP Support for PAI

Release	Modification
12.4(15)T	This feature was introduced.

History for the SIP History-info Header Support Feature

Release	Modification
12.4(22)T	This feature was introduced.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the [“Information About SIP Message Components, Session Timers, and Response Features”](#) section on page 6.

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

Contents

- [Prerequisites for SIP Message, Timer, and Response Features, page 4](#)
- [Restrictions for SIP Message, Timer, and Response Features, page 4](#)
- [Information About SIP Message Components, Session Timers, and Response Features, page 6](#)
- [How to Configure SIP Message, Timer, and Response Features, page 70](#)
- [Configuration Examples for SIP Message, Timer, and Response Features, page 127](#)
- [Additional References, page 159](#)

Prerequisites for SIP Message, Timer, and Response Features

All SIP Message Components, Session Timers, and Responses Features

- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.
Refer to the following Cisco IOS IP Configuration Guides by navigating to them from the [Product Support](http://www.cisco.com/web/psa/products/index.html?c=268438303) page (<http://www.cisco.com/web/psa/products/index.html?c=268438303>) according to your Cisco IOS release:
 - *Cisco IOS IP Addressing Services Configuration Guide*
 - *Cisco IOS IP Mobility Configuration Guide*
 - *Cisco IOS IP Multicast Configuration Guide*
 - *Cisco IOS IP Routing Protocols Configuration Guide*
- Configure VoIP.
- Configure SIP voice functionality.

SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion Feature

- For the reason header, do the following:
 - Configure the CLI reason-header override, in SIP user-agent (SIP UA) configuration mode, if you want the Reason header to take precedence over existing cause-code-mapping tables on the gateway receiving Reason header.
- For buffered calling name completion (such as buffer-invite timers), do the following:
 - Complete the prerequisites associated with the Support for the ISDN Calling Name Display feature in release 12.3(4)T (refer to the “[Configuring SIP DTMF Features](#)” chapter).
 - Configure a buffer invite timer value.
- Ensure that the incoming ISDN setup contains a name-to-follow indication as described in [Generic Requirements for ISDN Calling Name Identification Services for Primary Rate Interface \(PRI\) specification, GR-1367](#).

Restrictions for SIP Message, Timer, and Response Features

All SIP Message Components, Session Timers, and Responses Features

- Via handling for TCP is not implemented.

SIP Permit Hostname Command Features

- The maximum length of a hostname is 30 characters; SIP INVITE message support will truncate any hostname over 30 characters.

SIP Accept-Language Header Support Feature

- The Accept-Language header provided by the inbound SIP call leg is passed to the outbound call leg only if that call leg is SIP as well.

SIP Extensions for Caller Identity and Privacy Feature

- This feature does not support the Anonymity header described in the Internet Engineering Task Force (IETF) specification, draft-ietf-privacy-.02.txt. The feature implements presentation level anonymity at Layer 5, rather than at the IP address level. Since the SIP gateway assumes that all adjacent signaling devices are trusted, it is recommended that border SIP proxy servers enforce anonymity policies at administrative boundaries.
- The IETF specification, draft-ietf-privacy-.02.txt, for mapping of North American Numbering Plan Area (NANPA) defined Automatic Number Identification Information Indicators (ANI II) or Originating Line Information (OLI) digits, is still under development. The current implementation of Cisco IOS VoiceXML supports carrying the ANI II digits as digits, rather than as a string representation of the numbering plan-tagged ANI II digits.

SIP INVITE Request with Malformed Via Header Feature

- Distributed Call Signaling (DCS) headers and extensions are not supported.

SIP Session Timer Support Feature

- This feature enables the SIP Portable stack and IOS gateway to comply with IETF RFC 4028 specification for SIP session timer.
- Cisco SIP gateways cannot initiate the use of SIP session timers but do fully support session timers if another user agent requests it.
- The Min-SE value can be set only by using the **min-se** command described in this document. It cannot be set using the CISCO-SIP-UA-MIB.

SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers Feature

- For outbound calls, an application is allowed to pass any extended or nonstandard header except for the following:
 - Call-ID
 - Supported
 - Require
 - Min-SE
 - Session-Expires
 - Max-Forwards
 - CSeq
 - The “Tag” parameter within From and To headers (From and To headers themselves are allowed)

All other headers may be overwritten by the application to create the header lines in the SIP INVITE message.

- SUBSCRIBE and NOTIFY methods are supported for Tool Command Language (Tcl) applications only.

SIP Gateway Support for SDP Session Information Feature

- The maximum length of a received session information line is 1000 characters; SIP gateway support truncates any session information line over 1000 characters.

SIP: SIP Support for PAI

- Privacy for REGISTER messages is not supported. When a gateway registers with another endpoint, the gateway assumes this endpoint is within the trusted domain, therefore privacy regarding this transaction is unnecessary.

SIP History-info Header Support Feature

- History-info header support is provided on Cisco IOS SIP time-division multiplexing (TDM) gateways and SIP-SIP Cisco Unified Border Elements only.
- Cisco IOS SIP gateways cannot use the information in the history-info header to make routing decisions.

Information About SIP Message Components, Session Timers, and Response Features

This section contains the following information:

- [Internal Cause Code Consistency Between SIP and H.323, page 6](#)
- [SIP - Configurable PSTN Cause Code Mapping, page 8](#)
- [SIP: Accept-Language Header Support, page 11](#)
- [SIP Enhanced 180 Provisional Response Handling, page 13](#)
- [SIP Extensions for Caller Identity and Privacy, page 14](#)
- [SIP INVITE Request with Malformed Via Header, page 27](#)
- [SIP Gateway Support for SDP Session Information and Permit Hostname Command, page 27](#)
- [SIP Session Timer Support, page 27](#)
- [SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion, page 29](#)
- [SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers, page 32](#)
- [SIP Stack Portability, page 41](#)
- [SIP: Domain Name Support in SIP Headers, page 63](#)
- [SIP Gateway Support for SDP Session Information and Permit Hostname CLI, page 66](#)
- [Outbound Proxy Support for the SIP Gateway, page 68](#)
- [SIP: SIP Support for PAI, page 68](#)
- [SIP: History-info Header Support, page 68](#)

Internal Cause Code Consistency Between SIP and H.323

The Internal Cause Code Consistency Between SIP and H.323 feature establishes a standard set of categories for internal causes of voice call failures. Before this feature, the cause code that was passed when an internal failure occurred was not standardized or based on any defined rules. The nonstandardization led to confusing or incorrect cause code information, and possibly contributed to billing errors.

This feature establishes a standard set of categories for internal causes of voice call failures. Internal cause-code consistency enables more efficient and effective operation of combined SIP and H.323 networks, which reduces operational expenses and improves service availability.

**Note**

RFC 2543-bis-04 enhancements obsolete the SIP cause codes 303 *Redirection: See Other* and 411 *Client Error: Length required*. For information on RFC 2543-bis-04 enhancements, refer to the “[Achieving SIP RFC Compliance](#)” chapter.

H.323 and SIP standard cause codes that are now generated accurately reflect the nature of each internal failure. This capability makes the H.323 and SIP call control protocols consistent with cause codes that are generated for common problems. Also, for each internal failure, an ITU-T Q.850 release cause code is also assigned and [Table 1](#) maps the new standard categories with the Q.850 release cause code and description that is assigned to each category.

Table 1 H.323 and SIP Standard Category and Q.850 Cause Code Mapping

Standard Category	Standard Category Description	Q.850 Cause Code	Q.850 Release Cause Description
Socket Failure	Typical scenarios: <ul style="list-style-type: none"> • Transmission Control Protocol (TCP) socket connection failure. • Problem sending an H.323 SETUP. • Problem sending a SIP INVITE. • Send or receive error occurs on connected socket. 	27	CC_CAUSE_DESTINATION_OUT_OF_ORDER The destination indicated by the user cannot be reached because the destination’s interface is not functioning correctly. The signaling message was unable to be delivered to the remote party.
Destination Address Resolution Failure	Typical scenarios: <ul style="list-style-type: none"> • Domain Name System (DNS) resolution failure. • Invalid session target in configuration. 	3	CC_CAUSE_NO_ROUTE The called party cannot be reached because the network that the call has been routed through does not serve the desired destination.
Call Setup Timeout Failure	Typical scenarios: <ul style="list-style-type: none"> • No H.323 call proceeding. • No H.323 alerting or connect message received from the terminating gateway. • Invite expires timer reached maximum number of retries allowed. 	102	CC_CAUSE_RECOVERY_ON_TIMER_EXPIRY A procedure has been initiated by the expiry of a timer in association with error handling procedures.
Internal Resource Allocation Failure	Typical scenarios: <ul style="list-style-type: none"> • Out of memory. • Internal access to the TCP socket becomes unavailable. 	47	CC_CAUSE_NO_RESOURCE A “resource unavailable” event has occurred.
Invalid Message Received Error	Typical scenarios: <ul style="list-style-type: none"> • An invalid message was received. 	95	CC_CAUSE_INVALID_MESSAGE An invalid message event has occurred.

Table 1 H.323 and SIP Standard Category and Q.850 Cause Code Mapping (continued)

Standard Category	Standard Category Description	Q.850 Cause Code	Q.850 Release Cause Description
Mandatory IE Missing Error	Typical scenarios: <ul style="list-style-type: none"> Mandatory Contact field missing in SIP message. Session Description Protocol (SDP) body is missing. 	96	CC_CAUSE_MANDATORY_IE_MISSING The equipment sending this cause has received a message that is missing an information element (IE). This information element must be present in the message before the message can be processed.
Invalid IE Contents Error	Typical scenarios: <ul style="list-style-type: none"> SIP Contact field is present, but format is bad. 	100	CC_CAUSE_INVALID_IE_CONTENTS The equipment sending this cause code has received an information element that it has implemented. However, the equipment sending this cause code has not implemented one or more of the specific fields.
Message in Invalid Call State	Typical scenarios: <ul style="list-style-type: none"> An unexpected message was received that is incompatible with the call state. 	101	CC_CAUSE_MESSAGE_IN_INCOMP_CALL_STATE Indicates that a message has been received that is incompatible with the call state.
Internal Error	Typical scenarios: <ul style="list-style-type: none"> Failed to send message to Public Switched Telephone Network (PSTN). 	127	CC_CAUSE_INTERWORKING There has been interworking with a network that does not provide causes for actions it takes. Precise cause cannot be ascertained.
QoS Error	Typical scenarios: <ul style="list-style-type: none"> Quality of service (QoS) error. 	49	CC_CAUSE_QOS_UNAVAILABLE The requested QoS cannot be provided.
Media Negotiation Failure	Typical scenarios: <ul style="list-style-type: none"> No codec match occurred. H.323 or H.245 problem leading to failure in media negotiation. 	65	CC_CAUSE_BEARER_CAPABILITY_NOT_IMPLEMENTED The equipment sending this cause does not support the bearer capability requested.

SIP - Configurable PSTN Cause Code Mapping

For calls to be established between a SIP network and a PSTN network, the two networks must be able to interoperate. One aspect of their interoperation is the mapping of PSTN cause codes, which indicate reasons for PSTN call failure or completion, to SIP status codes or events. The opposite is also true: SIP status codes or events are mapped to PSTN cause codes. Event mapping tables found in this document show the standard or default mappings between SIP and PSTN.

However, you may want to customize the SIP user-agent software to override the default mappings between the SIP and PSTN networks. The SIP - Configurable PSTN Cause Code Mapping feature allows you to configure specific map settings between the PSTN and SIP networks. Thus, any SIP status code can be mapped to any PSTN cause code, or vice versa.

When set, these settings can be stored in the NVRAM and are restored automatically on bootup.

Default Mappings

Table 2 lists PSTN cause codes and the corresponding SIP event mappings that are set by default. Any code other than the codes listed are mapped by default to *500 Internal server error*.

Table 2 **Default PSTN Cause Code to SIP Event Mappings**

PSTN Cause Code	Description	SIP Event
1	Unallocated number	404 Not found
2	No route to specified transit network	404 Not found
3	No route to destination	404 Not found
17	User busy	486 Busy here
18	No user response	480 Temporarily unavailable
19	No answer from the user	
20	Subscriber absent	
21	Call rejected	403 Forbidden
22	Number changed	410 Gone
26	Non-selected user clearing	404 Not found
27	Destination out of order	404 Not found
28	Address incomplete	484 Address incomplete
29	Facility rejected	501 Not implemented
31	Normal, unspecified	404 Not found
34	No circuit available	503 Service unavailable
38	Network out of order	503 Service unavailable
41	Temporary failure	503 Service unavailable
42	Switching equipment congestion	503 Service unavailable
47	Resource unavailable	503 Service unavailable
55	Incoming class barred within Closed User Group (CUG)	403 Forbidden
57	Bearer capability not authorized	403 Forbidden
58	Bearer capability not presently available	501 Not implemented
65	Bearer capability not implemented	501 Not implemented
79	Service or option not implemented	501 Not implemented
87	User not member of Closed User Group (CUG)	503 Service Unavailable
88	Incompatible destination	400 Bad request
95	Invalid message	400 Bad request
102	Recover on Expires timeout	408 Request timeout
111	Protocol error	400 Bad request
Any code other than those listed above:		500 Internal server error

Table 3 lists the SIP events and the corresponding PSTN cause codes mappings that are set by default.

Table 3 *Default SIP Event to PSTN Cause Code Mapping*

SIP Event	PSTN Cause Code	Description
400 Bad request	127	Interworking, unspecified
401 Unauthorized	57	Bearer capability not authorized
402 Payment required	21	Call rejected
403 Forbidden	57	Bearer capability not authorized
404 Not found	1	Unallocated number
405 Method not allowed	127	Interworking, unspecified
406 Not acceptable		
407 Proxy authentication required	21	Call rejected
408 Request timeout	102	Recover on Expires timeout
409 Conflict	41	Temporary failure
410 Gone	1	Unallocated number
411 Length required	127	Interworking, unspecified
413 Request entity too long		
414 Request URI (URL) too long		
415 Unsupported media type	79	Service or option not implemented
420 Bad extension	127	Interworking, unspecified
480 Temporarily unavailable	18	No user response
481 Call leg does not exist	127	Interworking, unspecified
482 Loop detected		
483 Too many hops		
484 Address incomplete	28	Address incomplete
485 Address ambiguous	1	Unallocated number
486 Busy here	17	User busy
487 Request cancelled	127	Interworking, unspecified
488 Not acceptable here	127	Interworking, unspecified
500 Internal server error	41	Temporary failure
501 Not implemented	79	Service or option not implemented
502 Bad gateway	38	Network out of order
503 Service unavailable	63	Service or option unavailable
504 Gateway timeout	102	Recover on Expires timeout
505 Version not implemented	127	Interworking, unspecified
580 Precondition Failed	47	Resource unavailable, unspecified
600 Busy everywhere	17	User busy
603 Decline	21	Call rejected

Table 3 **Default SIP Event to PSTN Cause Code Mapping (continued)**

SIP Event	PSTN Cause Code	Description
604 Does not exist anywhere	1	Unallocated number
606 Not acceptable	58	Bearer capability not presently available

Benefits of SIP - Configurable PSTN Cause Code Mapping

The feature offers control and flexibility. By using CLI commands, you can easily customize the default or standard mappings that are currently available between PSTN and SIP networks. This allows for flexibility when setting up deployment sites.

SIP: Accept-Language Header Support

The SIP Accept-Language Header Support feature introduces support for the Accept-Language header in SIP INVITE messages and in OPTIONS responses. This feature enables you to configure up to nine languages to be carried in SIP messages and to indicate multiple language preferences of first choice, second choice, and so on.

Feature benefits include the following:

- Allows service providers to support language-based features
- Allows VXML applications providers to support language-based services

To configure Accept-Language header support, you need to understand the following concepts:

- [Feature Design of SIP Accept-Language Header Support, page 11](#)
- [Sample INVITE Message, page 11](#)
- [Sample OPTIONS Response, page 12](#)

Feature Design of SIP Accept-Language Header Support

Cisco implements this feature on SIP trunking gateways by supporting a new header, Accept-Language, as defined in the Internet Engineering Task Force (IETF) specification, draft-ietf-sip-rfc2543bis-09, *SIP: Session Initiation Protocol*. The Accept-Language header is used in SIP INVITEs, which establish media sessions between user agents, and in SIP OPTIONS responses, which list user-agent capabilities. The header specifies language preferences for reason phrases, session descriptions, or status responses. A SIP proxy may also use the Accept-Language header to route to a human operator.

The Accept-Language header supports 139 languages, as specified in the International Organization for Standardization (ISO) specification, ISO 639: *Codes for Representation of Names of Languages*. The SIP Accept-Language Header Support feature allows you to configure up to nine languages to be carried in INVITE messages and OPTIONS responses. This feature also supports the qvalue (q=) parameter, which allows you to indicate multiple language preferences, that is, first choice, second choice, and so on.

Sample INVITE Message

The following is a sample outgoing INVITE message for a gateway configured to support the Sindhi, Zulu, and Lingala languages.

```
11:38:42: Sent:
```

```

INVITE sip:36602@172.18.193.120:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: <sip:172.18.193.98>;tag=27FB000-42E
To: <sip:36602@172.18.193.120>
Date: Mon, 01 Mar 1993 11:38:42 GMT
Call-ID: 23970D87-155011CC-8009E835-18264FDE@172.18.193.98
Supported: timer,100rel
Min-SE: 1800
Cisco-Guid: 0-0-0-0
User-Agent: Cisco-SIPGateway/IOS-12.x
Accept-Language: sd, zu, ln;q=0.123
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE, NOTIFY,
INFO
CSeq: 101 INVITE
Max-Forwards: 10
Remote-Party-ID: <sip:172.18.193.98>;party=calling;screen=no;privacy=off
Timestamp: 730985922
Contact: <sip:172.18.193.98:5060>
Expires: 300
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 322

v=0
o=CiscoSystemsSIP-GW-UserAgent 5606 9265 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 16434 RTP/AVP 18 100 101
c=IN IP4 172.18.193.98
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:10

```

Sample OPTIONS Response

The following is a sample OPTIONS response from a gateway configured to support the Yoruba, Sindhi, and English languages.

```

11:28:44: Received:
OPTIONS sip:36601@172.18.193.98:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.120:5060
From: "user" <sip:36602@172.18.193.120>
To: <sip:36601@172.18.193.98>
Date: Mon, 01 Mar 1993 02:55:01 GMT
Call-ID: BB8A5738-14EE11CC-8008B310-2C18B10E@172.18.193.120
Accept: application/sdp
CSeq: 110 OPTIONS
Contact: <sip:36601@172.18.193.98:5060>
Content-Length: 0

11:28:44: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.120:5060
From: "user" <sip:36602@172.18.193.120>
To: <sip:36601@172.18.193.98>;tag=2768F24-1DB2
Date: Mon, 01 Mar 1993 11:28:44 GMT
Call-ID: BB8A5738-14EE11CC-8008B310-2C18B10E@172.18.193.120

```

```

Server: Cisco-SIPGateway/IOS-12.x
Content-Type: application/sdp
CSeq: 110 OPTIONS
Supported: 100rel
Accept-Language: yo, sd;q=0.234, en;q=0.123
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE, NOTIFY,
INFO
Accept: application/sdp
Allow-Events: telephone-event
Content-Length: 170

v=0
o=CiscoSystemsSIP-GW-UserAgent 7292 5756 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 0 RTP/AVP 18 0 8 4 2 15 3
c=IN IP4 172.18.193.98

```

SIP Enhanced 180 Provisional Response Handling

The SIP Enhanced 180 Provisional Response Handling feature provides the ability to enable or disable early media cut-through on Cisco IOS gateways for SIP 180 response messages. The feature allows you to specify whether 180 messages with Session Description Protocol (SDP) are handled in the same way as 183 responses with SDP. The 180 Ringing message is a provisional or informational response used to indicate that the INVITE message has been received by the user agent and that alerting is taking place. The 183 Session Progress response indicates that information about the call state is present in the message body media information. Both 180 and 183 messages may contain SDP which allow an early media session to be established prior to the call being answered.

Prior to the implementation of this feature, Cisco gateways handled a 180 Ringing response with SDP in the same manner as a 183 Session Progress response; that is, the SDP was assumed to be an indication that the far end would send early media. Cisco gateways handled a 180 response without SDP by providing local ringback, rather than early media cut-through. This feature provides the capability to ignore the presence or absence of SDP in 180 messages, and as a result, treat all 180 messages in a uniform manner. The SIP Enhanced 180 Provisional Response Handling feature introduces the new **disable-early-media 180** command that enables you to specify which call treatment, early media or local ringback, is provided for 180 responses with SDP.

[Table 4](#) shows the call treatments available with this feature.

Table 4 *Call Treatments with SIP Enhanced 180 Provisional Response Handling Feature*

Response Message	Cisco IOS VoiceXML Status	Treatment
180 response with SDP	Enabled (default)	Early media cut-through
180 response with SDP	Disabled	Local ringback
180 response without SDP	Not affected by the new feature	Local ringback
183 response with SDP	Not affected by the new feature	Early media cut-through

SIP Extensions for Caller Identity and Privacy

To configure the SIP Extensions for Caller Identity and Privacy feature, you must understand the following concepts:

- [Privacy, Screening, and Presentation Indicators, page 14](#)
- [Remote-Party-ID Implementation, page 15](#)
- [Inbound and Outbound Call Flows, page 16](#)
- [Remote-Party-ID in SIP and PSTN Messages, page 24](#)

Privacy, Screening, and Presentation Indicators

Cisco implements this feature on SIP trunking gateways by supporting a header, Remote-Party-ID, as defined in the IETF specification, draft-ietf-privacy-.02.txt, *SIP Extensions for Caller Identity and Privacy*. The Remote-Part-ID header identifies the calling party and carries presentation and screening information. In previous SIP implementations, the From header was used to indicate calling party identity, and once defined in the initial INVITE request, could not be modified for that session. Implementing the Remote-Part-ID header, which can be modified, added, or removed as a call session is being established, overcomes previous limitations and enables call participant privacy indication, screening, and verification. The feature uses the Remote-Part-ID header to support translation capability between Integrated Services Digital Networks (ISDN) messages and Remote-Party-ID SIP tags. The SIP header also enables support for certain telephony services, and some regulatory and public safety requirements, by providing screening and presentation indicators.

The SIP Extensions for Caller Identity and Privacy feature introduces command-line interface (CLI) commands to enable remote-party-id translations and to configure alternative calling information treatments for calls entering the SIP trunking gateway. Configurable treatment options are:

- Calling name and number pass-through (default).
- No calling name or number sent in the forwarded Setup message.
- Calling name unconditionally set to the configured string in the forwarded Setup message.
- Calling number unconditionally set to the configured string in the forwarded Setup message.

You can configure alternative calling information treatments for calls exiting the SIP trunking gateway. Configurable treatment options are as follows:

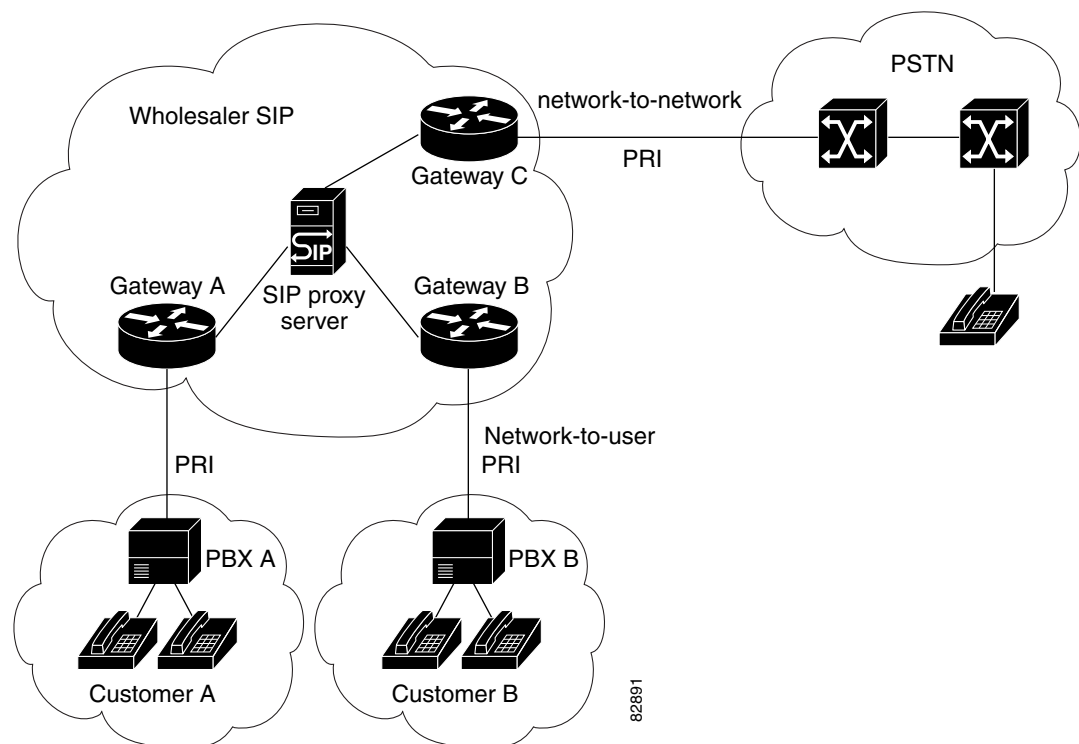
- Calling name and number pass-through (default).
- No calling name or number sent in the forwarded INVITE message.
- Display-name of the From header unconditionally set to the configured string in the forwarded INVITE message.
- User part of the From header unconditionally set to the configured string in the forwarded INVITE message.
- Display-name of the Remote-Party-ID header unconditionally set to the configured string in the forwarded INVITE message.
- User part of the Remote-Party-ID header unconditionally set to the configured string in the forwarded INVITE message.

Remote-Party-ID Implementation

This section discusses the implementation of the Remote-Party-ID feature in a SIP network. Before the implementation of this feature, there was no mechanism to modify the contents of the From header field. With the feature enabled, SIP gateways provide translation capability for ISDN screening and presentation identifiers in call setup messages. SIP gateways and proxy servers require configuration to support Remote-Party-ID feature.

[Figure 1](#) shows a typical network where the feature is implemented. Gateway C is configured for unscreened discard, that is, if the incoming SIP INVITE request does not contain a screened Remote-Party-ID header (;screen=yes), no calling name or number is sent in the forwarded Setup message.

Figure 1 Wholesaler SIP Network



Note

With respect to privacy and screening indication, it is the responsibility of the proxy server to protect display-name information and enforce privacy policies at the administrative boundary.

In the following sections, [Figure 2](#) through [Figure 9](#) illustrate various calling information treatment options using the commands available with the feature. Calling information treatment is determined by the parameters specified in the Setup message and Remote-Party-ID configuration on the SIP gateway.

Inbound and Outbound Call Flows

This section presents inbound and outbound call flows for the Remote-Party-ID feature. Figure 2 shows the SIP-to-PSTN default behavior where the calling party name and number are passed. The feature enables this treatment by default and no configuration is required.

Figure 2 SIP-to-PSTN Default Call Flow with Remote-Party-ID

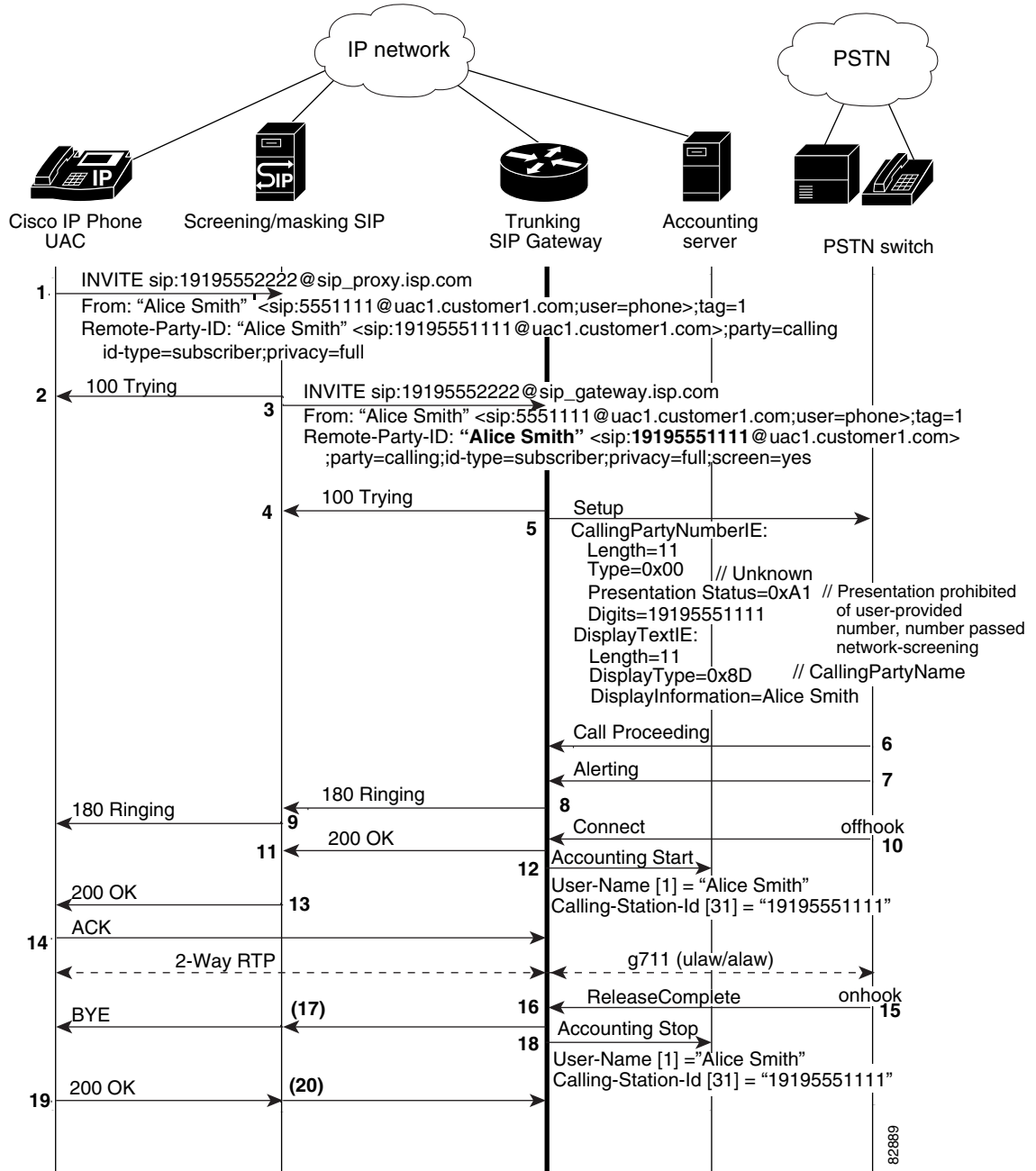
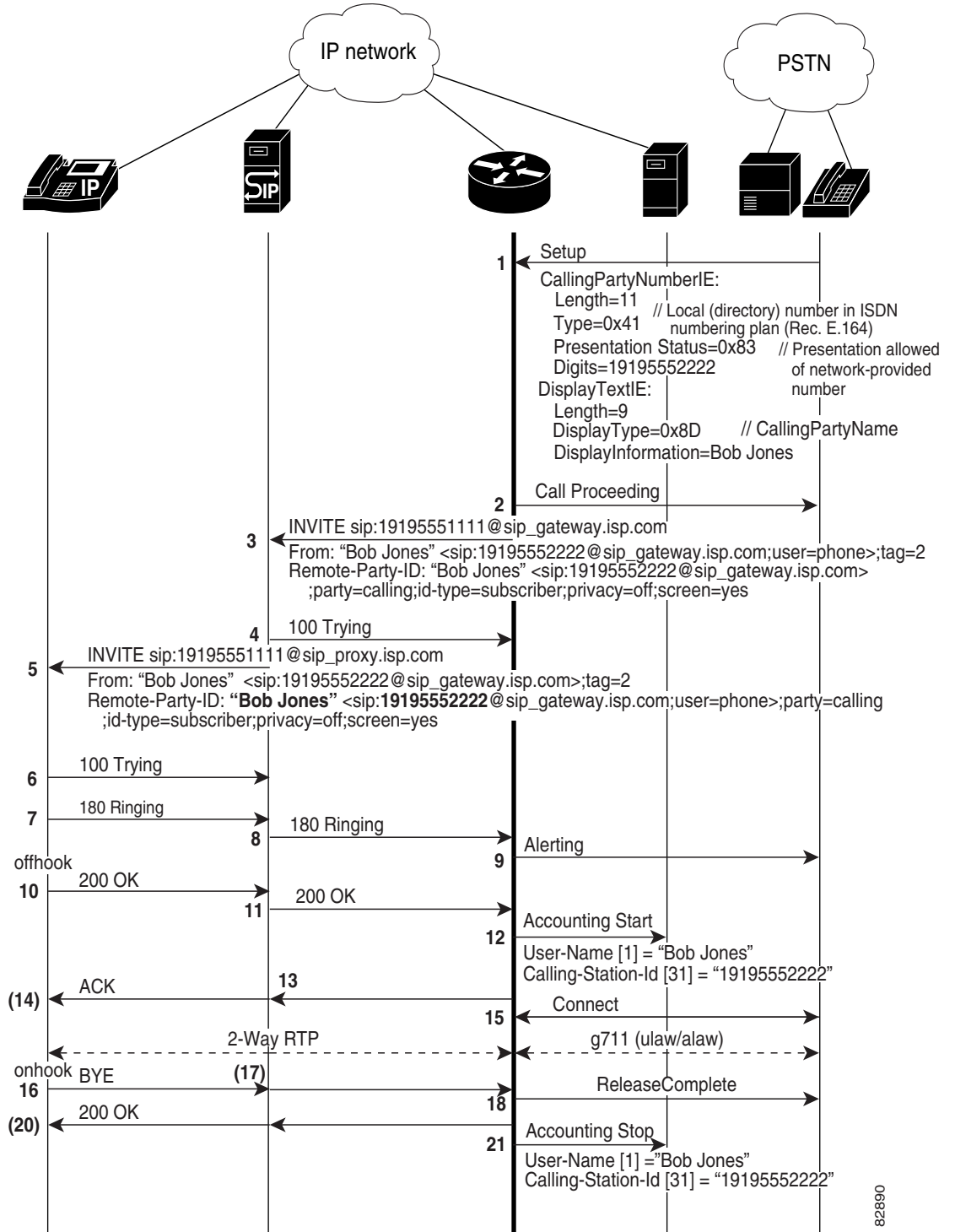


Figure 3 shows the PSTN-to-SIP default behavior where the calling party name and number are passed. This feature enables this treatment by default and no configuration is required.

Figure 3 *PSTN-to-SIP Default Call Flow with Remote-Party-ID Translation, No Privacy Requested*



82890

Figure 4 shows the call flow for discarding the calling name and number at Gateway B. The Setup message includes ISDN information elements (IEs) that specify calling information treatment. The INVITE message from Gateway A includes the corresponding Remote-Party-ID SIP tags.

Figure 4 Discarding Calling Name and Number at Gateway

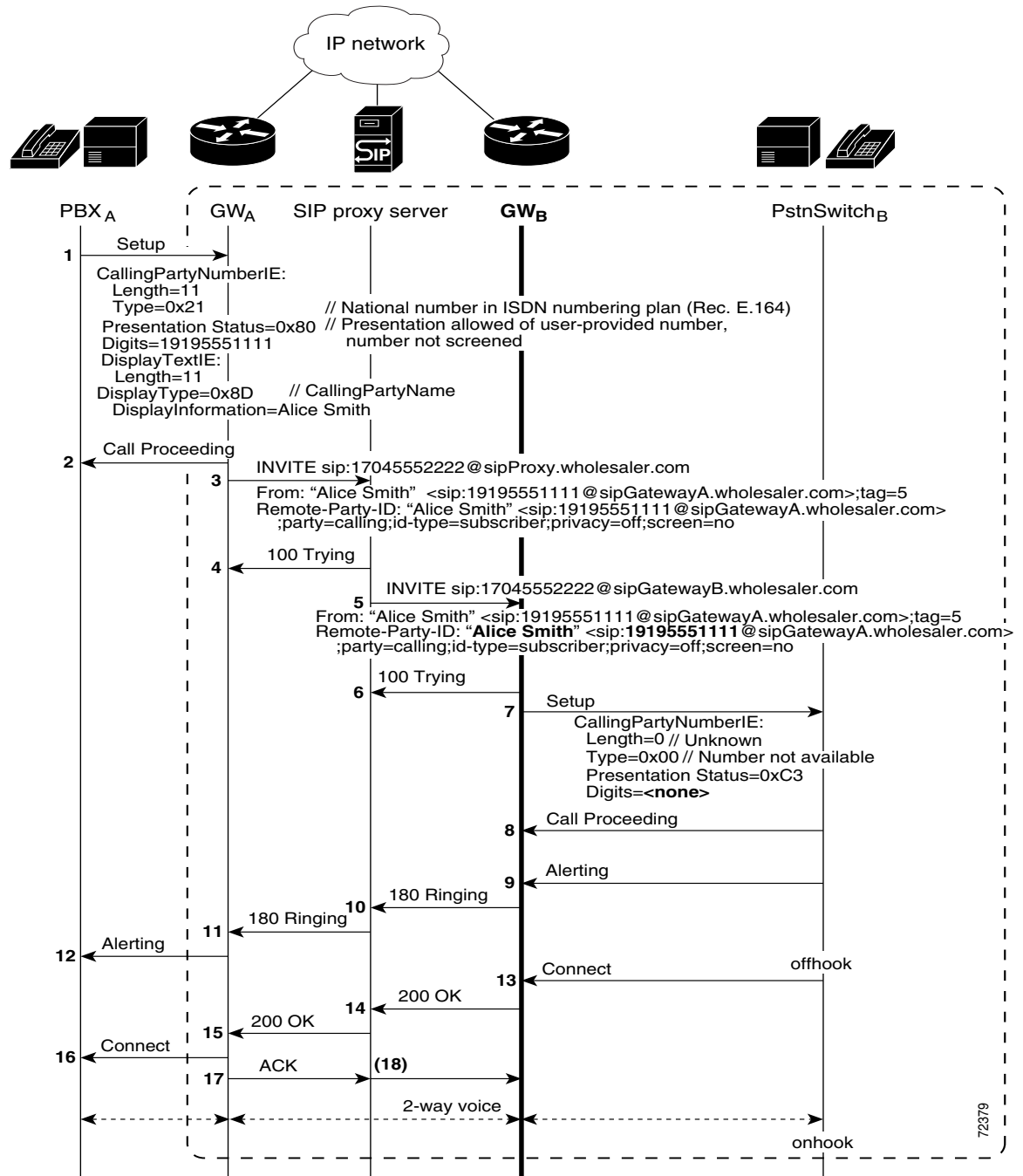
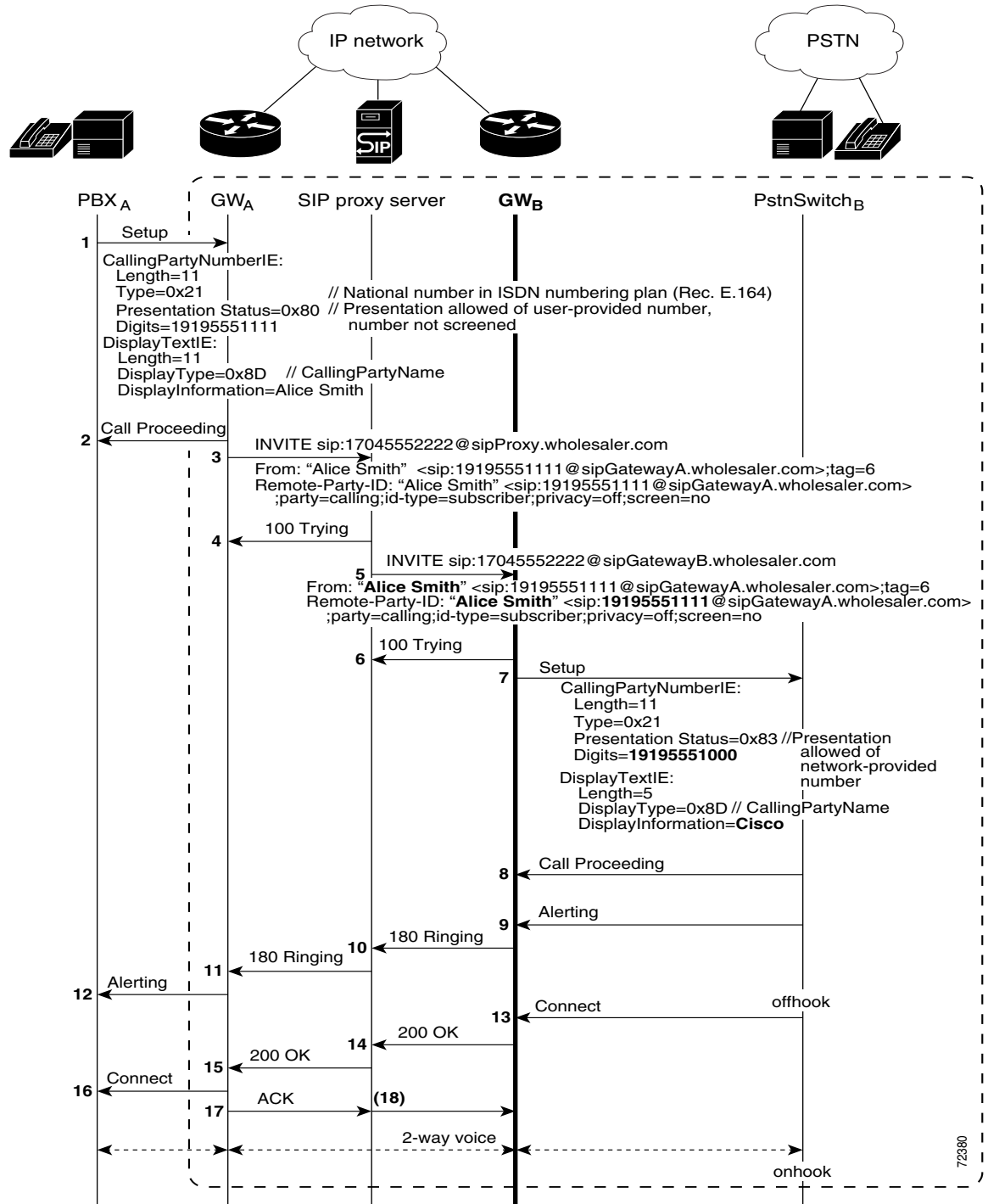


Figure 5 shows Gateway B overriding the calling name and number received in the Setup message from Gateway A. To configure Gateway B to override calling name and number, use the following commands:

- **remote-party-id**
- **calling-info sip-to-pstn name set *name***
- **calling-info sip-to-pstn number set *number***

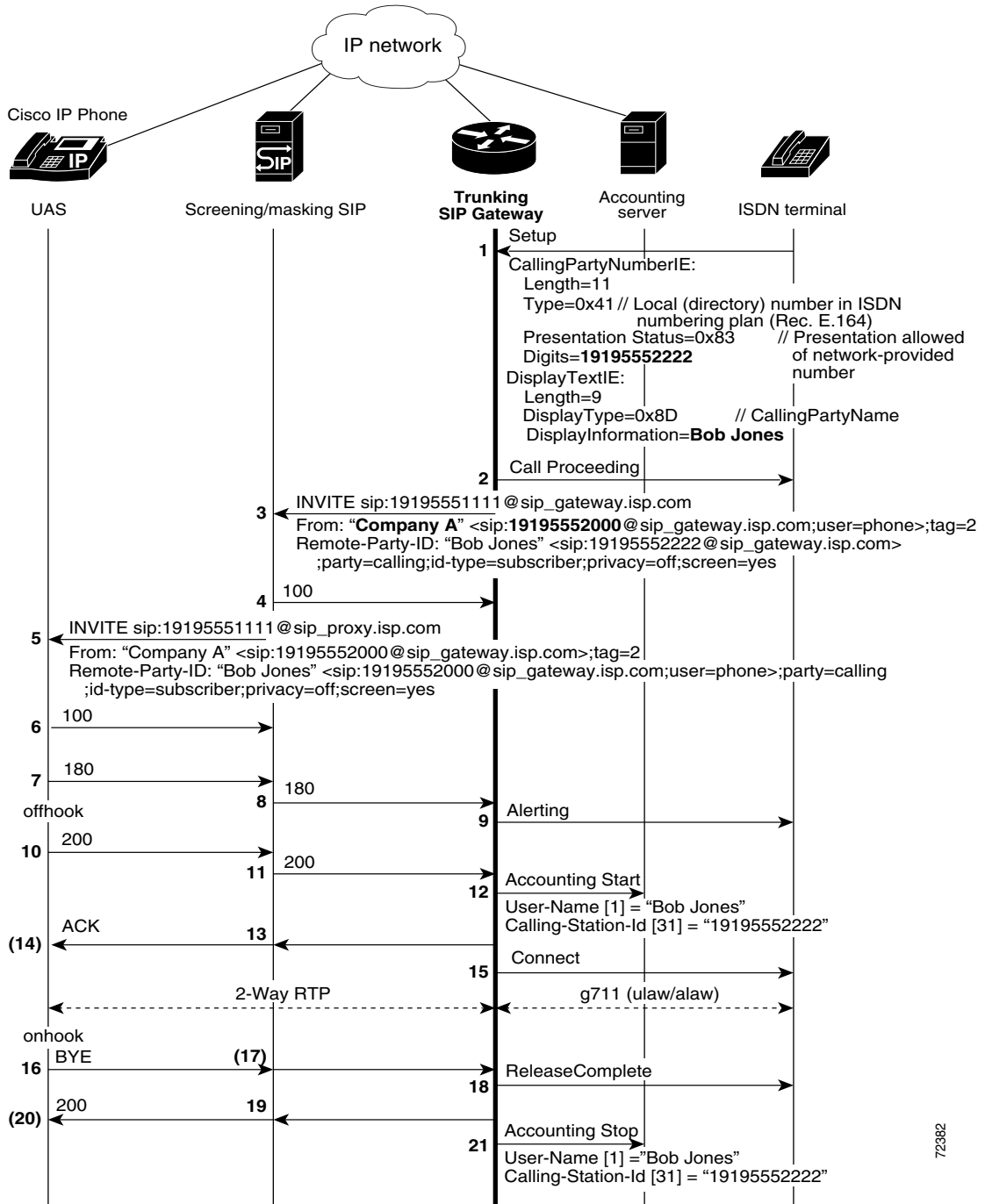
Figure 5 Overriding Calling Name and Number at Gateway



In [Figure 6](#) the trunking SIP gateway is configured to override the calling name and number of the From header. To configure this call treatment option, use the following commands:

- remote-party-id
- calling-info pstn-to-sip from name set *name*
- calling-info pstn-to-sip from number set *number*

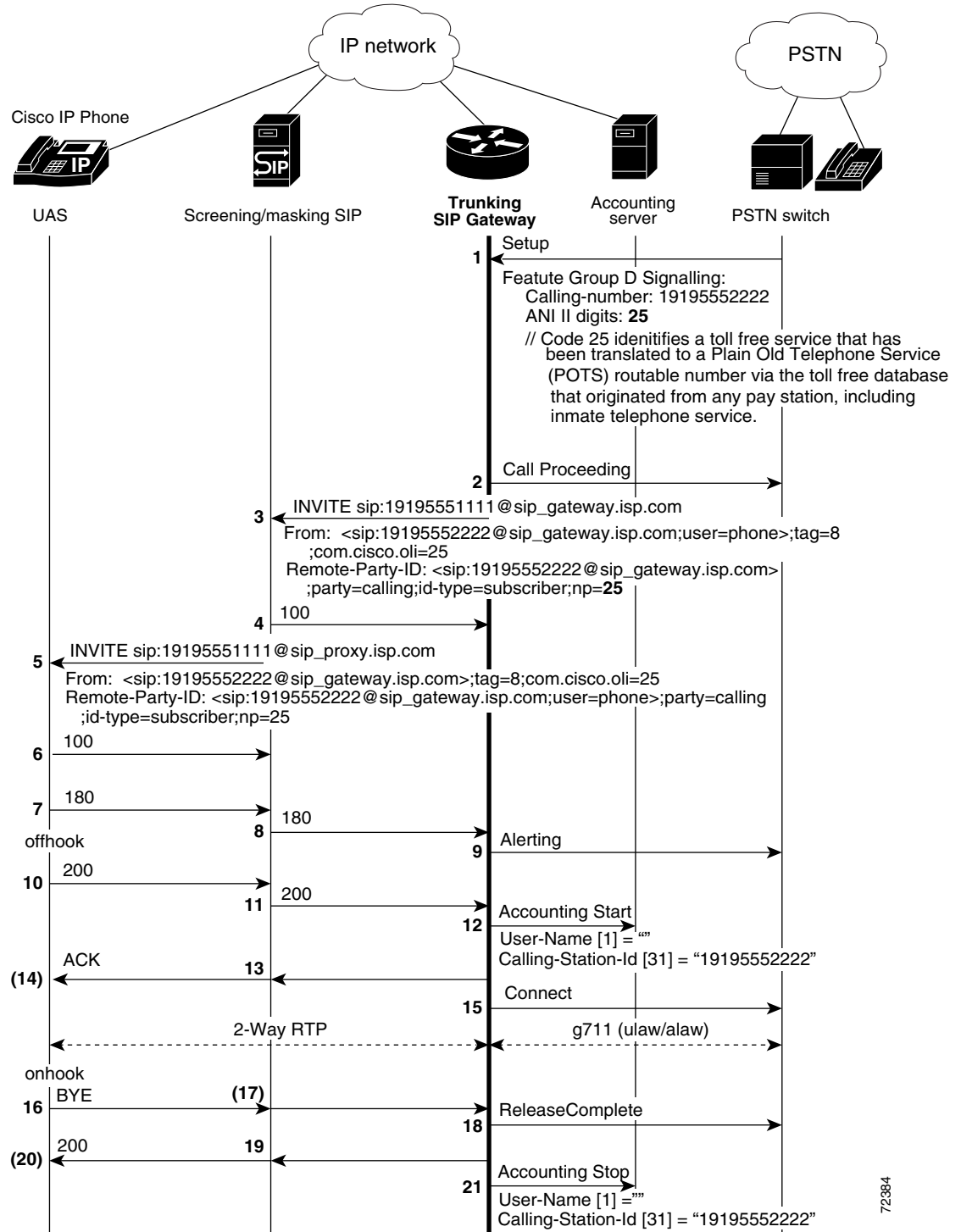
Figure 6 Overriding Calling Name and Number of From Header



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Figure 7 shows translation of OLI or ANI II digits for a billing application. The Remote-Party-ID feature enables this treatment by default; no configuration tasks are required. If the feature was disabled by using the **no remote-party-id** command, use the **remote-party-id** command to re-enable the feature.

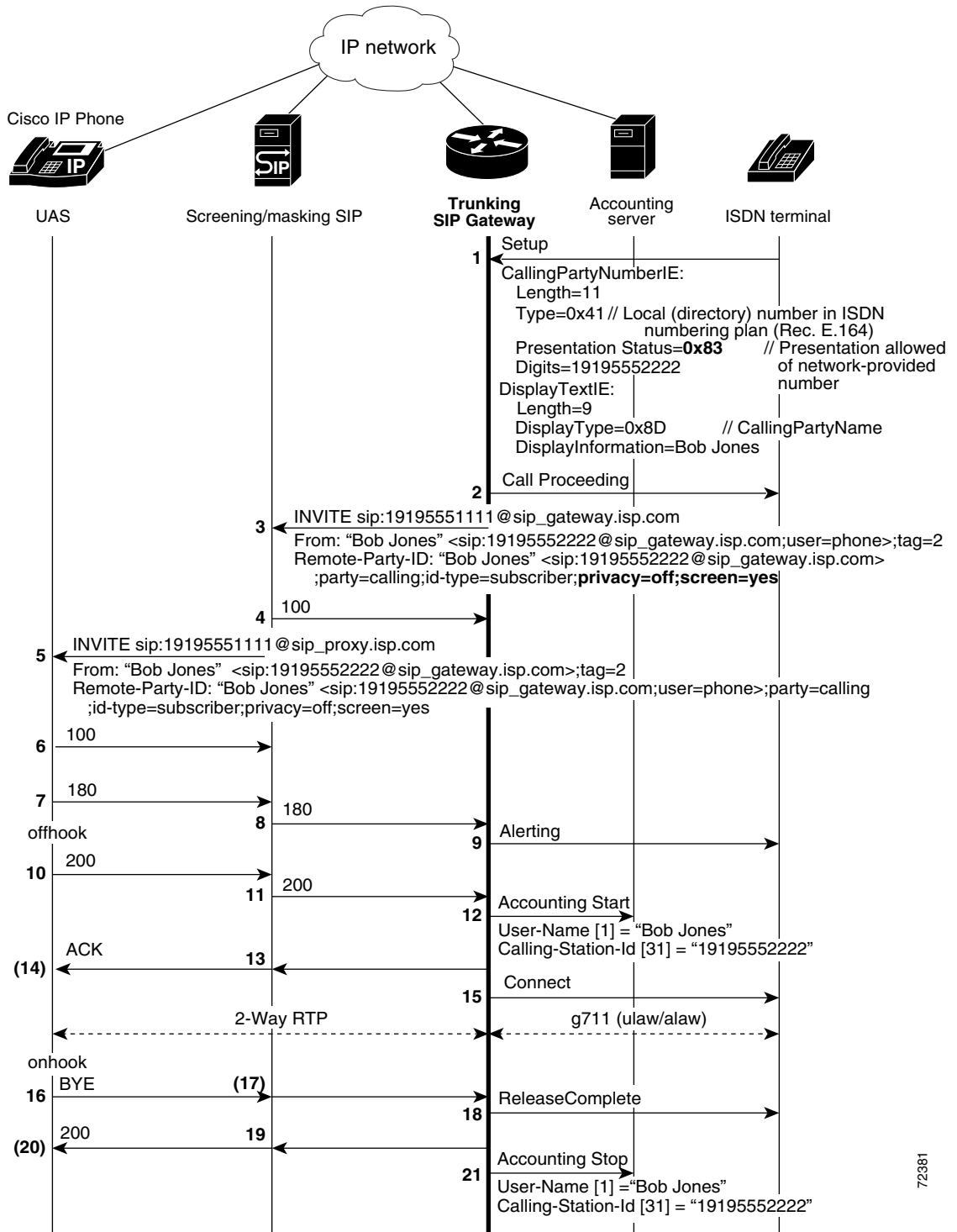
Figure 7 Passing OLI from CAS to SIP



72384

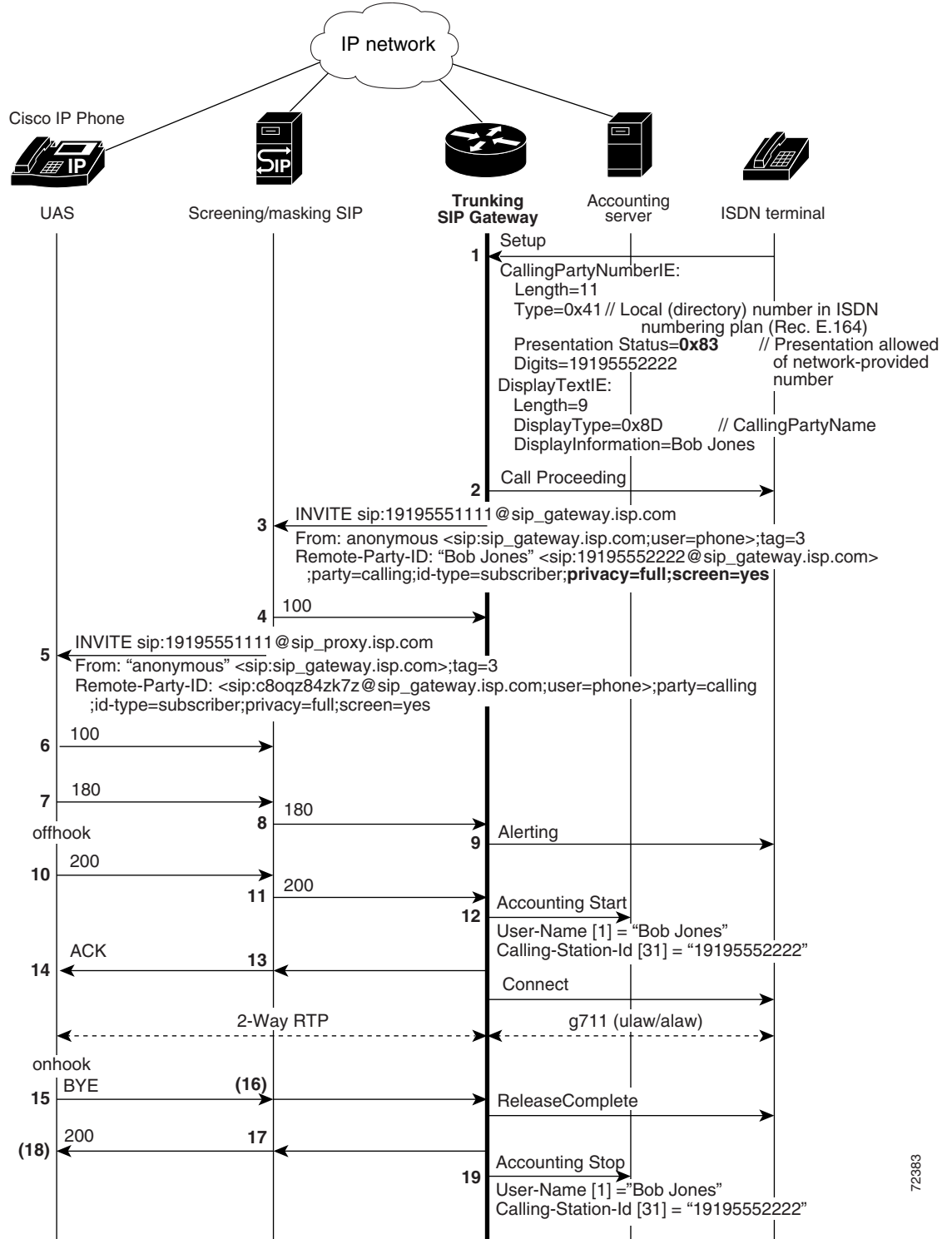
Figure 8 and Figure 9 show the SIP trunking gateway capability to provide translation between ISDN screening and presentation identifiers and SIP Remote-Party-ID extensions. The two figures show the difference in call treatment, with and without privacy requested. With no privacy requested, the calling party name and number are passed unchanged.

Figure 8 PSTN-to-SIP Call Flow with Remote-Party-ID Translation, No Privacy Requested



With privacy requested, as shown in Figure 9, screened identity information is still logged in accounting records for billing information, but the user field is not populated in the From header of the outgoing INVITE message, and the display-name is populated with “anonymous.”

Figure 9 PSTN-to-SIP Call Flow with Remote-Party-ID, Privacy Requested



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Remote-Party-ID in SIP and PSTN Messages

The ability to provide marking, screening, and PSTN translation of identity information to and from Remote-Party-ID extensions is supported in SIP INVITE and PSTN messages. This section discusses the formats of SIP INVITE and PSTN messages, and has the following subsections:

- [Remote-Party-ID Header, page 24](#)
- [Remote-Party-ID Syntax, page 24](#)
- [ISDN Syntax, page 25](#)
- [Screening and Presentation Information, page 25](#)

Remote-Party-ID Header

The SIP Remote-Party-ID header identifies the calling party and includes user, party, screen and privacy headers that specify how a call is presented and screened. The header contains a URL and an optional display name that identifies a user. A valid Remote-Party-ID header may be either a SIP URL or a TEL URL.



Note For information on header syntax, see the [“Remote-Party-ID Syntax” section on page 24](#) and [“Screening and Presentation Information” section on page 25](#).

The following example shows representative Remote-Party-ID headers, including user, party, screen, and privacy.

```
02:32:17:Received:
INVITE sip:3331000@172.27.184.118:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP 10.0.0.1:5070
Supported:org.ietf.sip.100rel
From:"alice" <sip:555-1001@10.0.0.1:5070>
To:sip:555-1002@172.27.184.118:5060
Remote-Party-ID:"Alice Smith"
<sip:5551111@192.0.2.67;user=phone>;party=calling;screen=no;privacy=off
Call-ID:00000001@10.0.0.1:5070
CSeq:1 INVITE
Contact:"alice" <sip:10.0.0.1:5070>
Content-Type:application/sdp

v=0
o=- 2890844526 2890844526 IN IP4 A3C47F2146789F0
s=-
c=IN IP4 10.0.0.1
t=36124033 0
m=audio 49170 RTP/AVP 0
```

Remote-Party-ID Syntax

Remote-Party-ID fields identify the calling party depending upon how the field is marked. If the party is unmarked, a Remote-Party-ID in a header represents the identity of the calling party.

Remote-Party-ID follows the Augmented Backus-Naur Format (ABNF). Refer to draft-ietf-sip-privacy-02.txt for the definitive specification. Fields are as follows:

- Remote-Party-ID = "Remote-Party-ID" ":" [display-name] "<" addr-spec ">" *(";" rpi-token)
- rpi-token = rpi-screen | rpi-pty-type | rpi-id-type | rpi-privacy | other-rpi-token
- rpi-screen = "screen" "=" ("no" | "yes")

- rpi-pty-type = "party" "=" ("calling" | "called" | token)
- rpi-id-type = "id-type" "=" ("subscriber" | "user" | "alias" | "return" | "term" | token)
- rpi-privacy = "privacy" "=" 1#("full" | "name" | "uri" | "off" | token) ["-" ("network" | token)])
- other-rpi-token = ["-"] token ["=" (token | quoted-string)]

ISDN Syntax

ISDN messages follow the format specified in *ISDN Primary Rate Interface Call Control Switching and Signalling Generic Requirements for Class II Equipment*, TR-NWT-001268, Revisions 1-4, Telcordia Technologies Technical Reference, 2001 and *ISDN Basic Rate Interface Call Control Switching and Signalling Generic Requirements*, GR-268-CORE, July 1998, to signal call control. ISDN messages are composed of information elements (IEs). The Cisco IOS VoiceXML feature uses Calling Party Number and Display Text IEs to provide specified screening and presentation treatment. The Calling Party Number IE specifies the origin of the calling number and presentation status, and the Display Text IE supplies calling party name information that is formatted for display by a terminal for a human user. See the Setup message in [Figure 2](#) for sample IE information.

Screening and Presentation Information

The Remote-Part-ID header and ISDN Setup messages contain tags used to specify screened identity information. [Table 5](#) lists translation of screening and presentation information included in the Remote-Party-ID SIP tags for SIP to PSTN networks. [Table 6](#) provides the same translation for PSTN to SIP networks.

Table 5 SIP to PSTN Translation of Screening and Presentation Information

Remote-Party-ID SIP Tags	PSTN Octet 3A
;privacy=off;screen=no	Presentation allowed of user-provided number, number not screened (0x80)
;privacy=off;screen=yes	Presentation allowed of user-provided number, number passed network screening (0x81)
;privacy=[full urilname];screen=no	Presentation prohibited of user-provided number, number not screened (0xA0)
;privacy=[full urilname];screen=yes	Presentation prohibited of user-provided number, number passed network screening (0xA1)
;screen=no	Presentation allowed of user-provided number, number not screened (0x80)
;screen=yes	Presentation allowed of user-provided number, number passed network screening (0x81)
;privacy=off	Presentation allowed of user-provided number, number not screened (0x80)
;privacy=[full urilname]	Presentation prohibited of user-provided number, number not screened (0xA0)
(no screen or privacy tags)	Presentation allowed of user-provided number, number not screened (0x80)

Table 6 PSTN to SIP Translation of Screening and Presentation Information

PSTN Octet 3A	Remote-Party-ID SIP Tags
Presentation allowed of user-provided number, number not screened (0x80)	;privacy=off;screen=no
Presentation allowed of user-provided number, number passed network screening (0x81)	;privacy=off;screen=yes
Presentation allowed of user-provided number, number failed network screening (0x82)	;privacy=off;screen=no
Presentation allowed of network-provided number (0x83)	;privacy=off;screen=yes
Presentation prohibited of user-provided number, number not screened (0xA0)	;privacy=full;screen=no
Presentation prohibited of user-provided number, number passed network screening (0xA1)	;privacy=full;screen=yes
Presentation prohibited of user-provided number, number failed network screening (0xA2)	;privacy=full;screen=no
Presentation prohibited of network-provided number (0xA3)	;privacy=full;screen=yes
Number not available (0xC3)	(no screen or privacy tags are sent)

Table 7 lists the corresponding translation for ISDN tags in binary and hex formats.

Table 7 ISDN Tags in Binary and Hex Formats

Binary (Bits) 8 7 6 5 4 3 2 1	Hex	Meaning
1 0 0 0 0 0 0 0	0x80	Presentation allowed of user-provided number, number not screened
1 0 0 0 0 0 0 1	0x81	Presentation allowed of user-provided number, number passed network screening
1 0 0 0 0 0 1 0	0x82	Presentation allowed of user-provided number, number failed network screening
1 0 0 0 0 0 1 1	0x83	Presentation allowed of network-provided number
1 0 1 0 0 0 0 0	0xA0	Presentation prohibited of user-provided number, number not screened
1 0 1 0 0 0 0 1	0xA1	Presentation prohibited of user-provided number, number passed network screening
1 0 1 0 0 0 1 1	0xA3	Presentation prohibited of network-provided number
1 1 0 0 0 0 1 1	0xC3	Number not available

Benefits of SIP Extensions for Caller Identity and Privacy

- Expands PSTN interoperability
- Supports the ability to override privacy and screening indicators
- Enables network verification and screening of a call participant identity by SIP proxy servers
- Supports logging of screened identity information in accounting records for billing information

- Provides enhanced subscriber information that supports the enabling of service creation platforms and application servers for service providers
- Allows the service provider enhanced control of the ability to identify a subscriber and its qualifications within the network

SIP INVITE Request with Malformed Via Header

A SIP INVITE requests that a user or service participate in a session. Each INVITE contains a Via header that indicates the transport path taken by the request so far, and where to send a response.

In the past, when an INVITE contained a malformed Via header, the gateway would print a debug message and discard the INVITE without incrementing a counter. However, the printed debug message was often inadequate, and it was difficult to detect that messages were being discarded.

The SIP INVITE Request with Malformed Via Header feature provides a response to the malformed request. A counter, *Client Error: Bad Request*, increments when a response is sent for a malformed Via field. *Bad Request* is a class 400 response and includes the explanation *Malformed Via Field*. The response is sent to the source IP address (the IP address where the SIP request originated) at User Datagram Protocol (UDP) port 5060.



Note

This feature applies to messages arriving on UDP, because the Via header is not used to respond to messages arriving on TCP.

Feature benefits include the following:

- The system now increments a counter and sends a response, rather than simply discarding an INVITE message that contains a malformed Via header.
- The counter provides a useful and immediate indication that an INVITE message has been discarded, and the response allows the result to be propagated back to the sender.

SIP Session Timer Support

The SIP Session Timer Support feature adds the capability to periodically refresh SIP sessions by sending repeated INVITE requests. The repeated INVITE requests (re-INVITEs), are sent during an active call leg to allow user agents or proxies to determine the status of a SIP session. Without this keepalive mechanism, proxies that remember incoming and outgoing requests (stateful proxies) may continue to retain call state needlessly. If a user agent fails to send a BYE message at the end of a session or if the BYE message is lost because of network problems, a stateful proxy does not know that the session has ended. The re-INVITEs ensure that active sessions stay active and completed sessions are terminated.

In addition to re-INVITEs, UPDATE can also be used as a method for session keepalives. The SIP stack supports both re-INVITE and UPDATE. The gateway continues to use re-INVITE for session refresh.

The SIP Session Timer Support feature also adds two new general headers that are used to negotiate the value of the refresh interval.

- A Session-Expires header is used in an INVITE if the user-agent client (UAC) wants to use the session timer.
- The Minimum Session Expiration (Min-SE) header conveys the minimum allowed value for the session expiration.

Role of the User Agents

The initial INVITE request establishes the duration of the session and may include a Session-Expires header and a Min-SE header. These headers indicate the session timer value required by the UAC. A receiving user-agent server (UAS) or proxy can lower the session timer value, but not lower than the value of the Min-SE header. If the session timer duration is lower than the configured minimum, the proxy or UAS can also send out a 422 response message. If the UAS or proxy finds that the session timer value is acceptable, it copies the Session-Expires header into the 2xx class response.

A UAS or proxy can also insert a Session-Expires header in the INVITE if the UAC did not include one. Thus a UAC can receive a Session-Expires header in a response even if none was present in the request.

In the 2xx response, the *refresher* parameter in the Session-Expires header indicates who performs the re-INVITES or UPDATE. For example, if the parameter contains the value UAC, the UAC performs the refreshes. For compatibility issues, only one of the two user agents needs to support the session timer feature, and in that case, the user agent that supports the feature performs the refreshes.

Re-INVITES are processed identically to INVITE requests, but go out in predetermined session intervals. Re-INVITES carry the new session expiration time. The user agent that is responsible for generating re-INVITE requests sends a re-INVITE out before the session expires. If there is no response, the user agent sends a BYE request to terminate the call before session expiration. If a re-INVITE is not sent before the session expiration, either the UAC or the UAS can send a BYE.

If the 2xx response does not contain a Session-Expires header, there is no session expiration and re-INVITES do not need to be sent.

Session-Expires Header

The Session-Expires header conveys the session interval for a SIP call. It is placed in an INVITE request and is allowed in any 2xx class response to an INVITE. Its presence indicates that the UAC wishes to use the session timer for this call. Unlike the SIP-Expires header, it can only contain a delta-time, which is the current time, plus the session interval from the response.

For example, if a UAS generates a 200 OK response to a INVITE that contained a Session-Expires header with a value of 90 seconds (1.5 minutes), the UAS computes the session expiration as 1.5 minutes after the time when the 200 OK response was sent. For each proxy, the session expiration is 1.5 minutes after the time when the 2xx was received or sent. For the UAC, the expiration time is 1.5 minutes after the receipt of the final response.

When the gateway acts as an UAS, it is responsible for refreshes. The refresh interval is a minimum of 32 seconds, or one-third the refresh interval. When the gateway act as an UAC, the refresh interval is one-half the refresh interval.

If the session is not refreshed, the minimum time to send a BYE before the session expires is 32 seconds.

The recommended value for the Session-Expires header is 90 seconds.

The syntax of the Session-Expires header is as follows:

```
Session-Expires = ("Session-Expires" | "x") ":" delta-seconds
                  [refresher]
refresher        = ";" "refresher" "=" "UAS" | "UAC"
```

The *refresher* parameter is optional in the initial INVITE, although the UAC can set it to UAC to indicate that it will do the refreshes. The 200 OK response must have the refresher parameter set.

Min-SE Header

Because of the processing load of INVITE requests, the proxy, UAC, and UAS can have a configured minimum timer value that they can accept. The **min-se** (SIP) command sets the minimum timer, and it is conveyed in the Min-SE header in the initial INVITE request.

When making a call, the presence of the Min-SE header informs the UAS and any proxies of the minimum value that the UAC accepts for the session timer duration, in units of delta-seconds. The default value is 90 seconds (1.5 minutes). By not reducing the session interval below the value set, the UAS and proxies prevent the UAC from having to reject a call with a 422 error. Once set, the **min-se** command value affects all calls originated by the router. If the Min-SE header is not present, the user agent accepts any value.

The syntax of the Min-SE header is:

```
Min-SE = "Min-SE" ":" delta-seconds
```

422 Response Message

If the value of the Session-Expires header is too small, the UAS or proxy rejects the call with a 422 *Session Timer Too Small* response message. With the 422 response message, the proxy or UAS includes a Min-SE header indicating the minimum session value it can accept. The UAC may then retry the call with a larger session timer value.

If a 422 response message is received after an INVITE request, the UAC can retry the INVITE.

Supported and Require Headers

The presence of the *timer* argument in the Supported header indicates that the user agent supports the SIP session timer. The presence of the *timer* argument in the Require header indicates that the opposite user agent must support the SIP session timer for the call to be successful.

Benefits of SIP Session Timer Support

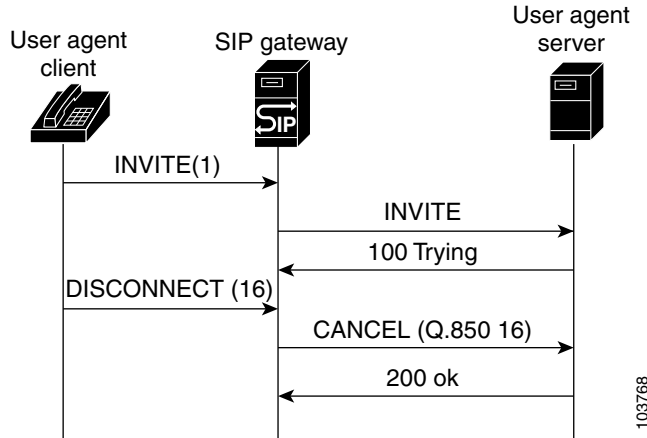
- This feature provides a periodic refresh of SIP sessions. The periodic refresh allows user agents and proxies to monitor the status of a SIP session, preventing hung network resources when network failures occur.
- Only one of the two user-agent or proxy participants in a call needs to have the SIP Session Timer Support feature implemented. This feature is easily compatible with older SIP networks.

SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion

Reason Header

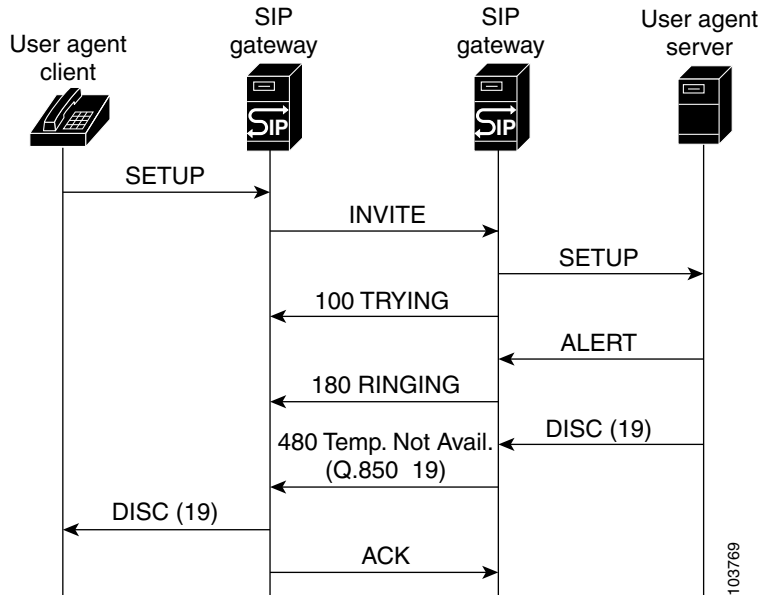
The Reason header facilitates PSTN interworking. This is achieved by having the side receiving a Disconnect message response append a Reason header to the outgoing Bye or Cancel message request and 4xx, 5xx, or 6xx message response, indicating the Q.850 cause code that passed down from the PSTN (see [Figure 10](#)).

Figure 10 PSTN Interworking Using Reason Header Example



SIP implementations on PSTN gateways are plagued with issues related to mapping ISDN-disconnect message-request cause codes to SIP response status codes, which stem from the mapping on the gateway receiving the disconnect. Specifically, more than one ISDN-disconnect message-request cause code maps to one SIP status code. For example, on SIP gateways, ISDN cause codes 18, 19, and 20 all map to the SIP status code of 480 message response. This makes it impossible to deterministically relay the cause-code value on the remote end. The Reason header can carry the actual cause code (see [Figure 11](#)).

Figure 11 Reason Header in Action; Extinguishing the Ambiguity in SIP Status Codes



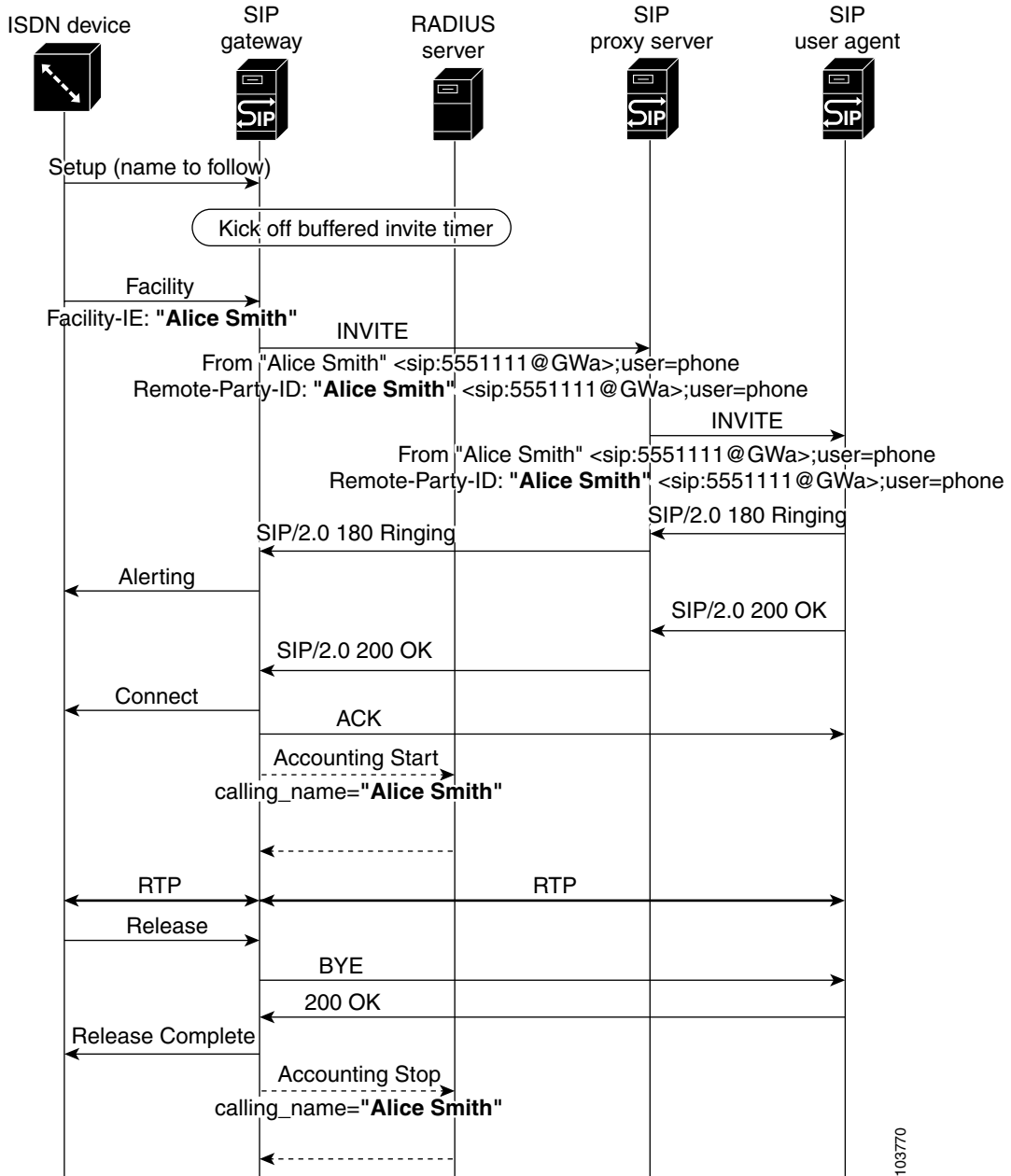
Buffered Calling-Name Completion

As shown in [Figure 12](#), Cisco IOS SIP has always supported receiving calling-name information in the display information element (IE) of a Setup message request. Support for receiving calling-name information in the facility IE of a Setup message request, of a Facility message request, and of a NOTIFY message request were supported through the Support for the ISDN Calling Name Display feature in release 12.3(4)T (refer to the “[Configuring SIP DTMF Features](#)” chapter).

The Buffered Calling Name Completion feature adds support for buffering the INVITE message request when the calling-name information is going to arrive in a subsequent facility IE of a Facility message request.

When an originating gateway (OGW) receives a Setup message with an indication that calling-name information is enabled, the configuration is checked for INVITE-message display-name buffering. When buffering is enabled, the INVITE message is buffered until the time specified in the configuration. If a Facility message with display information in the From and Remote Party ID headers of the INVITE message is received, then send it out. If no Facility message is received in the specified time, send out only the INVITE message.

Figure 12 Calling Name in Facility IE of Facility



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SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers

The SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers feature provides a mechanism for applications to send and receive SIP headers and to send SUBSCRIBE messages and receive NOTIFY events. Where appropriate, this section discusses separately the features that make up this feature set, the SIP Header Support feature along with the SUBSCRIBE and NOTIFY for External Triggers feature.

Feature benefits include the following:

- Enables the creation of presence-based, subscribe-to-be-notified services that are triggered by events external to a session
- Allows service providers to expand services to include VoiceXML-driven voice browser applications
- Allows the SIP gateway to subscribe to triggered applications and custom event-packages
- Supports distributed voice-web scenarios and call and contact center integration applications by providing access to SIP headers

This section contains the following information:

- [Feature Design of SIP Header Support, page 33](#)
- [Feature Design of SIP SUBSCRIBE and NOTIFY for External Triggers, page 33](#)

Feature Design of SIP Header Support

Prior to the implementation of this feature, voice applications running on the gateway did not have access to headers sent in SIP requests. The SIP Header Passing feature makes SIP headers, the fields which specify session details in SIP messages, available to applications. This feature supports the following capabilities for VoiceXML and Tcl IVR 2.0 applications:

- Set SIP headers for outgoing SIP INVITE messages.
- Obtain information about SIP headers for incoming calls and create session variables to access the headers in VoiceXML document or Tcl IVR 2.0 script.
- Set and obtain extended and non-standard headers (user-defined header attribute-value pairs)

Using headers in SIP INVITE messages, voice applications can pass information about a call to an application on another server. For example, if the caller has entered an account number and the application transfers the call to another application on another platform, the account number can be passed in a SIP Header. An example scenario is an airline application transferring the call to a hotel reservation application hosted at a different service provider. This feature enables the respective sites to share context information about the caller.

This feature introduces a new command, the **header-passing** command, to either enable or disable passing headers from INVITE messages to applications.

The SIP Header Passing feature also provides enhanced inbound and outbound dial-peer matching services.

Feature Design of SIP SUBSCRIBE and NOTIFY for External Triggers

This feature implements support for two SIP methods, SUBSCRIBE and NOTIFY, and for a new Event header, as defined in the IETF draft, draft-roach-sip-subscribe-notify-02.txt, *Event Notification in SIP*. More detailed information for this feature is described in the following sections:

- [Overview of the SUBSCRIBE and NOTIFY for External Triggers Application, page 34](#)
- [Example of a SUBSCRIBE and NOTIFY for External Triggers Application, page 34](#)
- [RFC 3265 Compliance for the SUBSCRIBE and NOTIFY for External Triggers Feature, page 35](#)
- [SUBSCRIBE and NOTIFY Message Flow, page 35](#)
- [Sample Messages, page 37](#)

Overview of the SUBSCRIBE and NOTIFY for External Triggers Application

The SIP event notification mechanism uses the SUBSCRIBE method to request notification of an event at a later time. The NOTIFY method provides notification that an event which has been requested by an earlier SUBSCRIBE method has occurred, or provides further details about the event. The new feature makes headers in incoming SIP INVITE, SUBSCRIBE, and NOTIFY messages available to applications for use in event subscription. Similarly, to allow an application to place an outbound call using SIP, this feature passes headers in the URL for use by the SIP service provider interface (SPI) to create an outgoing INVITE request.

The new feature also supports the capability to subscribe to standard event packages, such as Message Waiting Indicator and Presence, and to application-specific custom event packages, as defined in SIP-Specific Event Notification, an earlier draft of RFC 3265, *Session Initiation Protocol (SIP)-Specific Event Notification*.



Note

For information on these capabilities, see the following:

- [Cisco IOS Tcl IVR and VoiceXML Application Guide](#).
- [Cisco VoiceXML Programmer's Guide](#).
- [Tcl IVR API Version 2.0 Programming Guide](#).

Cisco implements the SUBSCRIBE and NOTIFY for External Triggers feature using the Application SUBSCRIBE/NOTIFY Layer (ASNL). ASNL is a software interface layer between the application and signaling protocol stacks that allows the application to subscribe to interested events and to pass notification when it is received.

The SUBSCRIBE and NOTIFY for External Triggers allows external SIP servers to trigger a particular voice application, behavior or activity on Cisco voice gateways. For example, a client application on the gateway subscribes to a particular event in a server. When the event takes place, the server notifies the client of that event. On receiving this event notification, the client application triggers a particular action in the gateway. The client and server must mutually agree on the events they can handle and the processing of those events.

Example of a SUBSCRIBE and NOTIFY for External Triggers Application

The SUBSCRIBE and NOTIFY for External Triggers feature supports various applications of external triggers. In the following scenario, a user requests a stock reminder service, for example “Let me know if Stock X reaches 100. Here is a phone number to reach me.” The SUBSCRIBE and NOTIFY for External Triggers feature supports an application like this in the following manner:

- The user dials into the gateway.
- The gateway sends a subscription request to the server on the user's behalf. The subscription request contains details of the event: event name, expiration time, and other information related to the event. The request can contain any application specific headers and content.
- When the server determines, through some other means, that Stock X has reached 100, it sends a notification request to the client. The SIP NOTIFY request from the server can contain any application specific headers and content.
- This notification request triggers the client on the gateway to call the specified user or destination.

Other external trigger applications include mid-call triggers such as call center queuing and subscription to a wake-up call service.

RFC 3265 Compliance for the SUBSCRIBE and NOTIFY for External Triggers Feature

The Cisco implementation of SIP SUBSCRIBE and NOTIFY methods is based on an earlier draft of *SIP-Specific Event Notification*, and deviates from RFC 3265, *Session Initiation Protocol (SIP)-Specific Event Notification in the following capabilities:*

- The Cisco client does not support the following:
 - Embedded parameters in event package names.
 - Subscription-State header. To terminate a subscription, the notifier or user agent sends a NOTIFY request to the Cisco gateway with the Expires header set to zero.
 - Forking.
 - State deltas.
- In the Cisco SIP implementation, a subscription request always creates a new dialog, and cannot send a SUBSCRIBE request for an existing dialog.
- The Cisco SIP implementation does not prevent man-in-the-middle attacks as defined in RFC 3265.
- Event package registration with the IANA is not required; instead you have the flexibility to specify your own event package.

SUBSCRIBE and NOTIFY Message Flow

Figure 13 shows a typical message flow for SUBSCRIBE and NOTIFY messages.

Figure 13 SUBSCRIBE and NOTIFY Message Flow

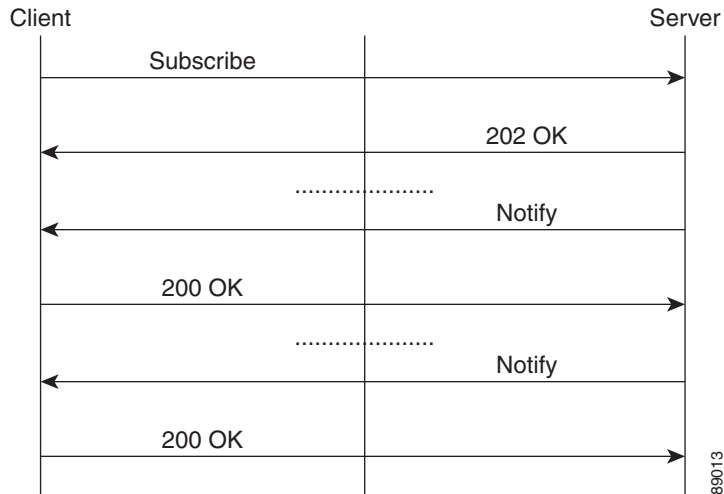


Figure 14 shows the message flow for a successful subscription.

Figure 14 Successful Subscription

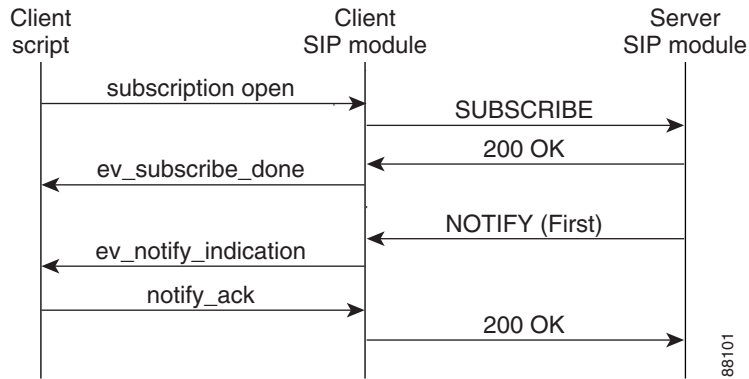


Figure 15 shows a completed subscription. The server can send any number of NOTIFY messages as long as the subscription is active.

Figure 15 Subscription Completed

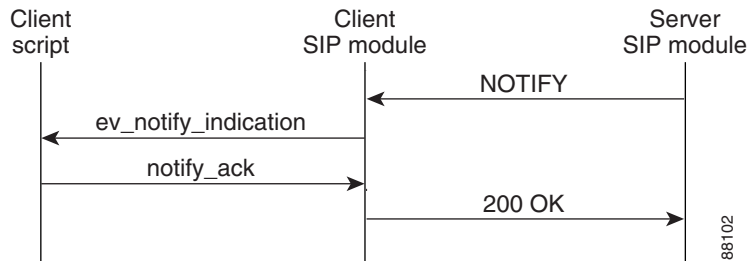


Figure 16 shows the message flow for subscription termination by the server.

Figure 16 Subscription Termination by the Server

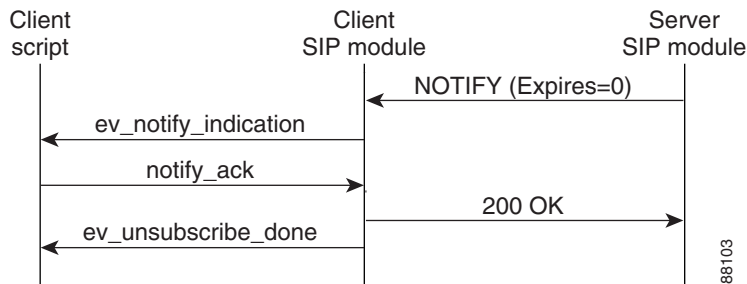
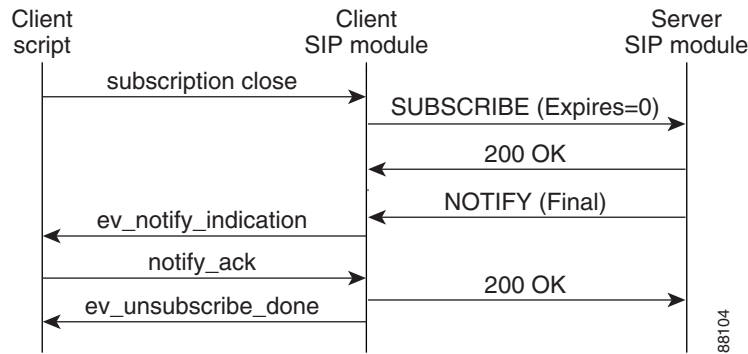


Figure 17 shows the message flow for subscription termination by the client.

Figure 17 Subscription Termination by the Client



Sample Messages

This section presents a sequence of SIP messages sent and received between gateways during the message flow shown in Figure 13 in the preceding section.

Example: Subscription Request Sent From Client

This example shows a SUBSCRIBE request sent to the server. The example includes a nonstandard Subject header and an Event header.

```

*Apr 19 08:38:52.525: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
Sending MWI client request to server
*Apr 19 08:38:52.525:
*Apr 19 08:38:52.529: Sent:
SUBSCRIBE sip:user@10.7.104.88:5060 SIP/2.0
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:10.7.102.35>;tag=1C24D44-20FD
To: <sip:user@10.7.104.88>
Date: Wed, 19 Apr 2000 08:38:52 UTC
Call-ID: C4BB7610-150411D4-802186E3-AD119804
CSeq: 101 SUBSCRIBE
Timestamp: 956133532
Subject: Hi There
Contact: <sip:10.7.102.35:5060>
Event: message-summary
Expires: 500 )
Content-Type: text/plain
Content-Length: 21

This is from client
    
```

Example: Subscription Response Received from the Server

This example shows a response from the server to a subscription request.

```

*Apr 19 08:38:52.537: Received:
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:10.7.102.35>;tag=1C24D44-20FD
To: <sip:user@10.7.104.88>;tag=1D80E90-2072
Date: Sun, 17 Nov 2002 02:59:19 GMT
Call-ID: C4BB7610-150411D4-802186E3-AD119804
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 956133532
Content-Length: 0
CSeq: 101 SUBSCRIBE
    
```

```
Expires: 500
*Apr 19 08:38:52.541: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: ***** act_Subscribe :
SUBSCRIPTION DONE received

*Apr 19 08:38:52.541: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: *** act_subscribe: subscription
status=sn_000
```

Example: NOTIFY Request from the Server

This example shows the initial NOTIFY request from a server and includes an application-specific nonstandard Hello header.

```
*Apr 19 08:38:52.545: Received:
NOTIFY sip:10.7.102.35:5060 SIP/2.0
Via: SIP/2.0/UDP 10.7.104.88:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Sun, 17 Nov 2002 02:59:19 GMT
Call-ID: C4BB7610-150411D4-802186E3-AD119804
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 1037501959
CSeq: 101 NOTIFY
Event: message-summary
Hello: Hello world
Contact: <sip:user@10.7.104.88:5060>
Content-Length: 43
Content-Type: text/plain
```

This is content(message body) from server

Example: An Application Reads Header and Body Information in a NOTIFY Request

This example shows an application accessing the From and Hello headers in the NOTIFY request.

```
*Apr 19 08:38:52.549: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: *** act_Notify : NOTIFY RECEIVED
t
*Apr 19 08:38:52.549: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
From header is: <sip:user@10.7.104.88>;tag=1D80E90-2072
*Apr 19 08:38:52.549: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
Hello header is: Hello world
*Apr 19 08:38:52.549: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
content_type received=text/plain
*Apr 19 08:38:52.549:
*Apr 19 08:38:52.553: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
content received=This is content(message body) from server
```

Example: NOTIFY Request Sent From the Client

This example shows a NOTIFY request sent from a client.

```
*Apr 19 08:38:52.553: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Wed, 19 Apr 2000 08:38:52 UTC
Call-ID: C4BB7610-150411D4-802186E3-AD119804
CSeq: 101 NOTIFY
Timestamp: 956133532
Event: message-summary
Content-Length: 0
```

Example: The Client receives a NOTIFY Message

This example shows a NOTIFY message received by a client.

```
c5300-5#
*Apr 19 08:38:57.565: Received:
NOTIFY sip:10.7.102.35:5060 SIP/2.0
Via: SIP/2.0/UDP 10.7.104.88:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Sun, 17 Nov 2002 02:59:19 GMT
Call-ID: C4BB7610-150411D4-802186E3-AD119804
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 1037501964
CSeq: 102 NOTIFY
Event: message-summary
Hello: Hello world
Contact: <sip:user@10.7.104.88:5060>
Content-Length: 35
Content-Type: text/plain

this is just a notify from server
*Apr 19 08:38:57.569: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: *** act_Notify : NOTIFY RECEIVED
*Apr 19 08:38:57.569: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
From header is: <sip:user@10.7.104.88>;tag=1D80E90-2072
*Apr 19 08:38:57.569: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
Hello header is: Hello world
*Apr 19 08:38:57.569: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
content_type received=text/plain
*Apr 19 08:38:57.569: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
content received=this is just a notify from server
```

Example: The Client Sends a NOTIFY Message

This example shows a client sending a NOTIFY message.

```
*Apr 19 08:38:57.573: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Wed, 19 Apr 2000 08:38:57 UTC
Call-ID: C4BB7610-150411D4-802186E3-AD119804
CSeq: 102 NOTIFY
Timestamp: 956133537
Event: message-summary
Content-Length: 0
```

Example: The Client Initiates a Subscription Termination

This example shows a client initiating a subscription termination request using the Expires header set to zero.

```
*Apr 19 08:38:57.577: Sent:
SUBSCRIBE sip:user@10.7.104.88:5060 SIP/2.0
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:10.7.102.35>;tag=1C24D44-20FD
To: <sip:user@10.7.104.88>;tag=1D80E90-2072
Date: Wed, 19 Apr 2000 08:38:57 UTC
Call-ID: C4BB7610-150411D4-802186E3-AD119804
CSeq: 102 SUBSCRIBE
Timestamp: 956133537
Subject: Hi There
Contact: <sip:10.7.102.35:5060>
```

```
Event: message-summary
Expires: 0
Content-Type: text/plain
Content-Length: 21
```

This is from client

Example: The Client Receives a Response to a Subscription Termination Request

This example shows a client receiving a response to a subscription termination request.

```
*Apr 19 08:38:57.589: Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:10.7.102.35>;tag=1C24D44-20FD
To: <sip:user@10.7.104.88>;tag=1D80E90-2072
Date: Sun, 17 Nov 2002 02:59:24 GMT
Call-ID: C4BB7610-150411D4-802186E3-AD119804
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 956133532
Content-Length: 0
CSeq: 102 SUBSCRIBE
Expires: 0
Contact: <sip:user@10.7.104.88:5060>
```

Example: The Client Receives a Final NOTIFY Message

This example shows a client receiving a final NOTIFY message that a subscription is finished.

```
c5300-5#
*Apr 19 08:39:02.585: Received:
NOTIFY sip:10.7.102.35:5060 SIP/2.0
Via: SIP/2.0/UDP 10.7.104.88:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Sun, 17 Nov 2002 02:59:24 GMT
Call-ID: C4BB7610-150411D4-802186E3-AD119804
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 1037501969
CSeq: 103 NOTIFY
Event: message-summary
Hello: Hello world
Contact: <sip:user@10.7.104.88:5060>
Content-Length: 35
Content-Type: text/plain
```

this is just a notify from server

```
*Apr 19 08:39:02.589: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: *** act_Notify : FINAL NOTIFY
RECEIVED
*Apr 19 08:39:02.589: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: status=sn_004
*Apr 19 08:39:02.589: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd:
From header is: <sip:user@10.7.104.88>;tag=1D80E90-2072
*Apr 19 08:39:02.593: //-1//TCL2:HN01C24D3C:/tcl_PutsCmd: act_UnsubscribeDone : !!!
SUBSCRIPTION IS OVER !!!
```

Example: A Final NOTIFY Message to a Server

This example shows a final NOTIFY message to a server.

```
*Apr 19 08:39:02.593: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.7.102.35:5060
From: <sip:user@10.7.104.88>;tag=1D80E90-2072
```



```
To: <sip:10.7.102.35>;tag=1C24D44-20FD
Date: Wed, 19 Apr 2000 08:39:02 UTC
Call-ID: C4BB7610-150411D4-802186E3-AD119804
CSeq: 103 NOTIFY
Timestamp: 956133542
Event: message-summary
Content-Length: 0
```

SIP Stack Portability

The SIP Stack Portability feature implements the following capabilities to the Cisco IOS SIP gateway stack:

- It receives inbound Refer message requests both within a dialog and outside of an existing dialog from the user agents (UAs).
- It sends and receives SUBSCRIBE or NOTIFY message requests via UAs.
- It receives unsolicited NOTIFY message requests without having to subscribe to the event that was generated by the NOTIFY message request.
- It supports outbound delayed media.

It sends an INVITE message request without Session Description Protocol (SDP) and provides SDP information in either the PRACK or ACK message request for both initial call establishment and mid-call re-INVITE message requests.

- It sets SIP headers and content body in requests and responses.

The stack applies certain rules and restrictions for a subset of headers and for some content types (such as SDP) to protect the integrity of the stack's functionality and to maintain backward compatibility. When receiving SIP message requests, it reads the SIP header and any attached body without any restrictions.

To make the best use of SIP call-transfer features, you should understand the following concepts:

- [SIP Call-Transfer Basics, page 41](#)
- [SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications, page 52](#)
- [SUBSCRIBE or NOTIFY Message Request Support, page 58](#)
- [SIP NOTIFY-Based Out-of-Band DTMF Relay, page 58](#)
- [Support for RFC 3312—QoS, page 60](#)
- [Support for the Achieving SIP RFC Compliance Feature, page 62](#)
- [Enhanced Redirect Handling, page 63](#)
- [Diversion Header Draft 06 Compliance, page 63](#)

SIP Call-Transfer Basics

This section contains the following information:

- [Basic Terminology of SIP Call Transfer, page 42](#)
- [Types of SIP Call Transfer Using the Refer Message Request, page 44](#)

Basic Terminology of SIP Call Transfer

Call transfer allows a wide variety of decentralized multiparty call operations. These decentralized call operations form the basis for third-party call control, and thus are important features for 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.

Refer Message Request

The SIP Refer message request provides call-transfer capabilities to supplement the SIP BYE and ALSO message requests already implemented on Cisco IOS SIP gateways. The Refer message request 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.



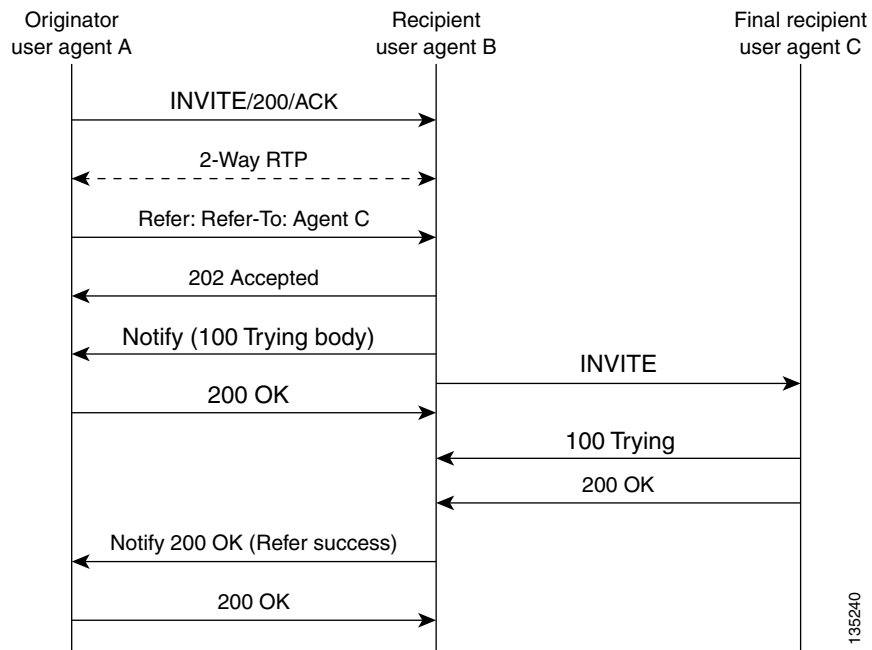
Note

A gateway can be a recipient or final recipient, but not an originator.

The Refer message request 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 contacted or not. The notification is accomplished using the SIP NOTIFY message request, SIP's event notification mechanism. A NOTIFY message with a message body of SIP 200 OK indicates a successful transfer, and a message 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.

Figure 18 represents the call flow of a successful Refer transaction initiated within the context of an existing call.

Figure 18 Successful Refer transaction



Refer-To Header

The recipient receives from the originator a Refer request that always contains a single Refer-To header. The Refer-To header includes a SIP URL that indicates the party to be invited and must be in SIP URL format.



Note

The TEL URL format cannot be used in a Refer-To header, because it does not provide a host portion, and without one, the triggered INVITE request cannot be routed.

The Refer-To header may contain three additional overloaded headers to form the triggered INVITE request. If any of these three headers are present, they are included in the triggered INVITE request. The three headers are:

- **Accept-Contact**—Optional in a Refer request. A SIP Cisco IOS gateway that receives an INVITE request with an Accept-Contact does not act upon this header. This header is defined in draft-ietf-sip-callerprefs-03.txt and may be used by user agents that support caller preferences.
- **Proxy-Authorization**—Nonstandard header that SIP gateways do not act on. It is echoed in the triggered INVITE request because proxies occasionally require it for billing purposes.
- **Replaces**—Header used by SIP gateways to indicate whether the originator of the Refer request is requesting a blind or attended transfer. It is required if the originator is performing an attended transfer, and not required for a blind transfer.

All other headers present in the Refer-To are ignored, and are not sent in the triggered INVITE.



Note

The Refer-To and Contact headers are required in the Refer request. The absence of these headers results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Refer-To header. Multiple Refer-To headers result in a 4xx class response.

Referred-By Header

The Referred-By header is required in a Refer request. It identifies the originator and may also contain a signature (included for security purposes). SIP gateways echo the contents of the Referred-By header in the triggered INVITE request, but on receiving an INVITE request with this header, gateways do not act on it.

**Note**

The Referred-By header is required in a Refer request. The absence of this header results in a 4xx class response to the Refer request. Also, the Refer request must contain exactly one Referred-By header. Multiple Referred-By headers result in a 4xx class response.

NOTIFY Message Request

Once 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 contacted or not. The notification is accomplished using the NOTIFY message request, 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.

The NOTIFY message must do the following:

- Reflect the same To, From, and Call-ID headers that were received in the Refer request.
- Contain an Event header refer.
- Contain a message body with a SIP response line. For example: SIP/2.0 200 OK to report a successful Refer transaction, or SIP/2.0 503 Service Unavailable to report a failure. To report that the recipient disconnected before the transfer finished, it must use SIP/2.0 487 Request Canceled.

Two Cisco IOS commands pertain to the NOTIFY message request:

- The **timers notify** command sets the amount of time that the recipient should wait before retransmitting a NOTIFY message to the originator.
- The **retry notify** command configures the number of times a NOTIFY message is retransmitted to the originator.

**Note**

For information on these commands, see the [Cisco IOS Voice Command Reference](#).

Types of SIP Call Transfer Using the Refer Message Request

This section discusses how the Refer message request facilitates call transfer.

There are two types of call transfer: blind and attended. The primary difference between the two is that the Replaces header is used in attended call transfers. The Replaces header is interpreted by the final recipient and contains a Call-ID header, indicating that the initial call leg is to be replaced with the incoming INVITE request.

As outlined in the Refer message request, there are 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.

A gateway can be a recipient or final recipient, but not an originator.

Blind Call-Transfer Process

A blind, or unattended, transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. This is different from a consultative, or attended, transfer in which one of the transferring parties either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. Blind transfers are often preferred by automated devices that do not have the capability to make consultation calls.

Blind transfer works as described in the [“Refer Message Request” section on page 42](#). The process is as follows:

1. Originator (user agent that initiates the transfer or Refer request) does the following:
 - a. Sets up a call with recipient (user agent that receives the Refer request)
 - b. Issues a Refer request to recipient
2. Recipient does the following:
 - a. Sends an INVITE request to final recipient (user agent introduced into a call with the recipient)
 - b. Returns a SIP 202 (Accepted) response to originator
 - c. Notifies originator of the outcome of the Refer transaction—whether final recipient was successfully (SIP 200 OK) contacted or not (SIP 503 Service Unavailable)
3. If successful, a call is established between recipient and final recipient.
4. The original signaling relationship between originator and recipient terminates when either of the following occurs:
 - One of the parties sends a Bye request.
 - Recipient sends a Bye request after successful transfer (if originator does not first send a Bye request after receiving an acknowledgment for the NOTIFY message).

[Figure 19](#) shows a successful blind or unattended call transfer in which the originator initiates a Bye request to terminate signaling with the recipient.

Figure 19 Successful Blind or Unattended Transfer—Originator Initiating a Bye Request

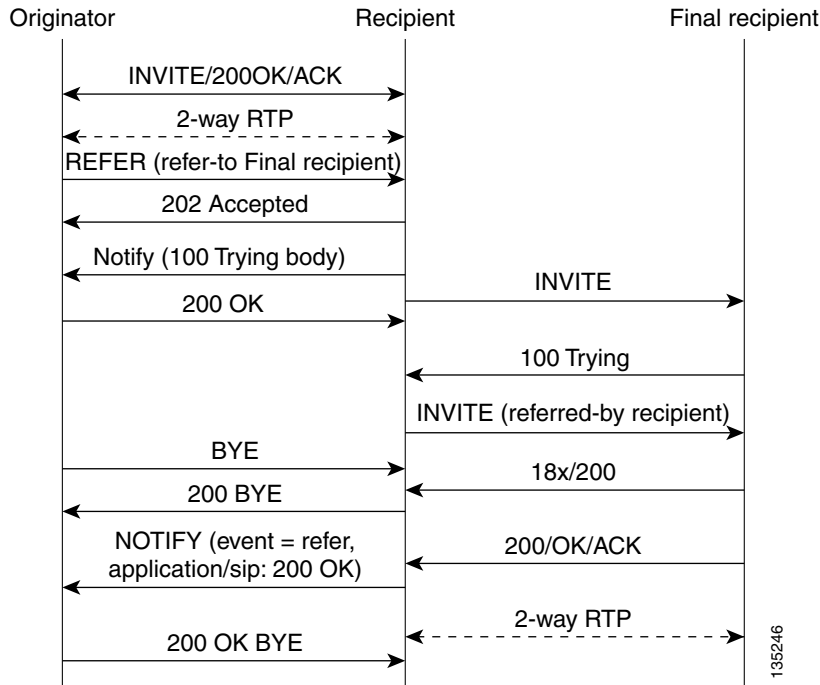
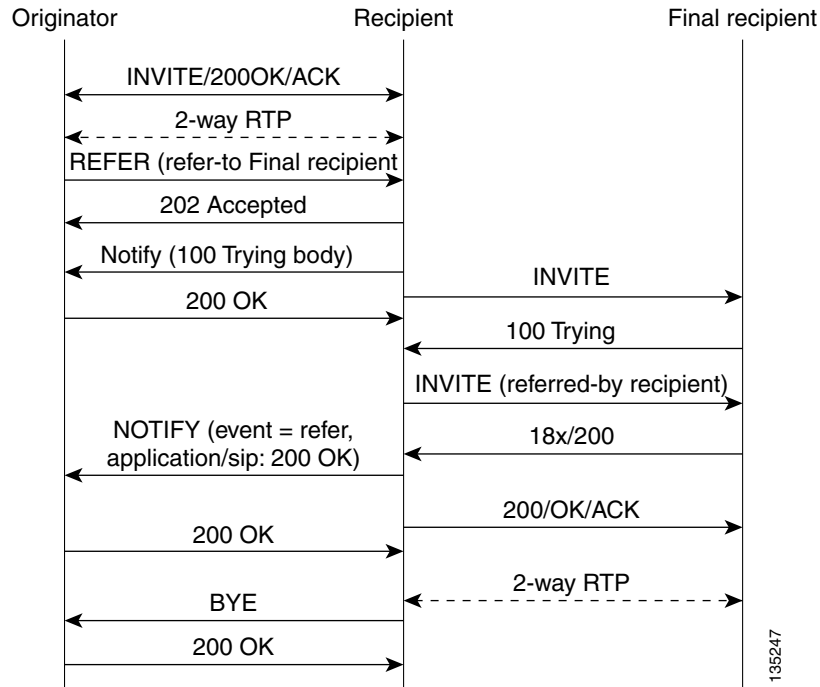


Figure 20 shows a successful blind or unattended call transfer in which the recipient initiates a Bye request to terminate signaling with the originator. A NOTIFY message is always sent by the recipient to the originator after the final outcome of the call is known.

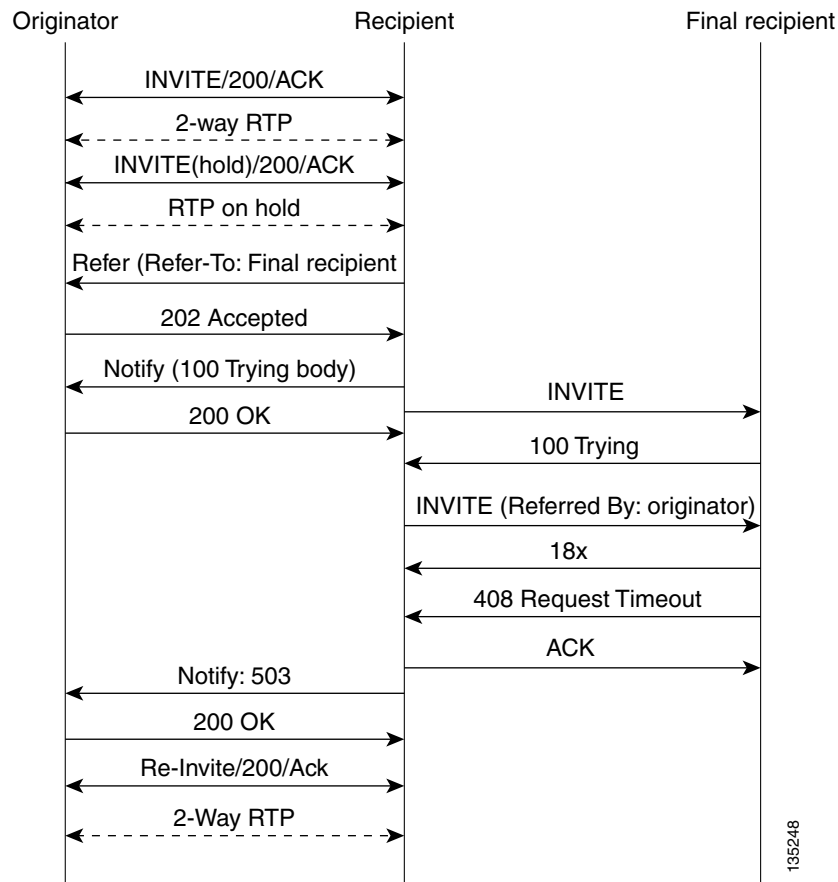
Figure 20 Successful Blind or Unattended Transfer—Recipient Initiating a Bye Request



If a failure occurs with the triggered INVITE to the final recipient, the call between originator and recipient is not disconnected. Rather, with blind transfer the process is as follows:

1. Originator sends a re-INVITE that takes the call off hold and returns to the original call with recipient.
2. Final recipient sends an 18x informational response to recipient.
3. The call fails; the originator cannot recover the call with recipient. Failure can be caused by an error condition or timeout.
4. The call leg between originator and recipient remains active (see [Figure 21](#)).
5. If the INVITE to final recipient fails (408 Request Timeout), the following occurs:
 - a. Recipient notifies originator of the failure with a NOTIFY message.
 - b. Originator sends a re-INVITE and returns to the original call with the recipient.

Figure 21 Failed Blind Transfer—Originator Returns to Original Call with Recipient



Attended Transfer

In attended transfers, the Replaces header is inserted by the initiator of the Refer message request as an overloaded header in the Refer-To and is copied into the triggered INVITE request sent to the final recipient. The header has no effect on the recipient, but is interpreted by the final recipient as a way to distinguish between blind transfer and attended transfer. The attended transfer process is as follows:

1. Originator does the following:

- a. Sets up a call with recipient.
 - b. Places recipient on hold.
 - c. Establishes a call to final recipient.
 - d. Sends recipient a Refer message request with an overloaded Replaces header in the Refer-To header.
2. Recipient does the following:
 - a. Sends a triggered INVITE request to final recipient. (Request includes the Replaces header, identifying the call leg between the originator and the final recipient.)
 - b. Recipient returns a SIP 202 (Accepted) response to originator. (Response acknowledges that the INVITE has been sent.)
 3. Final recipient establishes a direct signaling relationship with recipient. (Replaces header indicates that the initial call leg is to be shut down and replaced by the incoming INVITE request.)
 4. Recipient notifies originator of the outcome of the Refer transaction. (Outcome indicates whether or not the final recipient was successfully contacted.)
 5. Recipient terminates the session with originator by sending a Bye request.

Replaces Header

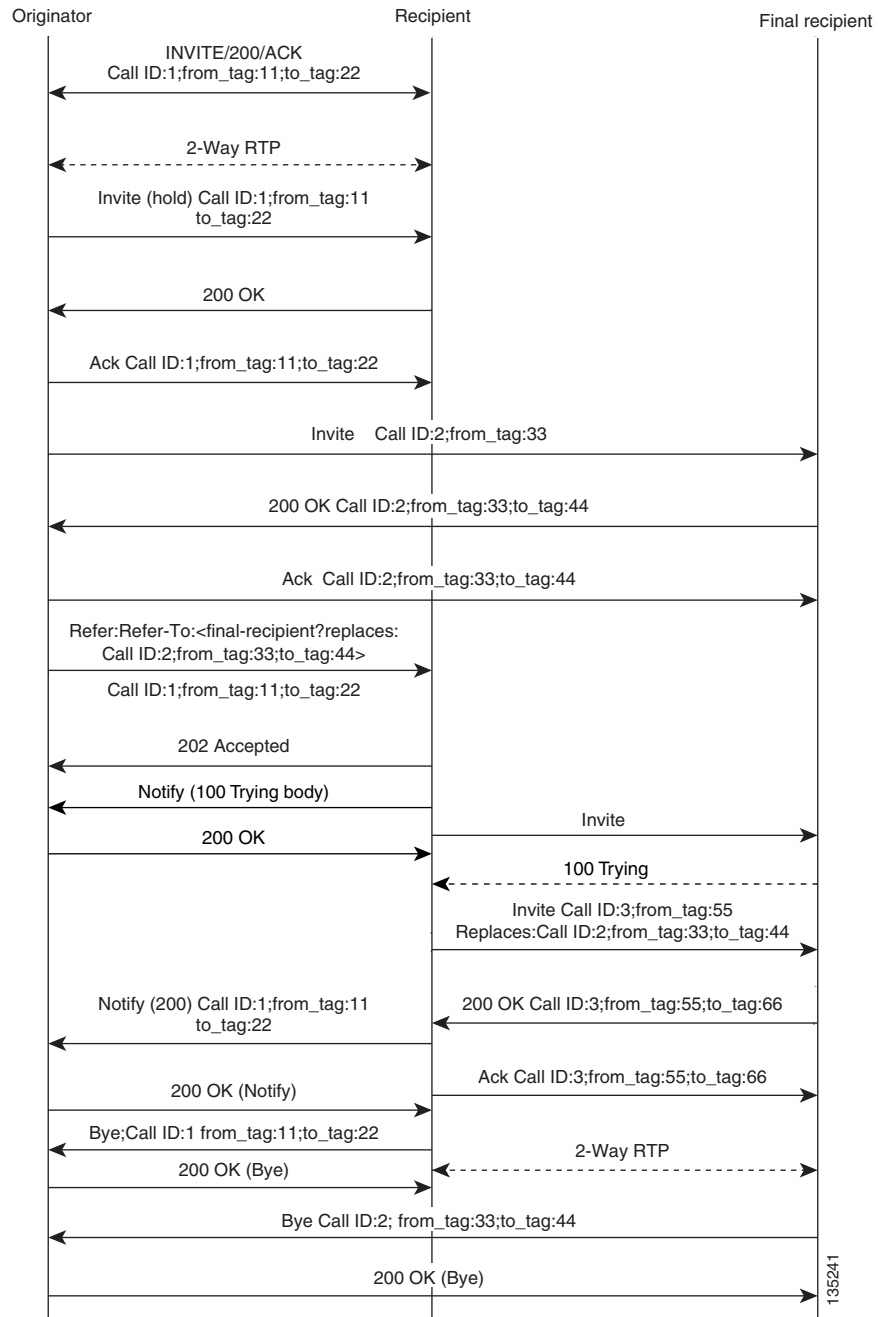
The Replaces header is required in attended transfers. It indicates to the final recipient that the initial call leg (identified by the Call-ID header and tags) is to be shut down and replaced by the incoming INVITE request. The final recipient sends a Bye request to the originator to terminate its session.

If the information provided by the Replaces header does not match an existing call leg, or if the information provided by the Replaces header matches a call leg but the call leg is not active (a Connect, 200 OK to the INVITE request has not been sent by the final-recipient), the triggered INVITE does not replace the initial call leg and the triggered INVITE request is processed normally.

Any failure resulting from the triggered INVITE request from the recipient to the final recipient does not drop the call between the originator and the final recipient. In these scenarios, all calls that are active (originator to recipient and originator to final recipient) remain active after the failed attended transfer attempt

Figure 22 shows a call flow for a successful attended transfer.

Figure 22 Successful Attended Transfer



Attended Transfer with Early Completion

Attended transfers allow the originator to have a call established between both the recipient and the final recipient. With attended transfer with early completion, the call between the originator and the final recipient does not have to be active, or in the talking state, before the originator can transfer it to the

recipient. The originator establishes a call with the recipient and only needs to be setting up a call with the final recipient. The final recipient may be ringing, but has not answered the call from the originator when it receives a re-INVITE to replace the call with the originator and the recipient.

The process for attended transfer with early completion is as follows (see [Figure 23](#)):

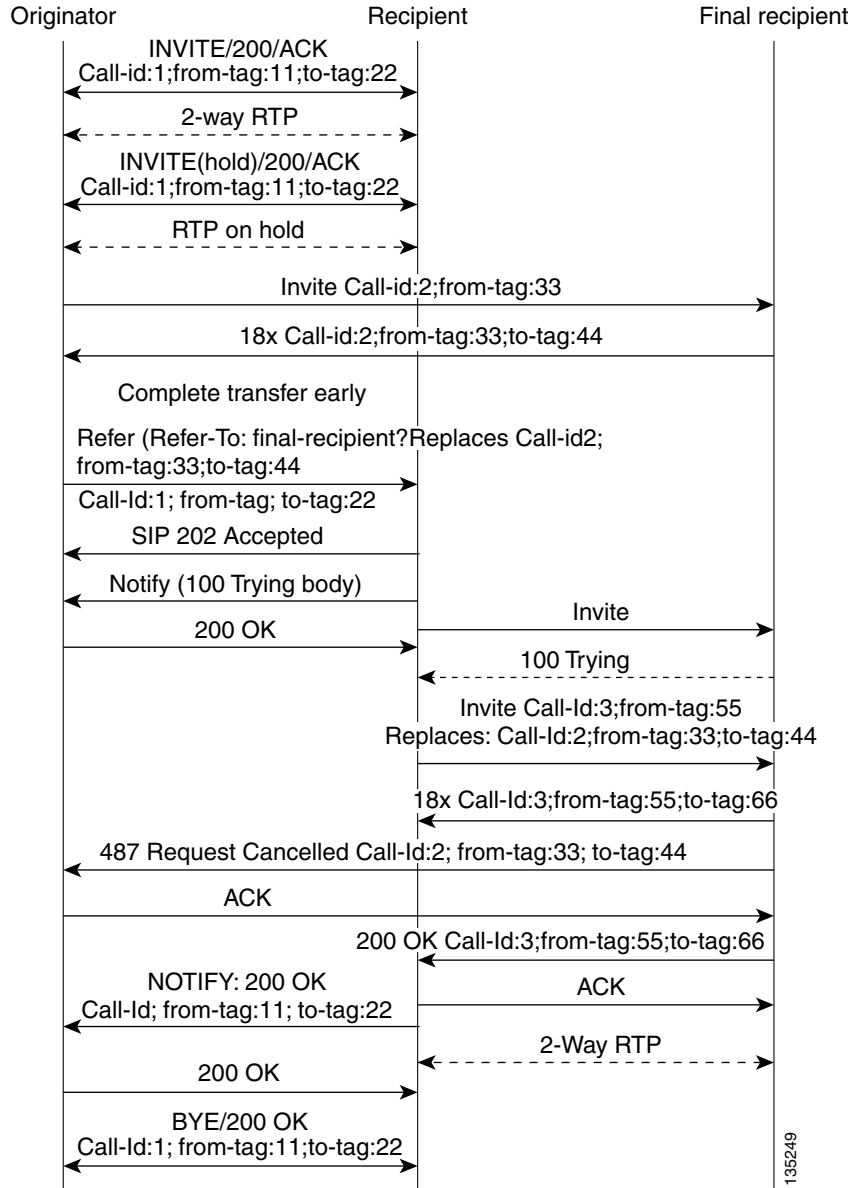
1. Originator does the following:
 - a. Sets up a call with recipient.
 - b. Places the recipient on hold.
 - c. Contacts the final recipient.
 - d. After receiving an indication that the final recipient is ringing, sends recipient a Refer message request with an overloaded Replaces header in the Refer-To header. (The Replaces header is required in attended transfers and distinguishes between blind transfer and attended transfers.)
2. Recipient does the following:
 - a. Returns a SIP 202 (Accepted) response to the originator. (to acknowledge that the INVITE has been sent.)
 - b. Upon receipt of the Refer message request, sends a triggered INVITE request to final recipient. (The request includes the Replaces header, which indicates that the initial call leg, as identified by the Call-ID header and tags, is to be shut down and replaced by the incoming INVITE request.)
3. Final recipient establishes a direct signaling relationship with recipient.
4. Final recipient tries to match the Call-ID header and the To or From tag in the Replaces header of the incoming INVITE with an active call leg in its call control block. If a matching active call leg is found, final recipient replies with the same status as the found call leg. However, it then terminates the found call leg with a 487 Request Cancelled response.



Note If early transfer is attempted and the call involves quality of service (QoS) or Resource Reservation Protocol (RSVP), the triggered INVITE from the recipient with the Replaces header is not processed and the transfer fails. The session between originator and final recipient remains unchanged.

5. Recipient notifies originator of the outcome of the Refer transaction—that is, whether final recipient was successfully contacted or not.
6. Recipient or originator terminates the session by sending a Bye request.

Figure 23 *Attended Transfer with Early Completion*



VSA for Call Transfer

You can use a vendor-specific attribute (VSA) for SIP call transfer.

Referred-By Header

For consistency with existing billing models, Referred-By and Requested-By headers are populated in call history tables as a VSA. Cisco VSAs are used for VoIP call authorization. The new VSA tag **supp-svc-xfer-by** helps to associate the call legs for call-detail-record (CDR) generation. The call legs can be originator-to-recipient or recipient-to-final-recipient.

The VSA tag **supp-svc-xfer-by** contains the user@host portion of the SIP URL of the Referred-By header for transfers performed with the Refer message request. For transfers performed with the Bye/Also message request, the tag contains user@host portion of the SIP URL of the Requested-By header. For each call on the gateway, two RADIUS records are generated: start and stop. The **supp-svc-xfer-by** VSA is generated only for stop records and is generated only on the recipient gateway—the gateway receiving the Refer or Bye/Also message.

The VSA is generated when a gateway that acts as a recipient receives a Refer or Bye/Also message with the Referred-By or Requested-By headers. There are usually two pairs of start and stop records. There is a start and stop record between the recipient and the originator and also between the recipient to final recipient. In the latter case, the VSA is generated between the recipient to the final recipient only.

Business Group Field

A new business group VSA field has been added that assists service providers with billing. The field allows service providers to add a proprietary header to call records. The VSA tag for business group ID is **cust-biz-grp-id** and is generated only for stop records. It is generated when the gateway receives an initial INVITE with a vendor dial-plan header to be used in call records. In cases when the gateway acts as a recipient, the VSA is populated in the stop records between the recipient and originator and the final recipient.



Note

For information on VSAs, see the [RADIUS VSA Voice Implementation Guide](#).

SIP Call Transfer and Call Forwarding Using Tcl IVR 2.0 and VoiceXML Applications

This section contains the following information about SIP Call Transfer and Call Forwarding with a Toolkit Command Language (Tcl) interactive-voice-response (IVR) or VoiceXML script:

- [SIP Call Transfer and Call Forwarding with a Tcl IVR Script, page 52](#)
- [Release Link Trunking on SIP Gateways, page 53](#)
- [SIP Gateway Initiation of Call Transfers, page 55](#)
- [SIP Call Forwarding, page 57](#)

SIP Call Transfer and Call Forwarding with a Tcl IVR Script

When using a Tcl IVR 2.0 application, you can implement SIP support of blind, or attended, call-transfer and call-forwarding requests from a Cisco IOS gateway. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. This is different from a consultative transfer in which one of the transferring parties either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. Blind transfers are often preferred by automated devices that do not have the capability to make consultation calls.

Before implementing blind transfer and call forwarding, you must write a custom Tcl IVR 2.0 script that implements call transfer and call forwarding. The script is responsible for receiving the hookflash event, providing dial tone, matching against the dial plan, initiating call transfer, and reestablishing the original call if the transfer attempt fails.



Note

For information on writing a Tcl IVR script, see the [Tcl IVR API Version 2.0 Programming Guide](#).

When the Tcl IVR script runs on the Cisco gateway, it can respond to requests to initiate blind call transfer (transfer without consultation) on a SIP call leg. SIP call forwarding on ephones (IP phones that are not configured on the gateway) is also supported.

**Note**

SIP call transfer and call forwarding are compliant with VoiceXML. VoiceXML scripts can also be used to implement call transfer and call forwarding.

Release Link Trunking on SIP Gateways

Release link trunking (RLT) functionality has been added to Cisco IOS SIP gateways. With RLT functionality, SIP call transfer now can be triggered by channel associated signaling (CAS) trunk signaling, which the custom Tcl IVR application can monitor. After a SIP call transfer has transpired and the CAS interface is no longer required, the CAS interface can be released.

The RLT functionality can be used to initiate blind transfers on SIP gateways. Blind call transfer uses the Refer message request. A full description of blind transfer and the Refer message request can be found in the [“Configuring SIP Call-Transfer Features”](#) chapter.

RLT and SIP Call Transfers

Call transfer can be triggered by CAS trunk signaling and then captured by the Tcl IVR script on a gateway. The process begins with the originator (the SIP user agent that initiates the transfer or Refer message request) responding with a dial tone once the originator receives the signal or hookflash from the PSTN call leg. The originator then prepares to receive dual-tone multifrequency (DTMF) digits that identify the final recipient (the user agent introduced into a call with the recipient).

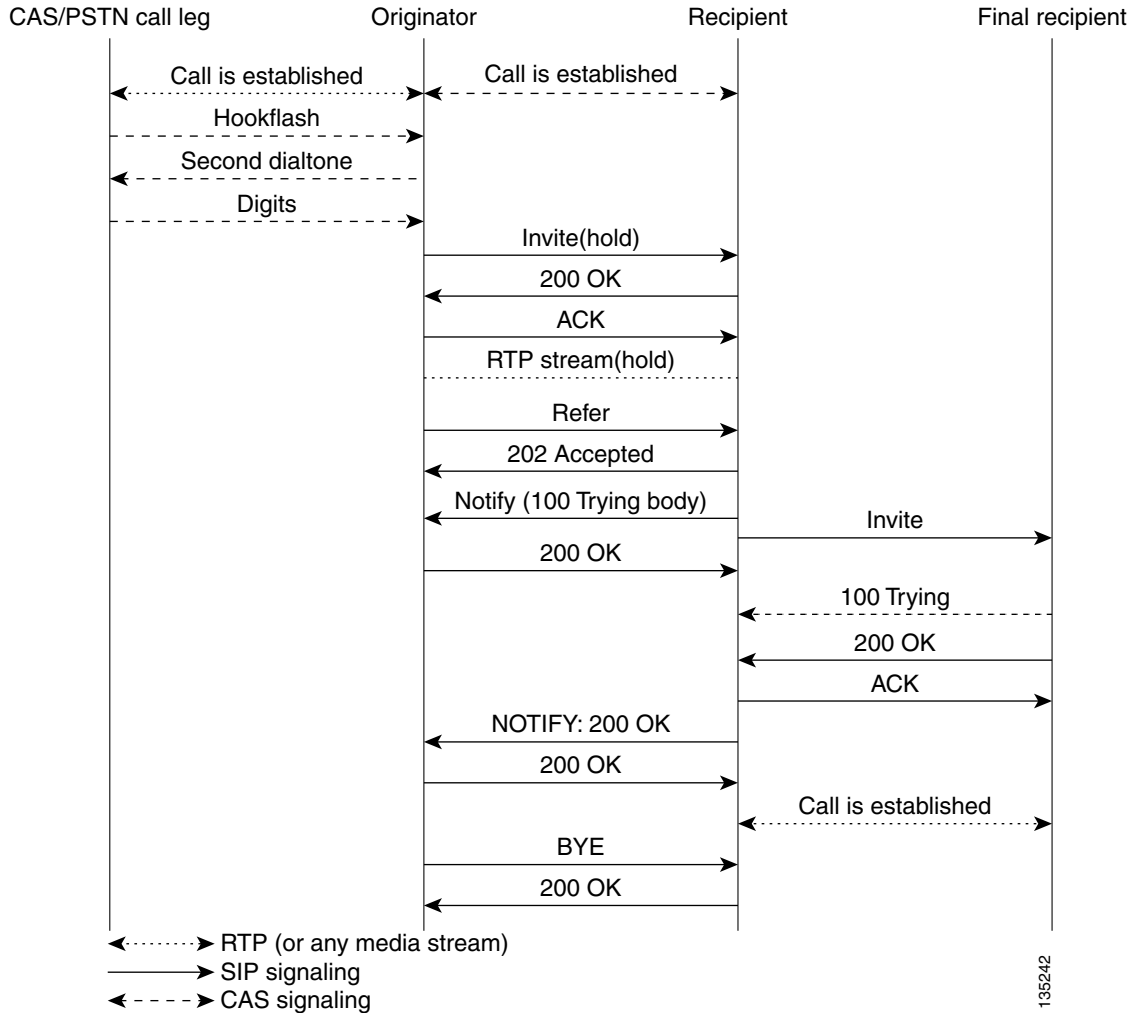
Once the first DTMF digit is received, the dial tone is discontinued. DTMF-digit collection is not completed until a 4-second interdigit timeout occurs, or an on-hook is received on that specific CAS time slot. Call transfer starts when DTMF-digit collection is successful. If digit collection fails, for example, if not enough DTMF digits or invalid digits are collected, the initial call is reestablished.

Once the DTMF digits are successfully collected, the Tcl IVR script can initiate call transfer. SIP messaging begins when the transfer is initiated with the Refer message request. The originator sends an INVITE to the recipient (the user agent that receives the Refer message request and is transferred to the final recipient) to hold the call and request that the recipient not return Real-Time Transport Protocol (RTP) packets to the originator. The originator then sends a SIP Refer message request to the recipient to start the transfer process. When the recipient receives the request, the recipient returns a 202 *Accepted* acknowledgment to the originator. The Tcl IVR script run by the originator can then release the CAS trunk and close the primary call. See [Figure 24](#).

If the recipient does not support the Refer message request, a 501 *Not implemented* message is returned. However, for backward compatibility purposes, the call transfer is automatically continued with the Bye/Also message request. The originator sends a Bye/Also request to the recipient and releases the CAS trunk with the PSTN call leg. The primary call between the originator and the recipient is closed when a 200 OK response is received.

In all other cases of call-transfer failures, the primary call between the originator and the recipient is immediately shut down.

Figure 24 Call Transfer Using the Refer Message Request



SIP and TEL URLs in Call Transfers

When the SIP call-transfer originator collects DTMF digits from the CAS trunk, it attempts to find a dial peer. If a dial peer is found, the session target in the dial peer is used to formulate a SIP URL. This URL can be used with both the Refer message request and the Bye/Also message request. A SIP URL is in the following form:

sip:JohnSmith@example.com

If a valid dial peer is not found, a Telephone Uniform Resource Locator (TEL URL) is formulated in the Refer-To header. A TEL URL is in the following form:

tel:+11235550100

The choice of which URL to use is critical when correctly routing SIP calls. For example, the originating gateway can send out a Bye with an Also header, but the Also header can carry only a SIP URL. The Also header cannot carry a TEL URL. That is, if the gateway decides to send a Bye/Also but cannot find a matched dial peer, the gateway reports an error on the transfer gateway and sends a Bye without the Also header.

If the recipient of a SIP call transfer is a SIP phone, the phone must have the capability to interpret either the Refer message request or the Bye/Also message request for the call transfer to work. If the recipient is a Cisco IOS gateway, there needs to be a matching dial peer for the Refer-To *user*. *User*, looking at the previous example, can be either *JohnSmith* or *11235550100*. The dial peer also needs to have an application session defined, where session can be the name of a Tcl IVR application. If there is no match, a 4xx error is sent back and no transfer occurs. If there is a POTS dial-peer match, a call is made to that POTS phone. Before Cisco IOS Release 12.2(15)T, if there is a VoIP match, the Refer-To URL is used to initiate a SIP call. In Release 12.2(15)T and later releases, the application session target in the dial peer is used for the SIP call.

SIP Gateway Initiation of Call Transfers

SIP gateways can also initiate, or originate, attended call transfers. The process begins when the originator establishes a call with the recipient. When the user on the PSTN call leg wants to transfer the call, the user uses hookflash to get a second dial tone and then enters the final recipient's number. The Tcl IVR script can then put the original call on hold and set up the call to the final-recipient, making the originator active with the final-recipient. The Refer message request is sent out when the user hangs up to transfer the call. The Refer message request contains a Replaces header that contains three tags: *SIP CallID*, *from*, and *to*. The tags are passed along in the INVITE from the recipient to the final recipient, giving the final recipient adequate information to replace the call leg. The host portion of the Refer message request is built from the established initial call. The following is an example of a Refer message request that contains a Replaces header:



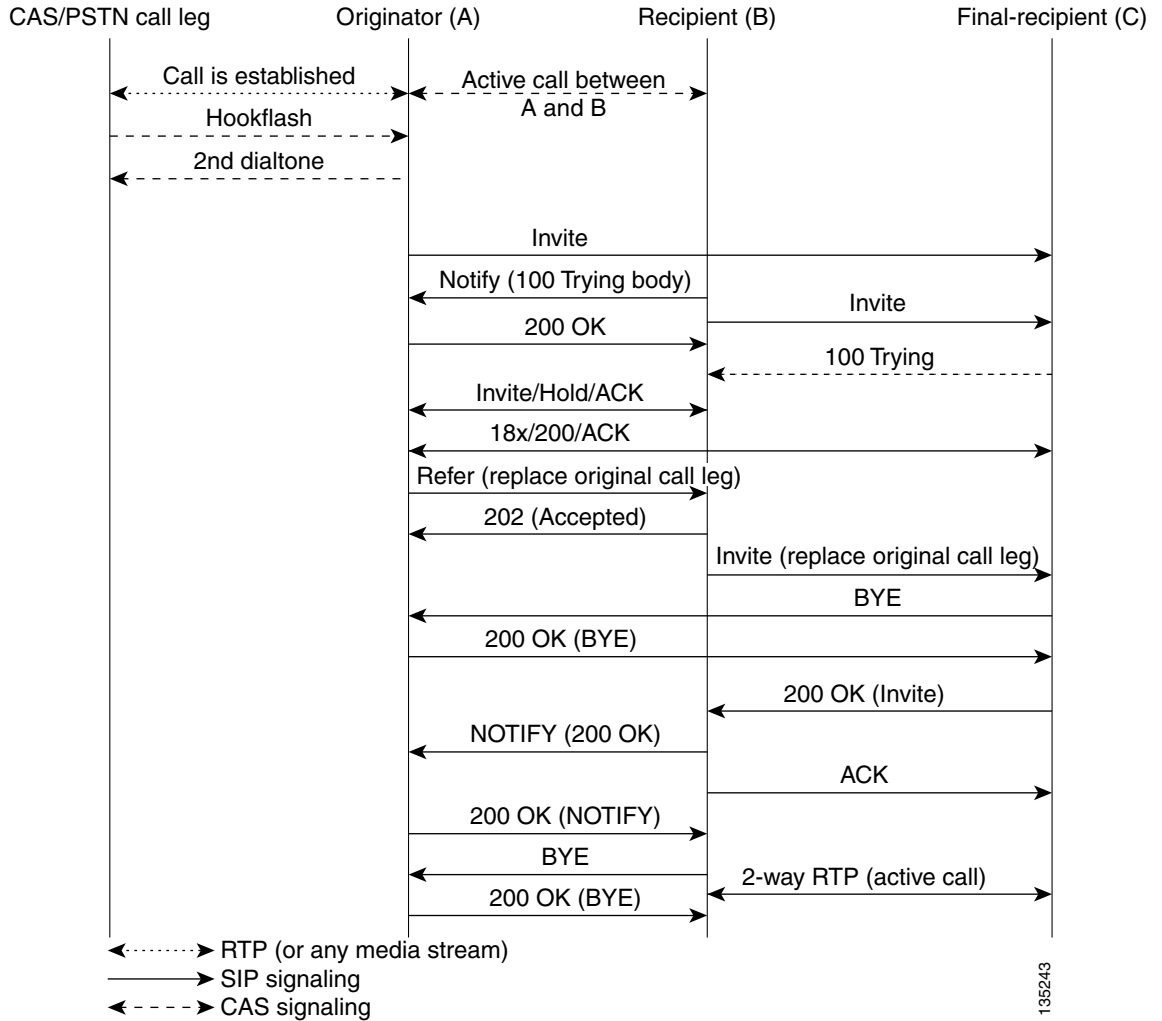
Note

IP addresses and host names in examples are fictitious.

```
Refer sip:5550100@172.16.190.100:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.16.190.99:5060
From: "5550101" <sip:5555555@172.16.190.187>
To: <sip:5550100@172.16.190.187>;tag=A7C2C-1E8C
Date: Sat, 01 Jan 2000 05:15:06 GMT
Call-ID: c2943000-106ae5-1c5f-3428@172.16.197.182
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 946685709
CSeq: 103 Refer
Refer-To:
sip:5550101@10.102.17.217?Replaces=DD713380-339C11CC-80BCF308-92BA812C@172.16.195.77;to-tag=A5438-23E4;from-tag=C9122EDB-2408
Referred-By: <sip:5550101@172.16.190.99>
Content-Length: 0
```

Once the NOTIFY is received by the originator, the Tcl IVR script can disconnect the call between the originator and the recipient. The call between the originator and the final recipient is disconnected by the recipient sending a BYE to the originator. See [Figure 25](#) for a call flow of a successful call transfer.

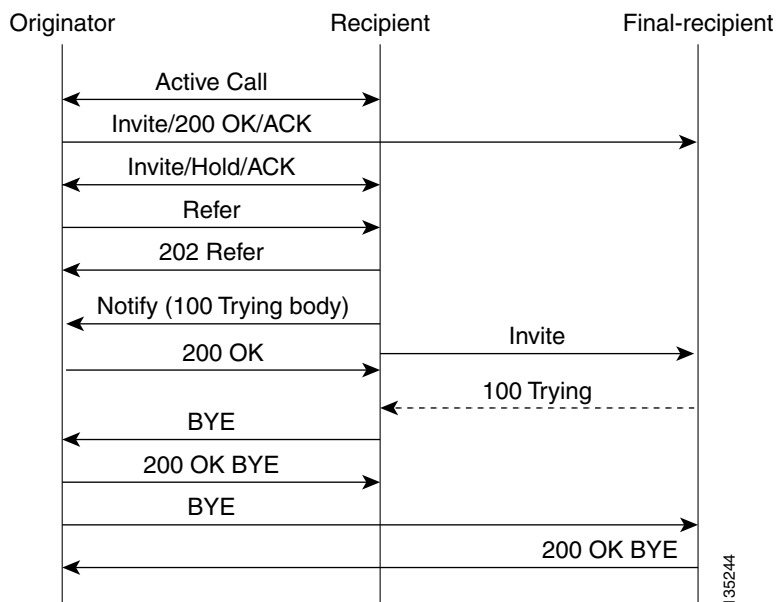
Figure 25 Successful Attended Call Transfer Initiated by the Originator



If the recipient does not support the Refer message request, a 501 *Not implemented* message is returned.

In all other cases of call-transfer failures, the primary call between the originator and the recipient is immediately shut down. Figure 26 shows the recipient hanging up the call before the transfer completes. The item to notice is that the NOTIFY message is never sent.

Figure 26 Unsuccessful Call Transfer—Recipient Hangs Up Before Transfer Completes



SIP Call Forwarding

SIP call forwarding is supported only on ephones—IP phones that are not configured on the gateway. Foreign exchange station (FXS), foreign exchange office (FXO), T1, E1, and CAS phones are not supported.

With ephones, four different types of SIP call forwarding are supported:

- Call Forward Unavailable
- Call Forward No Answer
- Call Forward Busy
- Call Forward Unconditional

In all four of these call-forwarding types, a 302 *Moved Temporarily* response is sent to the user-agent client. A Diversion header included in the 302 response indicates the type of forward.

The 302 response also includes a Contact header, which is generated by the calling number that is provided by the custom Tcl IVR script. The 302 response also includes the host portion found in the dial peer for that calling number. If the calling number cannot match a VoIP dial-peer or POTS dial-peer number, a 503 *Service Unavailable* message is sent, except in the case of the Call Forward No Answer. With Call Forward No Answer, call forwarding is ignored, the phone rings, and the expires timer clears the call if there is no answer.



Note

In Cisco IOS Release 12.2(15)T, when SIP with ephones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for ephones to initiate call forwarding.

SUBSCRIBE or NOTIFY Message Request Support

The Cisco IOS gateway accepts in dialog the SUBSCRIBE message requests with the same Call-Id and tags (to and from) for out-of-band (OOB) DTMF for Event header: telephone event. There can be an ID parameter in it, but the gateway supports in-dialog subscription for only one event. After the subscription is accepted, an initial NOTIFY message request is sent and includes a Subscription-State header as per RFC 3265.

When a digit is pressed on the PSTN end, the digit event is sent in the NOTIFY message requests. The Subscription-State header in these requests is active.

When the subscription expires before it is refreshed, the gateway terminates it by sending a NOTIFY message request with a Subscription-State header value set to terminated. The subscriber can always refresh the subscription by sending another SUBSCRIBE message request with the same Call-Id and tags as in the initial SUBSCRIBE message request.

If the INVITE message request dialog is terminated before the subscription expires, the subscription is terminated by sending a NOTIFY message request with a Subscription-State header value set to terminated. The gateway does not support generating in-dialog SUBSCRIBE message request.

SIP NOTIFY-Based Out-of-Band DTMF Relay

The Skinny Client Control Protocol (SCCP) IP phones do not support in-band DTMF digits; they are capable of sending only out-of-band DTMF digits. To support SCCP devices, originating and terminating SIP gateways can use Cisco-proprietary NOTIFY-based out-of-band DTMF relay. In addition, NOTIFY-based out-of-band DTMF relay can also be used by analog phones attached to analog voice ports (FXS) on the router.

NOTIFY-based out-of-band DTMF relay sends messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, NOTIFY-based out-of-band DTMF relay takes precedence.

The originating gateway sends an INVITE message with a SIP Call-Info header to indicate the use of NOTIFY-based out-of-band DTMF relay. The terminating gateway acknowledges the message with an 18x or 200 Response message, also using the Call-Info header. The Call-Info header for NOTIFY-based out-of-band relay appears as follows:

```
Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
```



Note

Duration is the interval between NOTIFY messages sent for a single digit and is set by means of the **notify telephone-event** command.

The NOTIFY-based out-of-band DTMF relay mechanism is negotiated by the SIP INVITE and 18x/200 Response messages. Then, when a DTMF event occurs, the gateway sends a SIP NOTIFY message for that event. In response, the gateway expects to receive a 200 OK message.

The NOTIFY-based out-of-band DTMF relay mechanism is similar to the DTMF message format described in RFC 2833. NOTIFY-based out-of-band DTMF relay consists of 4 bytes in a binary encoded format. The message format is shown in [Figure 27](#); [Table 8](#) describes the fields.

Figure 27 Message Format of NOTIFY-Based Out-of-Band DTMF Relay

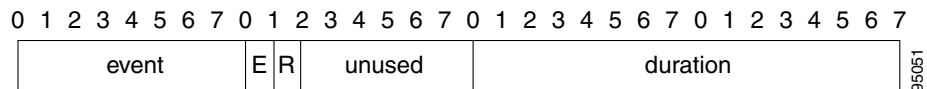


Table 8 Fields in NOTIFY-Based Out-of-Band DTMF relay Message

Field	Description
event	The DTMF event that is between 0—9, A, B, C, D, #, * and flash.
E	E signifies the end bit. If E is set to a value of 1, the NOTIFY message contains the end of the DTMF event. Thus, the duration parameter in this final NOTIFY message measures the complete duration of the event.
R	Reserved.
unused	In RFC 2833, unused corresponds to the volume field, but is not used in NOTIFY-based out-of-band DTMF relay.
duration	Duration of this DTMF event, in milliseconds.

Sending NOTIFY Messages

As soon as the DTMF event is recognized, the gateway sends out an initial NOTIFY message for this event with the duration negotiated in the Call-Info header of the SIP INVITE. For the initial NOTIFY message, the end bit is set to zero. Afterward, one of the following actions can occur:

- If the duration of the DTMF event is less than the negotiated duration, the originating gateway sends an end NOTIFY message for this event with the duration field containing the exact duration of the event and the end bit set to 1.
- If the duration of the DTMF event is greater than the negotiated duration, the originating gateway sends another NOTIFY message for this event after the initial timer runs out. The updated NOTIFY message has a duration of twice the negotiated duration. The end bit is set to 0 because the event is not yet over. If the event lasts beyond the duration specified in the first updated NOTIFY message, another updated NOTIFY message is sent with three times the negotiated duration.
- If the duration of the DTMF event is exactly the negotiated duration, either of the preceding two actions occurs, depending on whether the end of the DTMF event occurred before or after the timer ran out.

For example, if the negotiated duration is 600 ms, as soon as a DTMF event occurs, the initial NOTIFY message is sent with duration as 600 ms. Then a timer starts for this duration.

- If the DTMF event lasts only 300 ms, the timer stops and an end NOTIFY message is sent with the duration as 300 ms.
- If the DTMF event lasts longer than 600 ms (such as 1000 ms), when the timer expires an updated NOTIFY message is sent with the duration as 1200 ms and the timer restarts. When the DTMF event ends, an end NOTIFY message is sent with the duration set to 1000 ms.

Every DTMF event corresponds to at least two NOTIFY message requests: an initial NOTIFY message and an end NOTIFY message. There might also be some update NOTIFY message requests involved, if the total duration of the event is greater than the negotiated max-duration interval. Because DTMF events generally last for less than 1000 ms, setting the duration using the **notify telephone-event** command to more than 1000 ms reduces the total number of NOTIFY messages sent. The default value of the **notify telephone-event** command is 2000 ms.

Receiving NOTIFY Messages

Once a NOTIFY message is received by the terminating gateway, the DTMF tone plays and a timer is set for the value in the duration field. Afterward, one of the following actions can occur:

- If an end NOTIFY message for a DTMF event is received, the tone stops.
- If an update is received, the timer is updated according to the duration field.
- If an update or end NOTIFY message is not received before the timer expires, the tone stops and all subsequent NOTIFY messages for the same DTMF event or DTMF digit are ignored until an end NOTIFY message is received.
- If a NOTIFY message for a different DTMF event is received before an end NOTIFY message for the current DTMF event is received (which is an unlikely case), the current tone stops and the new tone plays. This is an unlikely case because for every DTMF event there needs to be an end NOTIFY message, and unless this is successfully sent and a 200 OK is received, the gateway cannot send other NOTIFY messages.



Note

In-band tones are not passed while NOTIFY-based out-of-band DTMF relay is used as the DTMF relay message request.

Two commands allow you to enable or disable NOTIFY-based out-of-band DTMF relay on a dial peer. The functionality is advertised to the other end using INVITE messages if it is enabled by the commands, and must be configured on both the originating and terminating SIP gateways. A third command allows you to verify DTMF relay status:

- **dtmf-relay (VoIP)**
- **notify telephone-event**
- **show sip-ua status**

The NOTIFY message request has a Subscription-State header per RFC 3265. Refer to the [“Configuring SIP DTMF Features”](#) module for additional information that relates to the DTMF feature.

Support for RFC 3312—QoS

This feature provides implementation on the gateway with suitable enhancements to the common stack to support quality of service (QoS) RSVP calls adhering to RFC 3312. This feature changes the existing implementation and follows RFC 3312 to provide QoS services using RSVP.

SIP Portable Stack Considerations for QoS

The portable SIP stack is unaware of the type of call (QoS or regular). All QoS-related information carried by SIP or SDP are passed by or to the application. The application takes the necessary steps to distinguish the type of call and handle it accordingly, transparent to the portable SIP stack. From the portable SIP stack’s perspective, the call flow for establishing a QoS call is similar to that of a non-QoS call. The only additions to the portable SIP stack application for establishing or modifying QoS calls are as follows:

- Ability to send the UPDATE message request
- Support for initiating and handling 183 and PRACK message request for midcall INVITE message requests

Behavior for QoS with RFC 3312 for Cisco IOS Gateways

The following lists the behavior that SIP QoS calls exhibits on Cisco IOS gateways, with RFC 3312 complaint stack as opposed to existing ICEE implementation:

- The QoS information is conveyed and confirmed through the following set of SDP attributes as proposed by RFC 3312.

Current Status—This attribute carries the current status of the network resources for a particular media stream in either offer or answer SDP. The gateways generates the following values depending on the state of the reservation.

Desired Status—This attribute states the preconditions for a particular media stream. For the Cisco IOS gateway the reservation is always applicable end-to-end status with resources reserved in either direction. The strength tag is configurable.

Confirmation Status—This attribute carries the information for the other gateway to notify back (using the UPDATE message request) once resource reservations are done on its end. On Cisco IOS gateways the originating gateways never request confirmation from the terminating gateway and if that fails, then the call is not presented and is terminated with a 580 (Precondition Failure) message response. The terminating gateway always asks for confirmation from the originating gateway when its reservations are done using the UPDATE message request. This is requested through the 183 message response for the INVITE message request.

- RFC 3312 requires the UA to use an UPDATE message request to notify the other gateway with the confirmation once the reservations are done on its end. The UPDATE message request transaction happens only if the received 183 message response contained the confirmation status attribute. The COMET message request is being used to convey that the reservations are met. With the RFC 3312 compliancy the COMET message request usage is obsolete.
- The originated INVITE message request contains the precondition option tag for use in Require and Supported header fields as in RFC 3312. With this the Content-Disposition header and Session=qos headers for QoS calls are no longer used.
- RFC 3312 suggests that the UA includes SDP (indicating QoS failure) in 580 Precondition Not Met message response. If a UAC does not make a QoS offer in the INVITE message request or gets a bad QoS offer in 18x or 2xx message response, then corresponding CANCEL or BYE message request contains an SDP body indicating QoS failure. This behavior is recommended but not mandatory as per RFC. This is kept as similar to existing implementation; 580, CANCEL, and BYE message requests continue to be sent, without SDP.
- RFC 3312 suggests the use of reliable provision responses (183/PRACK/200OK) for doing midcall QoS modifications. The current stack implementation uses the offer or answer model (Re-INVITE/200 OK/ACK) to do QoS modifications after the call is active. The new RFC recommendation for midcall does not give any advantage or extra functionality over the existing implementation. It complicates the midcall handling done by the stack. Midcall reliable provisional responses are not used by any other SIP feature, and there is no application that has an immediate need for this midcall functionality. Hence this feature continues using the existing stack's midcall INVITE offer/answer transaction for doing RSVP modifications for QoS calls.

Backward Compatibility

The QoS call flows are not backward compatible on Cisco IOS gateways. SIP continues to use existing RSVP Cisco IOS subsystem and its APIs but the SIP or SDP signalling involved is different from the existing implementation.

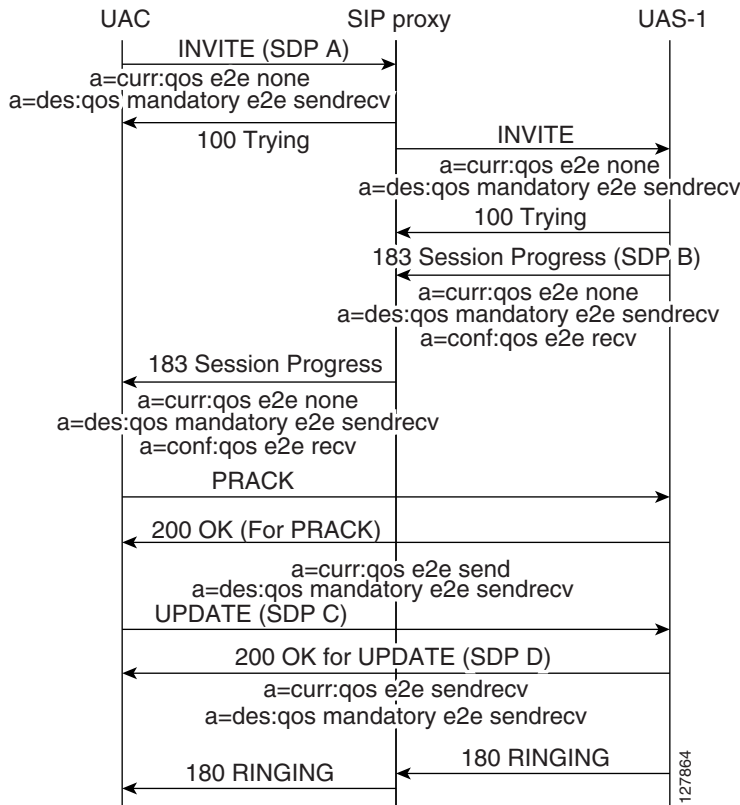
COMET Message Request Obsolescence

This feature stack obsoletes the usage (sending or receiving) of COMET message requests. This message request is replaced by UPDATE message request. This change has minor impact on the Call Admission Control feature on Cisco IOS gateways. QoS is the only other feature to use COMET. The CAC feature is using the UPDATE message request instead of COMET.

QoS Call Flow

The flows shown in [Figure 28](#) show a two-party call that invokes RSVP services to reserve resources before the called party is alerted. On the Cisco IOS gateway for this feature implementation the originating gateway does not need confirmation for INVITE message request preconditions. All the QoS SDP attributes shown are media-level attributes. If multiple media lines are associated with their own QoS attributes, then only the first media line QoS is honored.

Figure 28 Successful QoS Call Establishment



Support for the Achieving SIP RFC Compliance Feature

The Achieving SIP RFC Compliance feature enhances SIP gateway compliance for RFCs 3261, 3262, and 3264. This feature inherits these enhancements for the portable stack. Refer to the [“Achieving SIP RFC Compliance”](#) chapter for a description of introduced enhancements.

Enhanced Redirect Handling

The portable stack handles redirections (3xx or 485 message responses) internally. When a 3xx or 485 class message response is received by the SIP stack, the stack sends out a new INVITE message request to the contact in the 3xx message response, without notification to the application. In this feature, the functionality is opened up to the application. Upon receipt of a 3xx or 485 message response the application has the ability to take over the redirect response. When the application decides to handle the redirect, the SIP stack disconnects the original call that the 3xx or 485 message response received, and the application takes over responsibilities for setting up the new call.

Cisco IOS Behavior

There are no changes in the handling of redirects in Cisco IOS software. The stack continues to perform the redirections.

Diversion Header Draft 06 Compliance

This feature upgrades the Diversion header draft implementation to the draft-levy-diversion-06.txt version. This upgrade adds the capability to send or receive two new parameters in the Diversion header. The stack adds two new fields to set or pass this information to and from the application.



Note The draft-levy-diversion-06.txt version has since expired. Current standard uses History-Info header (refer to [RFC 4244, An Extension to the Session Initiation Protocol \(SIP\) for Request History Information](#)).

SIP: Domain Name Support in SIP Headers

The SIP: Domain Name Support in SIP Headers feature adds a command line interface (CLI) switch to provide a host or domain name in the host portion of the locally generated Session Initiation Protocol (SIP) headers (for example, From, RPID, and Call-ID). This feature also affects outgoing dialog-initiating SIP requests (for example, INVITE and SUBSCRIBE message requests).

To configure this feature, you should understand the following concepts:

- [Call Active and History VoIP Records, page 63](#)
- [SIP Headers, page 64](#)
- [Sample SIP Header Messages, page 64](#)

Vendor-specific attribute (VSA) is introduced to generate information about the locally configured host or domain name in the accounting records generated by the gateway. For a complete list of VSA changes, see the [RADIUS VSA Voice Implementation Guide](#).

Call Active and History VoIP Records

Call active and history VoIP records present the local hostname. They have the following format:

```
#show call active voice
VOIP:
LocalHostname=example.com
```

These records are generated for calls created in the context of the INVITE message request.

SIP Headers

The CLI affects the host portion of the following SIP headers generated for an outbound VoIP call from the SIP gateway:

- **Call-ID**—The Call-ID header in the SIP messages has an existing format of unique-string@ipaddr. With the CLI, the Call-ID has a value in the form of unique-string@localhostname or unique-string@domain-name. The dialog initiating the SIP requests that are affected namely are the INVITE and SUBSCRIBE message requests.
- **From**—The From header in the following dialog initiating requests. The INVITE and SUBSCRIBE message requests originating from the gateway have host or domain name in the host portion of the SIP URI. When the CLI is configured, the Remote-Party-ID header also has a hostname in the host portion of the SIP URI. The Remote-Party-ID header is sent out in the INVITE and INFO message requests from the gateway.

Other SIP headers such as Contact and Via are not affected by configuring the new CLI. Those headers continue to have IP addresses even when the CLI is configured.

These changes do *not* affect the Session Definition Protocol (SDP).

SIP headers that are provided by the application to SIP via header passing mechanisms always override headers generated by SIP.

Sample SIP Header Messages

This section contains the following sample SIP header messages with the SIP: Domain Name Support in SIP Headers feature disabled and enabled:

- [Feature Disabled—INVITE Message Request Sent from the Gateway, page 64](#)
- [Feature Enabled—INVITE Message Request Sent from the Gateway, page 65](#)

Feature Disabled—INVITE Message Request Sent from the Gateway

```
Sent:
INVITE sip:9002@example.sip.com:5060 SIP/2.0
Via:SIP/2.0/TCP 172.18.195.49;branch=z9hG4bK597
Remote-Party-ID:<sip:9001@172.18.195.49>;party=calling;screen=no;privacy=off
From:<sip:9001@172.18.195.49>;tag=3AA7574-11BA
To:<sip:9002@example.sip.com>
Date:Tue, 31 Aug 2004 13:40:57 GMT
Call-ID:3924408D-FA8A11D8-80208D32-72E3122E@172.18.195.49
Supported:100rel,timer,resource-priority
Min-SE:1800
Cisco-Guid:940277299-4203352536-2149420338-1927483950
User-Agent:Cisco-SIPGateway/IOS-12.x
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, Refer , SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq:101 INVITE
Max-Forwards:70
Timestamp:1093959657
Contact:<sip:9001@172.18.195.49:5060;transport=tcp>
Expires:180
Allow-Events:telephone-event
Content-Type:multipart/mixed;boundary=uniqueBoundary
Mime-Version:1.0
Content-Length:418

--uniqueBoundary
Content-Type:application/sdp
Content-Disposition:session;handling=required
```



```

v=0
o=CiscoSystemsSIP-GW-UserAgent 4780 5715 IN IP4 172.18.195.49
s=SIP Call
c=IN IP4 172.18.195.49
t=0 0
m=audio 18336 RTP/AVP 18 101 19
c=IN IP4 172.18.195.49
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20

--uniqueBoundary--

```

Feature Enabled—INVITE Message Request Sent from the Gateway

```

Sent:
INVITE sip:9002@example.sip.com:5060 SIP/2.0
Via:SIP/2.0/TCP 172.18.195.49;branch=z9hG4bK22C7
Remote-Party-ID:<sip:9001@gw11.example.com>;party=calling;screen=no;privacy=off
From:<sip:9001@gw11.example.com>;tag=39CF740-FFC
To:<sip:9002@example.sip.com>
Date:Tue, 31 Aug 2004 13:26:13 GMT
Call-ID:2A101AD3-FA8811D8-801C8D32-72E3122E@gw11.example.com
Supported:100rel,timer,resource-priority
Min-SE:1800
Cisco-Guid:686218050-4203221464-2149158194-1927483950
User-Agent:Cisco-SIPGateway/IOS-12.x
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, Refer , SUBSCRIBE, NOTIFY, INFO,
REGISTER
CSeq:101 INVITE
Max-Forwards:70
Timestamp:1093958773
Contact:<sip:9001@172.18.195.49:5060;transport=tcp>
Expires:180
Allow-Events:telephone-event
Content-Type:multipart/mixed;boundary=uniqueBoundary
Mime-Version:1.0
Content-Length:418

--uniqueBoundary
Content-Type:application/sdp
Content-Disposition:session;handling=required

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 5250 7833 IN IP4 172.18.195.49
s=SIP Call
c=IN IP4 172.18.195.49
t=0 0
m=audio 18998 RTP/AVP 18 101 19
c=IN IP4 172.18.195.49
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20

--uniqueBoundary--

```

SIP Gateway Support for SDP Session Information and Permit Hostname CLI

The SIP GW Support for SDP Session Information and Permit Hostname CLI Feature adds support to Cisco IOS SIP gateways for both SDP session information and validation of hostnames in initial INVITE requests. These features are described in the following sections:

- [SDP Changes for Session Information Line, page 27](#)
- [Validating Hostname in Initial INVITE Request URI, page 28](#)

SDP Changes for Session Information Line

The SDP Session Information line can exist multiple times within a session description. The line, represented by “i=” in the SDP, can be present at the session-level as well as the media-level. You can have only one session description per packet. The session description contains one session-level, but can have multiple media-levels.

The following is a sample SDP description. The highlighted lines represent the updates to reflect RFC 2327:

```
v=0
o=mhandley 2890844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
a=recvonly
m=audio 49170 RTP/AVP 0
i=media-information 1
m=video 51372 RTP/AVP 31
i=media-information 2
m=application 32416 udp wb
a=orient:portrait
```

The session information is optional, therefore internal structures are not built to expect this parameter. Specifically, internal memory is only allocated for this parameter when it is present in SDP, or when the application specifies that it be built into an outgoing message. In order to protect the internal operation of the Cisco IOS gateway, the maximum allowable length of a received session information line is 1000 characters. Session information lines over 1000 characters are truncated.

While the RFC detailing SDP indicates to only expect one session information line at the appropriate level, the Cisco IOS gateway will not “drop” the SDP in the event that this rule is violated. In the event that multiple “i=” lines are received at a particular level, the first parsed line that contains data is stored. All subsequent lines for that level are dropped.

Validating Hostname in Initial INVITE Request URI

Beginning with Cisco IOS Software Release 12.4(9)T, administrators can validate hostnames of incoming initial INVITE messages. When the gateway processes an initial INVITE, a determination is made whether or not the host portion is in ipv4 format or a domain name.

If the host portion is an IP address, its IP address is compared with the interfaces on the gateway. If a match is found, the INVITE is processed as normal. If there is not a match, the gateway sends a **400 Bad Request - 'Invalid IP Address'** message.

If the initial INVITE has a domain name in the host of the request URI, the gateway checks this domain name against a list of configured hostnames. If you configure no hostnames, existing behavior executes and the INVITE is processed. If you configure hostnames for this gateway, the gateway compares the host name in the request URI to the configured hostname list. If a match is found, the INVITE is processed as normal. If there is not a match, the gateway sends a **400 Bad Request - 'Invalid host'** message.

You can configure up to 10 hostnames by re-entering the **permit hostname dns** command. Use the **no** form of this command to remove any configured hostnames.

The following example shows a configured list of hostnames. The highlighted lines represent the updates to reflect RFC 2327.

```

sip-ua
retry invite 1
registrar ipv4:172.18.193.97 expires 3600
permit hostname dns:sinise.sip.com
permit hostname dns:liotta.sip.com
permit hostname dns:sipgw.sip.com
permit hostname dns:yourgw.sip.com
permit hostname dns:csp.sip.com
!

```

The following example shows an initial INVITE message with a hostname. The highlighted line represents the updates to reflect RFC 2327.

```

INVITE sip:777@sinise.sip.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.201.173:5060;branch=z9hG4bK2C419
To: <sip:777@172.18.197.154>
From: <sip:333@64.102.17.246>;tag=B87C0-B65
Date: Thu, 23 Feb 2006 16:49:26 GMT
Call-ID: 4EAF670B-A3C311DA-80148B65-6E225A8E@172.18.197.154
Contact: <sip:333@172.18.201.173>
Supported: 100rel, eatit
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE, NOTIFY, INFO
Max-Forwards: 70
Cseq: 104 INVITE
Expires: 60
Timestamp: 730947404
Content-Length: 211
Content-Type: application/sdp
^M
v=0
o=CiscoSystemsSIP-GW-UserAgent 6109 4520 IN IP4 172.18.201.173
s=SIP Call
c=IN IP4 111.11.111.111
t=0 0
m=audio 16880 RTP/AVP 0 19
c=IN IP4 111.11.111.111
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20

```

Outbound Proxy Support for the SIP Gateway

The Outbound Proxy Support for the SIP Gateway feature allows you to configure an outbound proxy server on a SIP gateway. You can use the **outbound-proxy** command to globally configure a SIP gateway to send all dialog initiating requests (such as INVITE, SUBSCRIBE, and REGISTER) to a specified destination. You can also use the **voice-class sip outbound-proxy** command to configure these settings on an individual dial peer, overriding the global gateway settings (refer to the [Cisco IOS Voice Command Reference](#)).

The request-uri of these dialog initiating requests are extracted from the session-target and does not reflect that the request is sent to a configured outbound-proxy server. The outbound-proxy server, based on the host in the request-uri, routes it accordingly. However, in some scenarios, it is possible that calls coming in over a SIP trunk to Cisco Unified Communications Manager Express (Cisco Unified CME) get forwarded to the outbound SIP proxy rather than directly to the phone. To correct this behavior, use the **outbound-proxy system** command to configure SIP line-side phones on a Cisco Unified CME (refer to the [Cisco Unified Communications Manager Express Command Reference](#)).

SIP: SIP Support for PAI

The SIP Support for PAI feature allows you to configure privacy headers into associated SIP request messages, as defined in RFC 3323 and RFC 3325. This feature introduces the **privacy and asserted-id commands which you can use to build various privacy-header requests into common SIP messages**, as shown in [Table 9](#).

Table 9 Privacy Header Request Options

Cisco IOS Command	SIP Message Header Options
privacy	ACK, BYE, CANCEL, INVITE, OPTIONS, SUBSCRIBE, NOTIFY, PRACK, INFO, and UPD
privacy pai	BYE, INVITE, OPTIONS, SUBSCRIBE, NOTIFY, and Refer
privacy ppi	BYE, INVITE, OPTIONS, SUBSCRIBE, NOTIFY, and Refer

SIP: History-info Header Support

The SIP History-info Header Support feature provides support for the history-info header in SIP INVITE messages only. The SIP gateway generates history information in the INVITE message for all forwarded and transferred calls. The history-info header records the call or dialog history. The receiving application uses the history-info header information to determine how and why the call has reached it.

The receiving application can use the call or dialog history to enhance services, such as calls to voice mail servers or sessions initiated to call centers from a click-to-talk SIP URL on a web page.

To configure the SIP History-info Header Support feature, you need to understand the following concepts:

- [Feature Design of SIP Accept-Language Header Support, page 11](#)

Feature Design of SIP History-info Header Support

Cisco implements this feature on SIP-TDM gateways and SIP-SIP Cisco Unified Border Element gateways by supporting the history-info header, as defined in RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*. The history-info header forms part of the SIP INVITE messages that establish media sessions between user agents, and the subsequent responses to the INVITE messages.

Support for history-info headers on a gateway is enabled using the **history-info** command. The system supports multiple history-info headers (up to a maximum of nine) for a single INVITE message. The headers are contained in a comma-separated list.

SIP History-info Header Support on SIP-TDM Gateways

When the TDM gateway sends an INVITE message, it creates the history-info header based on the request URI.

When the gateway receives a redirected PSTN call, it builds the history-info header using the redirect information provided by the PSTN source signaling address, the local host configuration (DNS name), and the host registrar.

To maintain the correct order and to record any redirection of a request, the header includes index information (as a series of dot-delimited digits). The index format is defined in RFC 4244 section 4.3.3.1.3.

If history-info headers are enabled for the SIP stage, the gateway sends both diversion headers and history-info headers in the outbound request. However, the history-info header takes preference when the gateway maps the header to the ISDN redirect number.

SIP History-info Header Support on SIP-SIP Cisco Unified Border Element Gateways

When the Cisco Unified Border Element gateway receives an inbound INVITE message without a history-info header, it generates the history-info header based on the request URI in the outbound INVITE message. If privacy is enabled on the gateway, then history is added to the privacy settings.

When the gateway receives an outbound message it creates the history-info header to the message based on the request URI. The maximum number of history-info headers supported by the gateway is nine. If the gateway receives a message with nine or more headers, it keeps the first eight messages only and adds the new header to the end of the header list.

When history-info privacy is configured on the gateway, it transparently passes all history-info and privacy headers in the message from one SIP stage to the next.

The gateway forwards history-info headers from one SIP stage to the next. If history-info headers are enabled for the SIP stage, the gateway behaves as follows:

- If no history-info header is present, the gateway converts the diversion headers to history-info headers and sets the *cause* parameter to 302. The gateway then sends both the diversion and the history-info headers.
- If no diversion headers are present, the gateway converts all the history-info headers where the *cause* parameter is set to 302 to diversion headers. The gateway then sends both the diversion and history-info headers.
- If both diversion headers and history-info headers are present, no conversion is performed.

If history-info headers are disabled for the SIP stage, the gateway sends all diversion headers (including any new diversion headers) to the next SIP stage.

How to Configure SIP Message, Timer, and Response Features

This section contains the following procedures:

- [Configuring Internal Cause Code Consistency Between SIP and H.323](#), page 70
- [Configuring SIP - Configurable PSTN Cause Code Mapping](#), page 71
- [Configuring SIP Accept-Language Header Support](#), page 74
- [Configuring SIP Enhanced 180 Provisional Response Handling](#), page 75
- [Configuring SIP Extensions for Caller Identity and Privacy](#), page 76
- [Configuring SIP INVITE Request with Malformed Via Header](#), page 81
- [Configuring Privacy Headers](#), page 81
- [Configuring SIP Session Timer Support](#), page 84
- [Configuring SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion](#), page 85
- [Configuring SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers](#), page 88
- [Configuring SIP Stack Portability](#), page 91
- [Configuring SIP: Domain Name Support in SIP Headers](#), page 91
- [Configuring SIP Gateway Support for Session Information](#), page 99
- [Configuring SIP Gateway Support for Permit Hostname CLI](#), page 100
- [Configuring Outbound Proxy Support for the SIP Gateway](#), page 101
- [Configuring SIP Support for PAI](#), page 104
- [Configuring SIP History-info Header Support](#), page 107
- [Verifying SIP Message Components, Session Timers, and Responses Configuration](#), page 110
- [Troubleshooting Tips for SIP Message, Timer, and Response Features](#), page 120

**Note**

-
- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP Message, Timer, and Response Features”](#) section on page 4
 - [“Restrictions for SIP Message, Timer, and Response Features”](#) section on page 4
 - For help with a procedure, see the verification and troubleshooting sections listed above.
-

Configuring Internal Cause Code Consistency Between SIP and H.323

To configure the Internal Cause Code Consistency Between SIP and H.323 feature, perform the following procedures.

- [Configure Internal Cause Code Consistency Between SIP and H.323](#), page 71 (optional)
- [Configuring SIP Enhanced 180 Provisional Response Handling](#), page 75

Configure Internal Cause Code Consistency Between SIP and H.323

The standard set of cause-code categories that is now generated for internal voice call failures is used by default. To configure internal failures with existing or nonstandard H.323 and SIP cause codes, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **cause-code legacy**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	cause-code legacy Example: Router(config-voi-srv)# cause-code legacy	Represents internal failures with existing or nonstandard H.323 or SIP cause codes. The keyword is as follows: <ul style="list-style-type: none"> • legacy—Sets the internal cause code to the former and nonstandard set of values. Used for backward compatibility.
Step 5	exit Example: Router(config-voi-srv)# exit	Exits the current mode.

Configuring SIP - Configurable PSTN Cause Code Mapping

To configure SIP - Configurable PSTN Cause Code Mapping, perform the following procedures.

- [Map PSTN Codes to SIP Status Codes, page 72](#) (optional)
- [Map SIP Status Codes to PSTN Cause Codes, page 73](#) (optional)

Map PSTN Codes to SIP Status Codes

To configure incoming PSTN cause codes to SIP status codes, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **set pstn-cause** *value* **sip-status** *value*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	set pstn-cause <i>value</i> sip-status <i>value</i> Example: Router(config-sip-ua)# set pstn-cause 111 sip-status 400	Use this command to map an incoming PSTN cause code to a SIP error status code. Keywords and arguments are as follows: <ul style="list-style-type: none"> • pstn-cause <i>value</i>—PSTN cause code. Range: 1 to 127. • sip-status <i>value</i>—SIP status code that is to correspond with the PSTN cause code. Range: 400 to 699.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Map SIP Status Codes to PSTN Cause Codes

To map incoming SIP status codes to PSTN cause codes, complete the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **set sip-status *value* pstn-cause *value***
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	set sip-status <i>value</i> pstn-cause <i>value</i> Example: Router(config-sip-ua)# set sip-status 400 pstn-cause 111	Maps an incoming PSTN cause code to a SIP error status code. Keywords and arguments are as follows: <ul style="list-style-type: none"> • sip-status <i>value</i>—SIP status code that is to correspond with the PSTN cause code. Range: 400 to 699. • pstn-cause <i>value</i>—PSTN cause code. Range: 1 to 127.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP Accept-Language Header Support

You can configure Accept-Language header support in two different configuration modes: voice service configuration mode and dial-peer voice configuration mode. The gateway first checks for languages configured under the dial-peer voice configuration mode and failing a match will then default to the global voice service configuration. If no languages are configured in either mode, then the header is not added.



Note

For the Accept-Language header to be included in the 200 OK response to an OPTIONS request, you must enable this feature in voice service configuration mode.

Perform this task to enable Accept-Language header support and specify languages carried in the Accept-Language header of SIP INVITE requests and OPTIONS responses.

SUMMARY STEPS

1. **enable**
 2. **configure terminal**
 3. **voice service pots**
- or
4. **dial-peer voice tag pots**
 5. **supported-language language-code language-param qvalue**
 6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode, or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service pots or dial-peer voice tag pots Example: Router(config)# voice service pots or Router(config)# dial-peer voice 1 pots	Enters global voice service configuration mode or dial-peer voice configuration mode. Note Voice service configuration mode configures the gateway to support the Accept-Language header in both outgoing SIP INVITE messages and OPTIONS responses. Dial-peer voice configuration mode configures it to support the header in INVITE messages only.

	Command or Action	Purpose
Step 4	<p>supported-language <i>language-code</i> language-param <i>qvalue</i></p> <p>Example: Router(conf-voi-serv)# supported-language EO language-param .25</p>	<p>Specifies languages carried in the Accept-Language header in outgoing SIP INVITE messages and OPTIONS responses. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • <i>language-code</i>—Any of 139 supported languages designated by a two-letter ISO-639 country code. The note below shows a partial list of supported language codes and languages. To display a complete listing, use the help command supported-language ? • language-param <i>qvalue</i>—Priority of the language, in descending order according to the assigned parameter value. You can assign a value for each language. Range: 0, a decimal fraction in the range 0.001 to 0.999, and 1. Default: 1 (highest priority).
Step 5	<p>exit</p> <p>Example: Router(conf-voi-serv)# exit</p>	<p>Exits the current mode.</p>



Note

The following is a partial list of supported language codes and languages. To display a complete listing, use the help command **supported-language ?**.

AR —Arabic	HE —Hebrew	ES —Spanish
ZH —Chinese	GA —Irish	SW —Swahili
EN —English	IT —Italian	SV —Swedish
EO —Esperanto	JA —Japanese	VI —Vietnamese
DE —German	KO —Korean	YI —Yiddish
EL —Greek	RU —Russian	ZU —Zulu

Configuring SIP Enhanced 180 Provisional Response Handling

This feature allows you to do the following:

- Enable or disable early media cut-through treatment for SIP 180 messages with SDP
- Configure uniform call treatment for 180 messages with or without SDP



Note

Early media cut-through for 180 messages with SDP is disabled by default; no configuration tasks are required to disable it. To re-enable the feature or to disable it after it has been re-enabled, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **[no] disable-early-media 180**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	disable-early-media 180 or no disable-early-media 180 Example: Router(config-sip-ua)# disable-early-media 180 or Router(config-sip-ua)# no disable-early-media 180	Disables or (by means of no form of the command) reenables early media cut-through for 180 messages with SDP.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP Extensions for Caller Identity and Privacy

To configure SIP extensions for caller identify and privacy, perform the following steps.

- [Configure Remote Party-ID, page 77](#) (optional)
- [Configure SIP-to-PSTN Calling-Info Policy, page 78](#) (optional)
- [Configure PSTN-to-SIP Calling-Info Policy, page 79](#) (optional)

Configure Remote Party-ID

This feature is enabled by default; no configuration tasks are required to enable this feature. If the feature is disabled by means of the **no remote-party-id** command, perform this task to re-enable the feature.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **remote-party-id**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
Step 4	remote-party-id Example: Router(config-sip-ua)# remote-party-id	(Optional) Configures remote-party-id translation. The following apply: <ul style="list-style-type: none"> • If a Remote-Party-ID header is present in the incoming INVITE message, the calling name and number extracted from the Remote-Party-ID header are sent as the calling name and number in the outgoing Setup message. This is the default behavior. Use the remote-party-id command to enable this option. • When no Remote-Party-ID header is available, no translation occurs so the calling name and number are extracted from the From header and are sent as the calling name and number in the outgoing Setup message. This treatment also occurs when the feature is disabled.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configure SIP-to-PSTN Calling-Info Policy

When Remote-Party-ID support is enabled, the default calling-info treatment is the following:

- The calling name and calling number are bidirectionally translated between the display-name and the user part of the Remote-Party-ID header of the SIP INVITE message and the calling name and calling number of the PSTN Setup message.
- If a PSTN to SIP call is marked as presentation prohibited, the display-name is populated with “anonymous”. Otherwise, the display-name and user part of the From header of the outgoing INVITE are populated with the calling name and calling number.

To override the default calling-info treatment, perform this task to optionally configure SIP to PSTN calling-info policy.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **calling-info sip-to-pstn**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p>enable</p> <p>Example: Router> enable</p>	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	<p>configure terminal</p> <p>Example: Router# configure terminal</p>	Enters global configuration mode.
Step 3	<p>sip-ua</p> <p>Example: Router(config)# sip-ua</p>	Enters SIP user-agent configuration mode.
Step 4	<p>calling-info sip-to-pstn [unscreened discard] [name set name] [number set number]</p> <p>Example: Router(config-sip-ua)# calling-info sip-to-pstn unscreened discard</p>	<p>Specifies calling-information treatment for SIP-to-PSTN calls. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • unscreened discard—Calling name and number are discarded. If the incoming SIP INVITE message does not contain a screened (;screen=yes) Remote-Party-ID header, then no name or number is sent in the forwarded Setup message. • name set name—Calling name is unconditionally set to the <i>name</i> argument, a configured ASCII string, in the forwarded Setup message. • number set number—Calling number is unconditional set to the <i>number</i> argument, a configured ASCII string, in the forwarded Setup message.
Step 5	<p>exit</p> <p>Example: Router(config-sip-ua)# exit</p>	Exits the current mode.

Configure PSTN-to-SIP Calling-Info Policy

To override the default calling-info treatment, perform this task to optionally configure PSTN to SIP calling-info policy.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **calling-info pstn-to-sip**

5. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p>enable</p> <p>Example: Router> enable</p>	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	<p>configure terminal</p> <p>Example: Router# configure terminal</p>	Enters global configuration mode.
Step 3	<p>sip-ua</p> <p>Example: Router(config)# sip-ua</p>	Enters SIP user-agent configuration mode.
Step 4	<p>calling-info pstn-to-sip [unscreened discard] [from [name set <i>name</i>] [number set <i>number</i>]] [remote-party-id [name set <i>name</i>] [number set <i>number</i>]]</p> <p>Example: Router(config-sip-ua)# calling-info pstn-to-sip unscreened discard</p>	<p>Specifies calling-information treatment for PSTN-to-SIP calls. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • unscreened discard—Calling name and number are discarded. Unless the screening indicator in the incoming Setup is marked as “user-provided, passed screening” or “network-provided,” no calling name or number is sent in the forwarded INVITE message. • from name set <i>name</i>—Display name of the From header is unconditionally set to the <i>name</i> argument, a configured ASCII string, in the forwarded INVITE message. • from number set <i>number</i>—User part of the From header is unconditionally set to the <i>number</i> argument, a configured ASCII string, in the forwarded INVITE message. • remote-part-id name set <i>name</i>—Display name of the Remote-Party-ID header is unconditionally set to the <i>name</i> argument, a configured ASCII string, in the forwarded INVITE message. • remote-party-id number set <i>number</i>—User part of the Remote-Party-ID header is unconditionally set to the <i>number</i> argument, a configured ASCII string, in the forwarded INVITE message.
Step 5	<p>exit</p> <p>Example: Router(config-sip-ua)# exit</p>	Exits the current mode.

Configuring SIP INVITE Request with Malformed Via Header

There are no configuration steps for this feature. Use the **show sip-ua statistics** command (see the “[Verifying SIP Message Components, Session Timers, and Responses Configuration](#)” section on page 110) to display the Bad Request counter.

Configuring Privacy Headers

You can configure privacy headers according to values defined in RFC 3323 and RFC 3325 as shown in [Table 10](#).

Table 10 Privacy Header Values

Header Value	Description
Header	Indicates that the privacy service enforces privacy for all headers in the SIP message which might identify information about the subscriber.
Session	Indicates that the information held in the session description protocol (SDP) is hidden outside the trusted domain.
User	Enforces user-level privacy for the subscriber by removing any user identification from the SIP message.
Critical	Indicates that the message is rejected if the privacy service cannot or will not enforce the specified privacy. Note You can only add critical to privacy headers if you also choose a privacy header for user, header, session, or ID.
ID	Indicates the Network-Asserted Identity remains private with respect to SIP entries outside the user-authenticated trusted domain.
PSTN	Indicates information passed in from the PSTN Octet 3a of the CALLING PARTY Information Element enables privacy information on the VoIP side of the call (see Privacy Header PSTN with UAC Gateway , page 82).
System	Use system settings.
Disable	Disables the privacy.

UAC Gateway Behavior

When you configure the **privacy** command to use one of the header values shown in [Table 9](#), then the gateway’s outgoing message request contains a privacy header set to the corresponding privacy value. The following example shows the format of the “From” header, if you configure the **privacy** command, based on RFC 3323:

```
From: "Anonymous" <sip:anonymous@anonymous.invalid>; tag=<tag value>
```

If you configure the **privacy critical** command, the gateway adds a Proxy-Require header with the value set to critical. Thus, in the unlikely event that the user agent sends a request to an intermediary that does not support the described extension, the request will fail.

If you configure the **asserted-id pai** command, the gateway builds a PAI into the common SIP stack. The **asserted-id pai** command has priority over the Remote-Party-ID (RPID) header and removes this header from any outbound message even if the router is configured to use the RPID header.

If you configure the **asserted-id ppi** command, the gateway builds a PPI into the common SIP stack. The **asserted-id ppi** command has priority over the Remote-Party-ID (RPID) header and removes this header from any outbound message even if the router is configured to use the RPID header.

Privacy Header PSTN with UAC Gateway

You can use the **privacy pstn** command to derive information passed in from the PSTN Octet 3a of the CALLING PARTY Information Element to enable privacy information on the VoIP side of the call. The data within the CALLING PARTY field indicates whether or not you want to relay calling information. The CALLING PARTY field also supplies information and details about who supplied the information, and whether or not the information has been verified.

Table 11 summarizes the relationship between the ISDN Octet 3a values and the SIP-header values that the UAC gateway generates, when you configure the **privacy pstn** command.

Table 11 ISDN Octet 3a-to-SIP Header Mapping for UAC Gateways

ISDN Octet 3a		SIP Headers		
Presentation Status Bit(5)	Number Origin Bits(0,1)	From	Asserted-ID	Privacy
Presentation allowed.	User provided, number not screened.	No change.	PPI.	Not included.
Presentation allowed.	User provided, number passed, and network screened.	No change.	PAI.	Not included.
Presentation allowed.	User provided, number failed, and network screened.	No change.	PPI.	Not included.
Presentation	Network-provided number.	No change.	PAI.	Not included.
Presentation prohibited.	User provided, number not screened (00).	Anonymous.	P-Preferred Identity.	ID.
Presentation prohibited.	User provided, number passed, and network screened (01).	Anonymous.	P-Asserted Identity.	ID.
Presentation prohibited.	Network provided number (11).	Anonymous.	P-Asserted Identity.	ID.

Privacy Header PSTN with UAS Gateway

When you configure a privacy header for PSTN by using the **privacy pstn** command, a UAS gateway maps headers in the incoming SIP messages to Octet 3a fields on the outbound side of the call.

If a UAS gateway receives a message that has a Privacy header with a valid value, it ignores the **privacy** or **asserted-id** commands. The UAS gateway marks the outbound Octet 3a value Presentation Prohibited. If the UAS gateway does not receive a Privacy header, then the UAS gateway marks the outbound Octet 3a value as Presentation Allowed.

If a UAS gateway receives an Asserted-ID header, and a valid Privacy header is within the same message, then the UAS gateway uses the Asserted-ID to derive the ISDN Name and Number fields. If the UAS gateway receives an Asserted-ID with PAI, then the Octet 3a Number Origin is marked as “User Provided, passed network screening.” If the received Asserted-ID is PPI, then the Octet 3a Number Origin is marked as “User provided, number not screened.”

If a UAS gateway receives an Asserted-ID header that has no Privacy header in the same message, then the UAS gateway checks the **asserted-id** command. If you configure the **asserted-id** command, then the asserted-ID is used. Otherwise, the information in the “From” header is used to populate the appropriate ISDN fields.

Table 12 summarizes the relationship between the ISDN Octet 3a values and SIP header values that the UAS gateway generates on the outbound side of the call.

Table 12 SIP Header-to-ISDN Mapping for UAS Gateways

SIP Headers			ISDN Fields				Comments
rpid	asserted-id	privacy	Name	Number	Octet 3a		
					Presentation Status	Number Origin	
Yes	No	No	See Comments				You must configure the rpid command, or the “From” header information is used.
No	PAI	ID	PAI		Presentation prohibited	User provided	Any privacy header you configure sets the “From” header to Anonymous, and Privacy PAI is used.
No	PPI	ID	PPI		Presentation prohibited		Any privacy header you configure sets the “From” header to Anonymous, and Privacy PPI is used.
No	PAI	No	PAI		Presentation allowed	User provided	
No	PPI	No	PPI		Presentation allowed		
Yes	No	Yes	rpid		Presentation prohibited	rpid	
Yes	PAI	No	See Comments				
Yes	PPI	No	See Comments				If you configure the asserted-id command, privacy PPI is used.

Interaction with Caller ID When Privacy Exists

When you configure the **privacy pstn** command, on the UAC gateway side of the call, after configuring the **substitute name** command under the **clid (voice-service-voip)** command and defining no “Display Name” parameter, then the PAI or PPI substitutes the calling number in the Display field.

The following example show a PAI header when the **substitute name** command is not set:

```
P-Asserted_Identity: <sip:5551212@example.com>
```

If you set the **substitute name** command, the header in the example is modified:

```
P-Asserted_Identity: "5551212" <sip:5551212@example.com>
```

When you configure the **privacy pstn** command, after configuring the **strip pi-restrict all** command under the **clid (voice-service-voip)** command, and if the CALLING INFORMATION Octet 3a indicates that the number is restricted, then the PAI/PPI value is not sent.

On the UAS gateway side of the call, if you configure the **clid network-provided** command, it will override any value you set by using the **privacy** command. If you configure the **clid network-provided** command and a PPI is received, the number in the Octet 3a is set to “Network Provided.” If you do not configure the **clid network-provided** command, the number in the Octet 3a is set to “User Provided.”

If you configure the **calling-info pstn-to-sip unscreened discard** command and the **privacy pstn** command, and if the calling number has a screening indicator of “User-provided, not screened,” or “User-provided, failed screen” the PAI/PPI is not sent.

Table 13 summarizes the interaction when you configure the **privacy pstn** command.

Table 13 Interactions When Using the privacy pstn Command

Presentation Indication	Screening Indication	calling-info pstn-to-sip Command	Generated Headers
See Table 9.	See Table 9.	Not set.	If you do not configure the calling-info pstn-to-sip command, then see Table 9.
Presentation allowed.	User provided, not screened.	Unscreened discard.	From: <sip:example.com>; tag=1 Contact: <sip:example.com>
Presentation allowed.	User provided number passed network screening.	Unscreened discard.	From: <sip:5551212@example.com>; tag=1 Contact: <sip:5551212@example.com:5060> P-Asserted-Identity: <sip:5551212@example.com>
Presentation allowed.	User provided number failed network screening.	Unscreened discard.	From: <sip:example.com>; tag=1 Contact: <sip:example.com>
Presentation prohibited.	User provided, number.	Unscreened discard.	From: <sip:5551212@example.com>; tag=1 Contact: <sip:5551212@example.com:5060> P-Asserted-Identity: <sip:5551212@example.com>
Presentation prohibited.	User provided number, not screened.	Unscreened discard.	From: “Anonymous” <sip:anonymous@anonymous.com>; tag=1 Contact: <sip:example.com> Privacy: ID
Presentation prohibited	User provided number, failed network screening.	Unscreened discard	From: “Anonymous” <sip:anonymous@anonymous.com>; tag=1 Contact: <sip:example.com> Privacy: ID
Presentation prohibited	User provided number, passed network screening.	Unscreened discard	From: “Anonymous” <sip:anonymous@anonymous.com>; tag=1 P-Asserted-Identity: Contact: <sip:5551212@example.com:> Privacy: ID

Configuring SIP Session Timer Support

To configure SIP session timer support including the Min-SE value, perform the following steps.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `min-se`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> <code>enable</code>	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# <code>configure terminal</code>	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: Router(config)# <code>voice service voip</code>	Enters voice service VoIP configuration mode.
Step 4	<code>sip</code> Example: Router(conf-voi-serv)# <code>sip</code>	Enters SIP configuration mode.
Step 5	<code>min-se time</code> Example: Router(conf-serv-ua)# <code>min-se 90</code>	Changes the minimum-session-expiration header value for all calls that use the SIP session timer support feature. The argument is as follows: <ul style="list-style-type: none"> • <i>time</i>—Time, in seconds. Range: 60 to 86400 (one day). Default: 90 (1.5 minutes).
Step 6	<code>exit</code> Example: Router(conf-serv-ua)# <code>exit</code>	Exits the current mode.

Configuring SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion

The SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion feature implements support for Reason headers and buffered calling-name completion. Reason-header support on Cisco IOS gateways is defined by RFC 3326.

Feature benefits include the following:

- Reason-header support facilitates PSTN internetworking by providing a more deterministic method of transporting the actual PSTN disconnect cause code to a remote PSTN gateway.
- Buffered calling-name completion (such as buffered-invite timers) makes the process of receiving ISDN-display information in a subsequent ISDN FACILITY message transparent to the remote SIP endpoint.
- The requirement of an external SIP user-agent server (UAS) to support INFO message responses before the call is active is removed.

This section contains the following procedures:

- [Configure Reason-Header Override, page 86](#)
- [Configure Buffer Calling-Name Completion, page 87](#)

Configure Reason-Header Override

To configure Reason-header override, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **reason-header override**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
Step 4	reason-header override Example: Router(config-sip-ua)# reason-header override	Enables Reason-header override support.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configure Buffer Calling-Name Completion

To configure buffer calling-name completion, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers buffer-invite**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP-user-agent configuration mode.

	Command or Action	Purpose
Step 4	<code>timers buffer-invite timer</code> Example: Router(config-sip-ua)# timers buffer-invite 500	Enables the SIP buffer-invite timer and sets the timer interval. The argument is as follows: <ul style="list-style-type: none"> <code>timer</code>—Buffer-invite timer value, in ms. Range: 50 to 5000.
Step 5	<code>exit</code> Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers

This section contains the following procedures:

- [Configure SIP Header Support, page 88](#) (required)
- [Configure SIP SUBSCRIBE and NOTIFY for External Triggers, page 89](#) (optional)

Configure SIP Header Support

To configure SIP header support, perform the following steps.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `header-passing`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-serv) # sip	Enters SIP configuration mode.
Step 5	header-passing Example: Router(conf-serv-sip) # header-passing	Enables or disables SIP header-passing to applications. When the gateway receives SIP INVITE, SUBSCRIBE, and NOTIFY messages, this command enables passing SIP headers associated with these messages to the target application in the gateway.
Step 6	exit Example: Router(conf-serv-sip) # exit	Exits the current mode.

Configure SIP SUBSCRIBE and NOTIFY for External Triggers

To configure SIP subscription options, perform the following steps.

Prerequisites

- Enable SIP header passing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **subscription asnl session history**
6. **subscription maximum originate**
7. **exit**
8. **sip-ua**
9. **retry subscribe**
10. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-serv)# sip	Enters SIP configuration mode.
Step 5	subscription asnl session history [duration minutes] [count number] Example: Router(conf-serv-sip)# subscription asnl session history duration 10 count 100	(Optional) Specifies how long to keep Application SUBSCRIBE/NOTIFY Layer (ASNL) subscription history records and how many records to keep in memory.
Step 6	subscription maximum originate number Example: Router(conf-serv-sip)# subscription maximum originate 10	(Optional) Specifies the maximum number of outstanding subscriptions to be originated by the gateway, up to two times the maximum number of dial peers on the platform. Default is the maximum number of dial peers on the platform.
Step 7	exit Example: Router(conf-serv-sip)# exit	Exits the current mode.
Step 8	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
Step 9	<code>retry subscribe number</code> Example: Router(config-sip-ua)# <code>retry subscribe 10</code>	(Optional) Sets the number of times that a SIP SUBSCRIBE message is resent to the other user agent.
Step 10	<code>exit</code> Example: Router(config-sip-ua)# <code>exit</code>	Exits the current mode.

Configuring SIP Stack Portability

No configuration tasks are required to configure the SIP stack portability feature. The feature is enabled by default.

Configuring SIP: Domain Name Support in SIP Headers

This section contains the following procedures:

- [Configure the Hostname in Locally Generated SIP Headers, page 91](#)
- [Monitor the Hostname in Locally Generated SIP Headers, page 93](#)

Configure the Hostname in Locally Generated SIP Headers

You can configure the hostname in either of two configuration modes:

- [Gateway-Wide Configuration Mode, page 91](#)
- [Dial-Peer-Specific Configuration Mode, page 92](#)



Note

Dial-peer-specific configuration takes precedence over more general gateway-wide configuration.

Gateway-Wide Configuration Mode

This procedure allows global configuration of the local hostname for use for locally generated URIs.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `localhost dns:local-host-name-string`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(config-voip-serv)# sip	Enters SIP configuration mode.
Step 5	localhost dns:local-host-name-string Example: Router(config-serv-sip)# localhost dns:example.com	Enters a local hostname string.
Step 6	exit Example: Router(config-serv-sip\)# exit	Exits the current mode.

Dial-Peer-Specific Configuration Mode

This procedure allows dial-peer configuration of the local hostname for use for locally generated URIs.

**Note**

This procedure takes precedence over more general gateway-wide configuration.

.SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **voice-class sip localhost [dns]:local-host-name-string**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router# dial-peer voice 100 voip	Enters dial-peer configuration mode for the specified dial peer.
Step 4	voice-class sip localhost [dns]:local host-name-string Example: Router(config-dial-peer)# voice-class sip localhost dns:example.com	Enters a local hostname string.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Monitor the Hostname in Locally Generated SIP Headers

This procedure monitors the gateway-wide or dial-peer-specific configuration.

SUMMARY STEPS

1. enable
2. show call active voice
3. show call history voice
4. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> <code>enable</code>	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<code>show call active voice</code> Example: Router# <code>show call active voice</code>	Displays call information for voice calls in progress.
Step 3	<code>show call history voice</code> Example: Router# <code>show call history voice</code>	Displays the call history table for voice calls.
Step 4	<code>exit</code> Example: Router# <code>exit</code>	Exits the current mode.

Examples

This section provides the following command output:

- [show call active Command Output: Example, page 94](#)
- [show call history Command Output: Example, page 97](#)

show call active Command Output: Example

The following example shows active-call command output when the local hostname is enabled.

```
Router# show call active voice

Telephony call-legs:1
SIP call-legs:1
H323 call-legs:0
Call agent controlled call-legs:0
Multicast call-legs:0
Total call-legs:2

GENERIC:
SetupTime=126640 ms
Index=1
PeerAddress=9001
PeerSubAddress=
PeerId=100
PeerIfIndex=6
LogicalIfIndex=4
ConnectTime=130300 ms
CallDuration=00:00:47 sec
CallState=4
CallOrigin=2
ChargedUnits=0
```

```
InfoType=speech
TransmitPackets=2431
TransmitBytes=48620
ReceivePackets=2431
ReceiveBytes=48620
TELE:
ConnectionId=[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
IncomingConnectionId=[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
CallID=1
TxDuration=48620 ms

VoiceTxDuration=48620 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-61
ACOMLevel=3
OutSignalLevel=-35
InSignalLevel=-30
InfoActivity=2
ERLLevel=3
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=9001
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwCollectedCalledNumber=9002

GENERIC:
SetupTime=128980 ms
Index=1
PeerAddress=9002
PeerSubAddress=
PeerId=3301
PeerIfIndex=7
LogicalIfIndex=0
ConnectTime=130300 ms
CallDuration=00:00:50 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=2587
TransmitBytes=51740
ReceivePackets=2587
ReceiveBytes=51740
VOIP:
ConnectionId[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
IncomingConnectionId[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
CallID=2
RemoteIPAddress=172.18.193.87
RemoteUDPPort=17602
RemoteSignallingIPAddress=172.18.193.87
RemoteSignallingPort=5060
```

```
RemoteMediaIPAddress=172.18.193.87
RemoteMediaPort=17602
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=sipv2
ProtocolCallId=A240B4DC-115511D9-8005EC82-AB4FD5BE@pip.example.com
SessionTarget=172.18.193.87
OnTimeRvPlayout=48620
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=69 ms
TxPakNumber=2434
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=48680
TxVoiceDuration=48680
RxPakNumber=2434
RxSignalPak=0
RxDuration=0
TxVoiceDuration=48670
VoiceRxDuration=48620
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=69
PlayDelayMin=69
PlayDelayMax=70
PlayDelayClockOffset=43547
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverflow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-35
InSignalLevel=-30
LevelTxPowerMean=0
LevelRxPowerMean=-302
LevelBgNoise=0
ERLLevel=3
ACOMLevel=3
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = enabled
```



```

CoderTypeRate=g729r8
CodecBytes=20
Media Setting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=9001
OriginalCallingOctet=0x0
OriginalCalledNumber=9002
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=9001
TranslatedCallingOctet=0x0
TranslatedCalledNumber=9002
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwCollectedCalledNumber=9002
GwOutpulsedCalledNumber=9002
GwOutpulsedCalledOctet3=0x80
GwOutpulsedCallingNumber=9001
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=
LocalHostname=pip.example.com ! LocalHostname field
Telephony call-legs:1
SIP call-legs:1
H323 call-legs:0
Call agent controlled call-legs:0
Multicast call-legs:0
Total call-legs:2

```

show call history Command Output: Example

The following example shows call-history command output when the local hostname is enabled.

```
Router# show call history voice
```

```

Telephony call-legs:1
SIP call-legs:1
H323 call-legs:0
Call agent controlled call-legs:0
Total call-legs:2

GENERIC:
SetupTime=128980 ms
Index=1
PeerAddress=9002
PeerSubAddress=
PeerId=3301
PeerIfIndex=7
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=130300 ms
DisconnectTime=329120 ms
CallDuration=00:03:18 sec
CallOrigin=1
ReleaseSource=4
ChargedUnits=0

```

```

InfoType=speech
TransmitPackets=9981
TransmitBytes=199601
ReceivePackets=9987
ReceiveBytes=199692
VOIP:
ConnectionId[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
IncomingConnectionId[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
CallID=2
RemoteIPAddress=172.18.193.87
RemoteUDPPort=17602
RemoteSignallingIPAddress=172.18.193.87
RemoteSignallingPort=5060
RemoteMediaIPAddress=172.18.193.87
RemoteMediaPort=17602
SRTP = off
RoundTripDelay=1 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=sipv2
ProtocolCallId=A240B4DC-115511D9-8005EC82-AB4FD5BE@pip.example.com
SessionTarget=172.18.193.87
OnTimeRvPayout=195880
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPayoutDelay=70 ms
LoWaterPayoutDelay=69 ms
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=2
MediaSetting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=9001
OriginalCallingOctet=0x0
OriginalCalledNumber=9002
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=9001
TranslatedCallingOctet=0x0
TranslatedCalledNumber=9002
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwCollectedCalledNumber=9002
GwOutpulsedCalledNumber=9002
GwOutpulsedCalledOctet3=0x80
GwOutpulsedCallingNumber=9001

```

```
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
LocalHostname=pip.example.com ! LocalHostname field
Username=

GENERIC:
SetupTime=126640 ms
Index=2
PeerAddress=9001
PeerSubAddress=
PeerId=100
PeerIfIndex=6
LogicalIfIndex=4
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=130300 ms
DisconnectTime=330080 ms
CallDuration=00:03:19 sec
CallOrigin=2
ReleaseSource=4
ChargedUnits=0
InfoType=speech
TransmitPackets=9987
TransmitBytes=199692
ReceivePackets=9981
ReceiveBytes=199601
TELE:
ConnectionId=[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
IncomingConnectionId=[0xA0DC41CF 0x115511D9 0x8002EC82 0xAB4FD5BE]
CallID=1
TxDuration=195940 ms
VoiceTxDuration=195940 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-73
ACOMLevel=4
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=9001
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwCollectedCalledNumber=9002
```

Configuring SIP Gateway Support for Session Information

There are no tasks for configuring SIP gateway support for session information.

Configuring SIP Gateway Support for Permit Hostname CLI

To configure a list of hostname to validate against incoming INVITE messages, perform the following steps.

Restrictions

Hostname can be a maximum of 30 characters; hostnames longer than 30 characters are truncated.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **permit hostname dns:<domain name>**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router (config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	permit hostname dns:<domain name> Example: Router (config-sip-ua)# permit hostname dns:sip.example.com	Validates a hostname of initial incoming INVITE messages. The argument is: <ul style="list-style-type: none"><i>domain name</i>—Domain name in DNS format. Domain names can be up to 30 characters in length; those domain names exceeding 30 characters are truncated.
Step 5	exit Example: Router (config-sip-ua)# exit	Exits the current mode.

Configuring Outbound Proxy Support for the SIP Gateway

This section describes the procedures for configuring an outbound-proxy server. These procedures include the following:

- [Configuring an Outbound-Proxy Server Globally on a Gateway, page 101](#)
- [Configuring an Outbound-Proxy Server on a Dial Peer, page 102](#)

Configuring an Outbound-Proxy Server Globally on a Gateway

To configure SIP support for an outbound-proxy server globally on a SIP gateway, follow these steps:

SUMMARY STEPS

- enable**
- configure terminal**
- voice service {pots | voatm | vofr | voip}**
- sip**
- outbound-proxy ip-address**
- exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service {pots voatm vofr voip} Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode and specifies the voice encapsulation.
Step 4	sip Example: Router(conf-voi-ser)# sip	Enters SIP configuration mode.
Step 5	outbound-proxy <i><ipv4:A:B:C:D:port/dns:host.domain></i> Example: Router(conf-ser-sip)# outbound-proxy dns:sipproxy.example.com	Configures an outbound proxy server. This example shows how to configure an outbound-proxy server to a SIP proxy server in the domain example.com.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Configuring an Outbound-Proxy Server on a Dial Peer

To configure an outbound-proxy server on a dial peer, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag {pots | vofr | voip}**
4. **voice-class sip**
5. **sip**
6. **outbound-proxy {ipv4:ip-address[:port-number] | dns:host:domain}**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag {pots vofr voip} Example: Router(conf)# dial-peer voice 111 voip	Define a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.
Step 4	voice-class sip Example: Router(config-dial-peer)# voice service voip	Enters dial-peer VoIP configuration mode.
Step 5	sip Example: Router(conf-voi-serv)# sip	Enters SIP configuration mode.
Step 6	outbound-proxy {ipv4:ip-address[:port-number] dns:host:domain} Example: Router(conf-serv-sip)# outbound-proxy ipv4:10.1.1.1	Configures an outbound proxy server. This example shows how to configure an outbound-proxy server to IP address 10.1.1.1.
Step 7	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Configuring SIP Support for PAI

This section provides procedures for configuring the following supplementary services:

- [Configuring a Privacy Header, page 104](#)
- [Configuring an Outbound-Proxy Server on a Dial Peer, page 102](#)
- [Configuring a Name and Number in the asserted-id Header, page 106](#)

Configuring a Privacy Header

To configure a privacy header in support of RFC 3323, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **privacy {pstn | *privacy-option* [critical]}**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-ser)# sip	Enters SIP configuration mode.
Step 5	privacy {pstn <i>privacy-option</i> [critical]} Example: Router(conf-ser-sip)# privacy pstn	Configures a privacy header set to a value supported by RFC 3323. In this example, the privacy information from the PSTN side of a call is passed on to the VoIP side. The PSTN information is passed in the Octet 3a of the CALLING PARTY Information Element.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Configuring a Privacy Level

To configure a privacy header level for PAI or PPI, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **asserted-id [pai | ppi]**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-ser)# sip	Enters SIP configuration mode.
Step 5	asserted-id [pai ppi] Example: Router(conf-ser-sip)# asserted-id ppi	Configures a privacy header. In this example, a P-Preferred-Identity header is configured.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Configuring a Name and Number in the asserted-id Header

To set a name and number in the asserted-id, follow these steps:

SUMMARY STEPS

- enable**
- configure terminal**
- voice service voip**
- sip**
- calling-info pstn-to-sip {unscreened discard | {from | remote-part-id | asserted-id {name set *name* | name set *number*}}}**
- exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-ser)# sip	Enters SIP configuration mode.
Step 5	caller-info pstn-to-sip {unscreened discard {from remote-party-id asserted-id} {name set name number set number}}} Example: Router(conf-ser-sip)# caller-info pstn-to-sip asserted-id name set example	Configures a name that is populated in the asserted-id field.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Configuring SIP History-info Header Support

You can configure SIP History-info Header Support in two different configuration modes: voice service sip configuration (global level) and dial peer voice configuration (dial-peer level) mode. The gateway first checks if support is configured under the dial peer voice configuration mode, and failing a match, then defaults to the voice service configuration. If support is not configured in either mode, then the header is not added.

This section contains the following procedures:

- [Configuring SIP History-info Header Support Globally, page 108](#)
- [Configuring SIP History-info Header Support at the Dial-Peer Level, page 109](#)

Configuring SIP History-info Header Support Globally

Perform this task to configure history-info header support at a global level in SIP configuration (conf-serv-sip) mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **history-info**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode, or other security level set by a system administrator. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-serv)# sip	Enters SIP configuration mode.
Step 5	history-info Example: Router(conf-serv-sip)# history-info	Configures history-info header support globally.
Step 6	exit Example: Router(conf-serv-sip)# exit	Exits the current mode.

Configuring SIP History-info Header Support at the Dial-Peer Level

Perform this task to configure history-info header support at the dial-peer level, in dial peer voice configuration (config-dial-peer) mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *tag* voip**
4. **voice-class sip history-info**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode, or other security level set by a system administrator. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>tag</i> voip Example: Router(config)# dial-peer voice 2 voip	Enters dial peer VoIP configuration mode.
Step 4	voice-class sip history-info Example: Router(config-dial-peer)# voice-class sip history-info	Configures history-info header support for a dial peer.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Verifying SIP Message Components, Session Timers, and Responses Configuration

To verify SIP message components, session timers, and responses configuration, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show call active voice**
2. **show call application sessions**
3. **show call history**
4. **show logging**
5. **show running-config**
6. **show sip min-se**
7. **show sip-ua map pstn-sip**
8. **show sip-ua map sip-pstn**
9. **show sip-ua statistics**
10. **show sip-ua status**
11. **show sip-ua timers**
12. **show subscription**

DETAILED STEPS

Step 1 **show call active voice**

Use this command to display call information for voice calls in progress.



Note For sample output, see the [“Monitor the Hostname in Locally Generated SIP Headers”](#) section on page 93.

Step 2 **show call application sessions**

Use this command to view whether the application is running.

```
Router# show call application sessions
```

```
TCL Sessions
```

```
There are 1 active TCL sessions
```

SID	Name	Called	Calling	App Name	Legs
1		50276	50280	testapp31	4

```
VXML Sessions
```

```
No running VXML sessions
```

Step 3 **show call history**

Use this command, optionally with the **voice** keyword, to display the call history table for voice calls.

```
Router# show call history
```

```
DisconnectCause=10
DisconnectText=normal call clearing
.
.
.
```



Note For more sample output, see the [“Monitor the Hostname in Locally Generated SIP Headers” section on page 93.](#)

Step 4 show logging

Use this command to display the state of logging (syslog).

The following partial sample output shows that the outgoing gateway is receiving a 180 message with SDP and is configured to ignore the SDP.

```
Router# show logging

Log Buffer (600000 bytes):

00:12:19:%SYS-5-CONFIG_I:Configured from console by console
00:12:19:%SYS-5-CONFIG_I:Configured from console by console
00:12:20:0x639F6EEC :State change from (STATE_NONE, SUBSTATE_NONE) to
(STATE_IDLE, SUBSTATE_NONE)
00:12:20:****Adding to UAC table

00:12:20:adding call id 2 to table

00:12:20: Queued event from SIP SPI :SIPSPI_EV_CC_CALL_SETUP
00:12:20:CCSIP-SPI-CONTROL: act_idle_call_setup
00:12:20: act_idle_call_setup:Not using Voice Class Codec

00:12:20:act_idle_call_setup:preferred_codec set[0] type :g711ulaw
bytes:160
00:12:20:sipSPICopyPeerDataToCCB:From CLI:Modem NSE payload = 100,
Passthrough = 0,Modem relay = 0, Gw-Xid = 1
SPRT latency 200, SPRT Retries = 12, Dict Size = 1024
String Len = 32, Compress dir = 3
00:12:20:sipSPICanSetFallbackFlag - Local Fallback is not active
00:12:20:****Deleting from UAC table

00:12:20:****Adding to UAC table

00:12:20: Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION
00:12:20:0x639F6EEC :State change from (STATE_IDLE, SUBSTATE_NONE) to
(STATE_IDLE, SUBSTATE_CONNECTING)
00:12:20:0x639F6EEC :State change from (STATE_IDLE,
SUBSTATE_CONNECTING) to (STATE_IDLE, SUBSTATE_CONNECTING)
00:12:20:sipSPIUsetBillingProfile:sipCallId for billing records =
41585FCE-14F011CC-8005AF80-D4AA3153@10.1.1.42
00:12:20:CCSIP-SPI-CONTROL: act_idle_connection_created
00:12:20:CCSIP-SPI-CONTROL: act_idle_connection_created:Connid(1)
created to 10.1.1.15:5060, local_port 57838
00:12:20:CCSIP-SPI-CONTROL: sipSPIOutgoingCallSDP
00:12:20:sipSPISetMediaSrcAddr: media src addr for stream 1 = 10.1.1.42
00:12:20:sipSPIReserveRtpPort:reserved port 18978 for stream 1
00:12:20: convert_codec_bytes_to_ptime:Values :Codec:g711ulaw
codecbytes :160, ptime:20

00:12:20:sip_generate_sdp_xcaps_list:Modem Relay disabled. X-cap not needed

00:12:20:Received Octet3A=0x00 -> Setting ;screen=no ;privacy=off
```

```

00:12:20:sipSPIAddLocalContact
00:12:20: Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE
00:12:20:sip_stats_method
00:12:20:sipSPIProcessRtpSessions
00:12:20:sipSPIAddStream:Adding stream 1 (callid 2) to the VOIP RTP library
00:12:20:sipSPISetMediaSrcAddr: media src addr for stream 1 = 10.1.1.42
00:12:20:sipSPIUpdateRtcpSession:for m-line 1
00:12:20:sipSPIUpdateRtcpSession:rtcp_session info
laddr = 10.1.1.42, lport = 18978, raddr = 0.0.0.0,
rport=0, do_rtcp=FALSE
src_callid = 2, dest_callid = -1

00:12:20:sipSPIUpdateRtcpSession:No rtp session, creating a new one

00:12:20:sipSPIAddStream:In State Idle
00:12:20:act_idle_connection_created:Transaction active. Facilities will be queued.
00:12:20:0x639F6EEC :State change from (STATE_IDLE,
SUBSTATE_CONNECTING) to (STATE_SENT_INVITE, SUBSTATE_NONE)
00:12:20:Sent:
INVITE sip:222@10.1.1.15:5060 SIP/2.0
Via:SIP/2.0/UDP 10.1.1.42:5060
From:"111" <sip:111@10.1.1.42>;tag=B4DC4-9E1
To:<sip:222@10.1.1.15>
Date:Mon, 01 Mar 1993 00:12:20 GMT
Call-ID:41585FCE-14F011CC-8005AF80-D4AA3153@10.1.1.42
Supported:timer
Min-SE: 1800
Cisco-Guid:1096070726-351277516-2147659648-3567923539
User-Agent:Cisco-SIPGateway/IOS-12.x
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE,
NOTIFY, INFO
CSeq:101 INVITE
Max-Forwards:6
Remote-Party-ID:<sip:111@10.1.1.42>;party=calling;screen=no;privacy=off
Timestamp:730944740
Contact:<sip:111@10.1.1.42:5060>
Expires:180
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:230

v=0
o=CiscoSystemsSIP-GW-UserAgent 4629 354 IN IP4 10.1.1.42
s=SIP Call
c=IN IP4 10.1.1.42
t=0 0
m=audio 18978 RTP/AVP 0 100
c=IN IP4 10.1.1.42
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20

00:12:21:Received:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP 10.1.1.42:5060
From:"111" <sip:111@10.1.1.42>;tag=B4DC4-9E1
To:<sip:222@10.1.1.15>;tag=442AC-22
Date:Wed, 16 Feb 2000 18:19:56 GMT
Call-ID:41585FCE-14F011CC-8005AF80-D4AA3153@10.1.1.42
Timestamp:730944740
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event

```



```

Content-Length:0

00:12:21:HandleUdpSocketReads :Msg enqueued for SPI with IPAddr:
10.1.1.15:5060
00:12:21:CCSIP-SPI-CONTROL: act_sentininvite_new_message
00:12:21:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:12:21:sip_stats_status_code
00:12:21: Roundtrip delay 420 milliseconds for method INVITE

00:12:21:0x639F6EEC :State change from (STATE_SENT_INVITE,
SUBSTATE_NONE) to (STATE_REC'D_PROCEEDING, SUBSTATE_PROCEEDING_PROCEEDING)
00:12:21:Received:
SIP/2.0 180 Ringing
Via:SIP/2.0/UDP 10.1.1.42:5060
From:"111" <sip:111@10.1.1.42>;tag=B4DC4-9E1
To:<sip:222@10.1.1.15>;tag=442AC-22
Date:Wed, 16 Feb 2000 18:19:56 GMT
Call-ID:41585FCE-14F011CC-8005AF80-D4AA3153@10.1.1.42
Timestamp:730944740
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Contact:<sip:222@192.0.2.59:5060>
Record-Route:<sip:222@10.1.1.15:5060;maddr=10.1.1.15>
Content-Length:230
Content-Type:application/sdp

v=0
o=CiscoSystemsSIP-GW-UserAgent 4629 354 IN IP4 10.1.1.42
s=SIP Call
c=IN IP4 10.1.1.42
t=0 0
m=audio 18978 RTP/AVP 0 100
c=IN IP4 10.1.1.42
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20

00:12:21:HandleUdpSocketReads :Msg enqueued for SPI with IPAddr:
10.1.1.15:5060
00:12:21:CCSIP-SPI-CONTROL: act_rec'dproc_new_message
00:12:21:CCSIP-SPI-CONTROL: act_rec'dproc_new_message_response
00:12:21:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:12:21:sip_stats_status_code
00:12:21: Roundtrip delay 496 milliseconds for method INVITE
00:12:21:CCSIP-SPI-CONTROL: act_rec'dproc_new_message_response :Early
media disabled for 180:Ignoring SDP if present
00:12:21:HandleSIP1xxRinging:SDP in 180 will be ignored if present: No
early media cut through
00:12:21:HandleSIP1xxRinging:SDP Body either absent or ignored in 180
RINGING:- would wait for 200 OK to do negotiation.
00:12:21:HandleSIP1xxRinging:MediaNegotiation expected in 200 OK

00:12:21:sipSPIGetGtdBody:No valid GTD body found.
00:12:21:sipSPICreateRawMsg:No GTD passed.
00:12:21:0x639F6EEC :State change from (STATE_REC'D_PROCEEDING,
SUBSTATE_PROCEEDING_PROCEEDING) to (STATE_REC'D_PROCEEDING,
SUBSTATE_PROCEEDING_ALERTING)
00:12:21:HandleSIP1xxRinging:Transaction Complete. Lock on Facilities
released.
00:12:22:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 10.1.1.42:5060

```

```

From: "111" <sip:111@10.1.1.42>;tag=B4DC4-9E1
To:<sip:222@10.1.1.15>;tag=442AC-22
Date:Wed, 16 Feb 2000 18:19:56 GMT
Call-ID:41585FCE-14F011CC-8005AF80-D4AA3153@10.1.1.42
Timestamp:730944740
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow:INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE,
NOTIFY, INFO
Allow-Events:telephone-event
Contact:<sip:222@10.1.1.59:5060>
Record-Route:<sip:222@10.1.1.15:5060;maddr=10.1.1.15>
Content-Type:application/sdp
Content-Length:231

v=0
o=CiscoSystemsSIP-GW-UserAgent 9600 4816 IN IP4 10.1.1.59
s=SIP Call
c=IN IP4 10.1.1.59
t=0 0
m=audio 19174 RTP/AVP 0 100
c=IN IP4 10.1.1.59
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20

```

Step 5 show running-config

Use this command to display the contents of the currently running configuration file or the configuration for a specific interface. Use it to display the current configuration and to verify header passing and subscription configuration.



Note If early media (the default setting) is enabled, this command does not show any information related to the feature.

The following sample output shows that the **disable-early-media 180** command was used.

```

Router# show running-config

.
.
.
dial-peer voice 223 pots
  application session
  destination-pattern 223
  port 1/0/0
!
gateway
!
sip-ua
  disable-early-media 180

```

Step 6 show sip min-se

Use this command to show the current value of a minimum-session-expiration header for calls that use SIP.

```

Router# show sip min-se

SIP UA MIN-SE Value (seconds)
Min-SE: 90

```

Step 7 `show sip-ua map pstn-sip`

Use this command to display the mapping table of PSTN cause codes and their corresponding SIP error status codes or the mapping table of PSTN-to-SIP codes.

```
Router# show sip-ua map pstn-sip
```

PSTN-Cause	Configured SIP-Status	Default SIP-Status
1	404	404
2	404	404
3	404	404
4	500	500
.		
.		
110	500	500
111	400	400
126	500	500
127	500	500

Step 8 `show sip-ua map sip-pstn`

Use this command to display the mapping table of PSTN cause codes and their corresponding SIP error status codes or the mapping table of SIP-to-PSTN codes.

```
Router# show sip-ua map sip-pstn
```

SIP-Status	Configured PSTN-Cause	Default PSTN-Cause
400	127	127
401	57	57
402	21	21
403	57	57
.		
.		
600	17	17
603	21	21
604	1	1
606	58	58

Step 9 `show sip-ua statistics`

Use this command to display response, traffic, and retry SIP statistics, including the Bad Request counter. Use it to verify configuration of the SIP INVITE Request with Malformed Via Header feature, which increments a counter (shown as *Client Error: Bad Request*) when a malformed Via header is received.



Note To reset counters after you view statistics, use the `clear sip-ua statistics` command.

The following sample output shows response, traffic, and retry SIP statistics, including the Bad Request counter. Use it to verify configuration of the SIP INVITE Request with Malformed Via Header feature, which increments a counter (shown as *Client Error: Bad Request*) when a malformed Via header is received.

```
Router# show sip-ua statistics
```

SIP Response Statistics (Inbound/**Outbound**)
 Informational:
 Trying 0/0, Ringing 0/0,

```

    Forwarded 0/0, Queued 0/0,
    SessionProgress 0/0
Success:
    OkInvite 0/0, OkBye 0/0,
    OkCancel 0/0, OkOptions 0/0,
    OkPrack 0/0, OkPreconditionMet 0/0
Redirection (Inbound only):
    MultipleChoice 0, MovedPermanently 0,
    MovedTemporarily 0, SeeOther 0,
    UseProxy 0, AlternateService 0
Client Error:
    BadRequest 0/0, Unauthorized 0/0,
    PaymentRequired 0/0, Forbidden 0/0,
    NotFound 0/0, MethodNotAllowed 0/0,
    NotAcceptable 0/0, ProxyAuthReqd 0/0,
    ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
    LengthRequired 0/0, ReqEntityTooLarge 0/0,
    ReqURITooLarge 0/0, UnsupportedMediaType 0/0,
    BadExtension 0/0, TempNotAvailable 0/0,
    CallLegNonExistent 0/0, LoopDetected 0/0,
    TooManyHops 0/0, AddrIncomplete 0/0,
    Ambiguous 0/0, BusyHere 0/0,
    RequestCancel 0/0, NotAcceptableMedia 0/0
Server Error:
    InternalError 0/0, NotImplemented 0/0,
    BadGateway 0/0, ServiceUnavail 0/0,
    GatewayTimeout 0/0, BadSipVer 0/0,
    PreCondFailure 0/0
Global Failure:
    BusyEverywhere 0/0, Decline 0/0,
    NotExistAnywhere 0/0, NotAcceptable 0/0

SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0

Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliable1xx 0

```

Step 10 show sip-ua status

Use this command to display status for the SIP user agent.

The following sample output shows status for the SIP user agent after the **disable-early-media 180** command was used.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP :ENABLED
SIP User Agent for TCP :ENABLED
SIP User Agent bind status(signaling):ENABLED 10.0.0.0
SIP User Agent bind status(media):ENABLED 0.0.0.0
SIP early-media for 180 responses with SDP:DISABLED
SIP max-forwards :6
SIP DNS SRV version:2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP:NONE
Check media source packets:DISABLED
Redirection (3xx) message handling:ENABLED

SDP application configuration:

```

```

Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported:audio image
Network types supported:IN
Address types supported:IP4
Transport types supported:RTP/AVP udpt1

```

Step 11 show sip-ua timers

Use this command to display all SIP UA information.

```
Router# show sip-ua timers
```

```

SIP UA Timer Values (milliseconds)
trying 500, expires 150000, connect 500, disconnect 500
comet 500, prack 500, rellxx 500, notify 500, refer 500,
hold 2880 minutes, buffer-invite 500

```

Step 12 show subscription { asnl session { active | history [errors | session-id *session-id* | url] | statistics } | sip } [summary]

Use this command to display information about Application SUBSCRIBE/NOTIFY Layer (ASNL)-based and non-ASNL-based SIP subscriptions.

```
Router# show subscription asnl session history
```

```

ASNL Subscription History Records Details:
=====
Total history records = 1
Total error count = 0
Total subscription requests sent = 1
Total subscription requests received = 0
Total notification requests sent = 0
Total notification requests received = 3
URL: sip:user@10.7.104.88
  Event Name : stress
  Session ID : 8
  Expiration Time : 50 seconds
  Subscription Duration : 10 seconds
  Protocol : ASNL_PROTO_SIP
  Remote IP address : 10.7.104.88
  Port : 5060
  Call ID : 5
  Total Subscriptions Sent : 1
  Total Subscriptions Received: 0
  Total Notifications Sent : 0
  Total Notifications Received : 3
  Last response code : ASNL_UNSUBSCRIBE_SUCCESS
  Last error code : ASNL_NONE
  First Subscription Time : 10:55:12 UTC Apr 9 2000
  Last Subscription Time : 10:55:12 UTC Apr 9 2000
  First Notify Time : 10:55:12 UTC Apr 9 2000
  Last Notify Time : 10:55:22 UTC Apr 9 2000

```

```
Router# show subscription asnl session history summary
```

```

ASNL Subscription History Records Summary:
=====
Total history records = 2
Total error count = 0
Total subscription requests sent = 2
Total subscription requests received = 0
Total notification requests sent = 0

```

```
Total notification requests received = 6
```

URL	Session ID	Call ID
---	-----	-----
sip:user@10.7.104.88	9	5
sip:user@10.7.104.88	8	5

The following sample output shows the error type ASNL_SUBSCRIBE_FAILED. This error indicates that the subscription request has failed.

```
Router# show subscription asnl session history summary
```

```
ASNL Subscription History Records Summary:
```

```
=====
Total history records = 8
Total error count = 6
  Total error type (ASNL_SUBSCRIBE_FAILED) = 6
Total subscription requests sent = 8
Total subscription requests received = 0
Total notification requests sent = 0
Total notification requests received = 6
```

URL	Session ID	Call ID
---	-----	-----
sip:user@10.7.104.88	15	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	14	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	13	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	12	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	11	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	10	N/A
ASNL_SUBSCRIBE_FAILED		
sip:user@10.7.104.88	9	5
sip:user@10.7.104.88	8	5

```
Router# show subscription asnl session history error
```

```
ASNL Subscription History Error Statistics:
```

```
=====
Total history records = 8
Total history records with errors = 6
URL : sip:user@10.7.104.88
Session ID: 15
Call ID : N/A
Event Name: newstress
Error : ASNL_SUBSCRIBE_FAILED
URL : sip:user@10.7.104.88
Session ID: 14
Call ID : N/A
Event Name: newstress
Error : ASNL_SUBSCRIBE_FAILED
URL : sip:user@10.7.104.88
Session ID: 13
Call ID : N/A
Event Name: newstress
Error : ASNL_SUBSCRIBE_FAILED
URL : sip:user@10.7.104.88
Session ID: 12
Call ID : N/A
Event Name: newstress
Error : ASNL_SUBSCRIBE_FAILED
```

```

URL      : sip:user@10.7.104.88
Session ID: 11
Call ID  : N/A
Event Name: newstress
Error    : ASNL_SUBSCRIBE_FAILED
URL      : sip:user@10.7.104.88
Session ID: 10
Call ID  : N/A
Event Name: newstress
Error    : ASNL_SUBSCRIBE_FAILED

```

Router# **show subscription asnl session history url**

```

ASNL Subscription History URL Records Details:
=====
Total history records = 3
Total history records with errors = 0
Total number of different URLs = 1
Total number of different events = 2
Total subscription requests sent = 3
Total subscription requests received = 0
Total notification requests sent = 0
Total notification requests received = 9
URL: sip:user@10.7.104.88
  Event Name: stress1
    Session ID: 19          Call ID: N/A

  Event Name: stress
    Session ID: 18          Call ID: 5

  Event Name: newstress
    Session ID: 17          Call ID: N/A
Total error count for this URL                = 0
Total events subscribed by this URL           = 0
Total subscription requests sent for this URL = 3
Total subscription requests received for this URL = 0
Total notification requests sent for this URL = 0
Total notification requests received for this URL = 9

```

Router# **show subscription asnl session history url summary**

```

ASNL Subscription History URL Records Summary:
=====
Total history records = 3
Total history records with errors = 0
Total number of different URLs = 1
Total number of different events = 2
Total subscription requests sent = 3
Total subscription requests received = 0
Total notification requests sent = 0
Total notification requests received = 9

```

Router# **show subscription asnl session statistics**

```

ASNL Subscription and Notification Statistics:
=====
Total subscription requests sent = 3

```

Troubleshooting Tips for SIP Message, Timer, and Response Features



Note

For general troubleshooting tips and a list of important **debug** commands, see the “[General Troubleshooting Tips](#)” section on page 18.

- Make sure that you can make a voice call.
- Use the **debug asnl events** command to verify that the SIP subscription server is up. For example, the output displays a pending message if the client is unsuccessful in communicating with the server:
- If this is an H.323 gateway, use the **debug cch323** family of commands to enable H.323 debugging capabilities.
- If this is a SIP gateway, use the **debug ccsip** family of commands to enable SIP debugging capabilities. Use the **debug ccsip all** command to view all the SIP messages to trace a call.
- Use the **debug isdn q931** command to display information about call setup and tear down of ISDN network connections (layer 3) between the local router (user side) and the network.
- Use the **debug radius** command to display information associated with RADIUS.
- Use the **debug voip ccapi protoheaders** command to view messages sent between the originating and terminating gateways. If no headers are being received by the terminating gateway, verify that the **header-passing** command is enabled on the originating gateway.
- Use the **debug voip ivr script** command to display any errors that might occur when the Tcl script is run.

Following is sample output for some of these commands:

- [Sample Output for the debug asnl events Command, page 120](#)
- [Sample Output for the debug ccsip all Command: Originating Gateway, page 120](#)
- [Sample Output for the debug ccsip all Command: Terminating Gateway, page 121](#)
- [Sample Output for the debug ccsip messages Command, page 122](#)
- [Sample Output for the debug isdn q931 Command, page 125](#)
- [Sample Output for the debug voip ccapi protoheaders Command: Originating Gateway, page 126](#)
- [Sample Output for the debug voip ccapi protoheaders Command: Terminating Gateway, page 126](#)
- [Sample Output for the debug voip ivr script Command, page 126](#)

Sample Output for the debug asnl events Command

```
Router# debug asnl events
```

```
*Mar 1 02:48:48.831: //-1//ASNL:SUB7:/asnl_subscribe: resp = ASNL_SUBSCRIBE_PENDING[2]
```

Sample Output for the debug ccsip all Command: Originating Gateway

```
Router# debug ccsip all
```

```
*Mar 1 01:45:53.783: Sent:
INVITE sip:debbie@example.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.109:5060
From: sip:nobody;tag=60F374-1061
To: sip:debbie@example.com
Date: Mon, 01 Mar 1993 01:45:53 GMT
Call-ID: 52F25057-14FD11CC-802B86FA-EE2DDC42@10.1.1.109
Subject: HelloSipTCL
AccountInfo: 123123
Priority: Urgent
testID: AL_FEAT_SIP_URL_O_RV_11
```



```
Supported: timer,100rel
Min-SE: 1800
Cisco-Guid: 1145332256-352129484-2150139642-3995982914
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE, NOTIFY, INFO
CSeq: 101 INVITE
Max-Forwards: 6
Remote-Party-ID: <sip:50006@10.1.1.109>;party=calling;screen=no;privacy=off
Timestamp: 730950353
Contact: <sip:50006@10.1.1.109:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 2819 5222 IN IP4 10.1.1.109
s=SIP Call
c=IN IP4 10.1.1.109
t=0 0
m=audio 16488 RTP/AVP 0 100
c=IN IP4 10.1.1.109
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
```

Sample Output for the debug ccsip all Command: Terminating Gateway

```
*Jan 26 00:15:39.250: Received:
INVITE sip:debbie@example.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.109:5060
From: sip:nobody;tag=60F374-1061
To: sip:debbie@example.com
Date: Mon, 01 Mar 1993 01:45:53 GMT
Call-ID: 52F25057-14FD11CC-802B86FA-EE2DDC42@10.1.1.109
Subject: HelloSipTCL
AccountInfo: 123123
Priority: Urgent
testID: AL_FEAT_SIP_URL_O_RV_11
Supported: timer,100rel
Min-SE: 1800
Cisco-Guid: 1145332256-352129484-2150139642-3995982914
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, Refer , SUBSCRIBE, NOTIFY, INFO
CSeq: 101 INVITE
Max-Forwards: 6
Remote-Party-ID: <sip:50006@10.1.1.109>;party=calling;screen=no;privacy=off
Timestamp: 730950353
Contact: <sip:50006@10.1.1.109:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 2819 5222 IN IP4 10.1.1.109
s=SIP Call
c=IN IP4 10.1.1.109
t=0 0
m=audio 16488 RTP/AVP 0 100
c=IN IP4 10.1.1.109
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
```

```
a=fmtp:100 192-194
a=ptime:20
```

Sample Output for the debug ccsip messages Command

The following shows sample output for one side of a call.

```
Router# debug ccsip messages
```

```
SIP Call messages tracing is enabled
Router#
*Mar 6 14:19:14: Sent:
INVITE sip:3660210@192.0.2.231;user=phone;phone-context=unknown SIP/2.0
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Sat, 06 Mar 1993 19:19:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Cisco-Guid: 2881152943-2184249568-0-483551624
User-Agent: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 731427554
Contact: <sip:3660110@192.0.2.230:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 138
v=0
o=CiscoSystemsSIP-GW-UserAgent 5596 7982 IN IP4 192.0.2.230
s=SIP Call
t=0 0
c=IN IP4 192.0.2.230
m=audio 20762 RTP/AVP 0
*Mar 6 14:19:14: Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Content-Length: 0
*Mar 6 14:19:14: Received:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 138
v=0
o=CiscoSystemsSIP-GW-UserAgent 1193 7927 IN IP4 192.0.2.231
s=SIP Call
t=0 0
c=IN IP4 192.0.2.231
m=audio 20224 RTP/AVP 0
*Mar 6 14:19:16: Received:
SIP/2.0 200 OK
```

```

Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Contact: <sip:3660210@192.0.2.231:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 138
v=0
o=CiscoSystemsSIP-GW-UserAgent 1193 7927 IN IP4 192.0.2.231
s=SIP Call
t=0 0
c=IN IP4 192.0.2.231
m=audio 20224 RTP/AVP 0
*Mar 6 14:19:16: Sent:
ACK sip:3660210@192.0.2.231:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
Date: Sat, 06 Mar 1993 19:19:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Max-Forwards: 6
Content-Type: application/sdp
Content-Length: 138
CSeq: 101 ACK
v=0
o=CiscoSystemsSIP-GW-UserAgent 5596 7982 IN IP4 192.0.2.230
s=SIP Call
t=0 0
c=IN IP4 192.0.2.230
m=audio 20762 RTP/AVP 0
*Mar 6 14:19:19: Received:
BYE sip:3660110@192.0.2.230:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.0.2.231:53600
From: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
To: "3660110" <sip:3660110@192.0.2.230>
Date: Mon, 08 Mar 1993 22:45:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
User-Agent: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Max-Forwards: 6
Timestamp: 731612717
CSeq: 101 BYE
Content-Length: 0
*Mar 6 14:19:19: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.231:53600
From: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
To: "3660110" <sip:3660110@192.0.2.230>
Date: Sat, 06 Mar 1993 19:19:19 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Timestamp: 731612717
Content-Length:
CSeq: 101 BYE

```

The following shows sample output for the other side of the call.

```

Router# debug ccsip messages

SIP Call messages tracing is enabled
Router#

```

```

*Mar 8 17:45:12: Received:
INVITE sip:3660210@192.0.2.231;user=phone;phone-context=unknown SIP/2.0
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Sat, 06 Mar 1993 19:19:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Cisco-Guid: 2881152943-2184249568-0-483551624
User-Agent: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 731427554
Contact: <sip:3660110@192.0.2.230:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 138
v=0
o=CiscoSystemsSIP-GW-UserAgent 5596 7982 IN IP4 192.0.2.230
s=SIP Call
t=0 0
c=IN IP4 192.0.2.230
m=audio 20762 RTP/AVP 0
*Mar 8 17:45:12: Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Content-Length: 0
*Mar 8 17:45:12: Sent:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 138
v=0
o=CiscoSystemsSIP-GW-UserAgent 1193 7927 IN IP4 192.0.2.231
s=SIP Call
t=0 0
c=IN IP4 192.0.2.231
m=audio 20224 RTP/AVP 0
*Mar 8 17:45:14: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
Date: Mon, 08 Mar 1993 22:45:12 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Timestamp: 731427554
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Contact: <sip:3660210@192.0.2.231:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 138

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 1193 7927 IN IP4 192.0.2.231
s=SIP Call
t=0 0
c=IN IP4 192.0.2.231
m=audio 20224 RTP/AVP 0
*Mar 8 17:45:14: Received:
ACK sip:3660210@192.0.2.231:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.0.2.230:55820
From: "3660110" <sip:3660110@192.0.2.230>
To: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
Date: Sat, 06 Mar 1993 19:19:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Max-Forwards: 6
Content-Type: application/sdp
Content-Length: 138
CSeq: 101 ACK
v=0
o=CiscoSystemsSIP-GW-UserAgent 5596 7982 IN IP4 192.0.2.230
s=SIP Call
t=0 0
c=IN IP4 192.0.2.230
m=audio 20762 RTP/AVP 0
*Mar 8 17:45:17: Sent:
BYE sip:3660110@192.0.2.230:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.0.2.231:53600
From: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
To: "3660110" <sip:3660110@192.0.2.230>
Date: Mon, 08 Mar 1993 22:45:14 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
User-Agent: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Max-Forwards: 6
Timestamp: 731612717
CSeq: 101 BYE
Content-Length: 0
*Mar 8 17:45:17: Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.231:53600
From: <sip:3660210@192.0.2.231;user=phone;phone-context=unknown>;tag=27DBC6D8-1357
To: "3660110" <sip:3660110@192.0.2.230>
Date: Sat, 06 Mar 1993 19:19:19 GMT
Call-ID: ABBAE7AF-823100E2-0-1CD274BC@172.18.192.194
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Timestamp: 731612717
Content-Length: 0
CSeq: 101 BYE

```

Sample Output for the debug isdn q931 Command

The following shows sample output of a call-setup procedure for an outgoing call.

```

Router# debug isdn q931

Router# debug isdn q931
TX -> SETUP pd = 8 callref = 0x04
Bearer Capability i = 0x8890
Channel ID i = 0x83
Called Party Number i = 0x80, `415555121202'
RX <- CALL_PROC pd = 8 callref = 0x84
Channel ID i = 0x89
RX <- CONNECT pd = 8 callref = 0x84
TX -> CONNECT_ACK pd = 8 callref = 0x04....
Success rate is 0 percent (0/5)

```

The following shows sample output of a call-setup procedure for an incoming call.

```
Router# debug isdn q931

RX <- SETUP pd = 8 callref = 0x06
Bearer Capability i = 0x8890
Channel ID i = 0x89
Calling Party Number i = 0x0083, `81012345678902'
TX -> CONNECT pd = 8 callref = 0x86
RX <- CONNECT_ACK pd = 8 callref = 0x06
```

The following shows sample output of a call teardown procedure from the network.

```
Router# debug isdn q931

RX <- DISCONNECT pd = 8 callref = 0x84
Cause i = 0x8790
Looking Shift to Codeset 6
Codeset 6 IE 0x1 1 0x82 `10'
TX -> RELEASE pd = 8 callref = 0x04
Cause i = 0x8090
RX <- RELEASE_COMP pd = 8 callref = 0x84
```

The following shows sample output of a call teardown procedure from the router.

```
Router# debug isdn q931

TX -> DISCONNECT pd = 8 callref = 0x05
Cause i = 0x879081
RX <- RELEASE pd = 8 callref = 0x85
Looking Shift to Codeset 6
Codeset 6 IE 0x1 1 0x82 `10'
TX <- RELEASE_COMP pd = 8 callref = 0x05
```

Sample Output for the debug voip ccapi protoheaders Command: Originating Gateway

```
Router# debug voip ccapi protoheaders
voip ccAPI protocol headers/bodies passing info debugging is on

*Mar 1 01:23:14.711: //-1/xxxxxxxxxxxx/CCAPI/ccGetUriDataFromTDCContainer: urlp=642D8EF0,
urlp->original_url=sip:debbie@example.com?Subject=Hello&Priority=Urgent&testID=AL_FEAT_SIP
_URL_O_RV_11
*Mar 1 01:23:14.711: //-1/xxxxxxxxxxxx/CCAPI/ccSetupReqDataTDFreeHelper: data=6472C678
*Mar 1 01:23:25.155: //-1/xxxxxxxxxxxx/CCAPI/ccSetupReqDataTDFreeHelper: data=632FFD54
```

Sample Output for the debug voip ccapi protoheaders Command: Terminating Gateway

```
*Jan 25 23:53:00.102: //-1/xxxxxxxxxxxx/CCAPI/ccGetUriDataFromTDCContainer: urlp=63CFFCD4,
urlp->original_url=sip:nobody;tag=4C3670-14E3
*Jan 25 23:53:00.102: //-1/xxxxxxxxxxxx/CCAPI/ccGetUriDataFromTDCContainer: urlp=652DAF54,
urlp->original_url=sip:debbie@example.com:5060
*Jan 25 23:53:00.110: //-1/xxxxxxxxxxxx/CCAPI/ccGetUriDataFromTDCContainer: urlp=63CFFCD4,
urlp->original_url=sip:nobody;tag=4C3670-14E3
*Jan 25 23:53:00.110: //-1/xxxxxxxxxxxx/CCAPI/ccGetUriDataFromTDCContainer: urlp=652DAF54,
urlp->original_url=sip:debbie@example.com:5060
*Jan 25 23:53:00.122: //148/256F0CED801A/CCAPI/ccGetAvlistProtoHeader: tag=35,
reqHeader=64417738, reqCount=1, sess_protocol=SIP
```

Sample Output for the debug voip ivr script Command

In the following example, the script fails because the application that is specified in the notificationReceiver field in the script is not configured on the gateway with the **call application voice** command:

```
Router# debug voip ivr script
```

```
*Mar 1 02:44:24.927: //73//TCL2:/TclInterpDriver: Tcl_Eval Failed in action=act_Setup
code=1
code=ERROR
*Mar 1 02:44:24.927: IVR TCL script failure
    Result:
        notifyRecr is not a configured application. Processing
subscriptionInfo array failed.
*Mar 1 02:44:24.927: IVR TCL script failure errorInfo:
    notifyRecr is not a configured application. Processing
subscriptionInfo array failed.
    while executing
"subscription open sip:anglee@sip-server1 subinfo"
    invoked from within
"set subscription_id [subscription open sip:anglee@sip-server1 subinfo]..."
    (procedure "subscribeService" line 49)
    invoked from within
"subscribeService"
    (procedure "act_Setup" line 44)
    invoked from within
"act_Setup"
```

Configuration Examples for SIP Message, Timer, and Response Features

This section provides the following configuration examples:

- [Internal Cause Code Consistency Between SIP and H.323: Example, page 127](#)
- [SIP - Configurable PSTN Cause Code Mapping: Example, page 130](#)
- [SIP Accept-Language Header Support: Examples, page 132](#)
- [SIP Extensions for Caller Identity and Privacy: Example, page 132](#)
- [SIP Session Timer Support: Example, page 134](#)
- [SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion: Examples, page 135](#)
- [SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers: Examples, page 151](#)
- [SIP: Domain Name Support in SIP Headers: Examples, page 154](#)
- [SIP Gateway Support for Permit Hostname: Example, page 155](#)
- [Outbound-Proxy Support for the SIP Gateway: Examples, page 155](#)
- [SIP: SIP Support for PAI: Examples, page 156](#)
- [SIP History-Info Header Support: Examples, page 157](#)

Internal Cause Code Consistency Between SIP and H.323: Example

This example shows a H.323 and SIP configuration with the **cause-code legacy** command configured. The **cause-code legacy** command sets internal failures with nonstandard H.323 or SIP cause codes. The **cause-code legacy** command is generally used for backward compatibility purposes, as standard cause codes are used by default.

```

Router# show running-config

Building configuration...
Current configuration : 4271 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
enable password %1$5dQA*@
!
voice-card 3
!
ip subnet-zero
!
ip domain-name example.com
ip name-server 10.100.0.40
!
isdn switch-type primary-net5
!
voice service voip
cause-code legacy
h323
call start slow
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
ccm-manager mgcp
!
controller E1 3/0
pri-group timeslots 1-31
!
controller E1 3/1
pri-group timeslots 1-31
!
interface FastEthernet0/0
ip address 10.102.0.33 255.255.255.0
duplex auto
speed auto
!
interface FastEthernet0/1
ip address 10.101.0.33 255.255.255.0
duplex auto
speed auto
h323-gateway voip interface
h323-gateway voip id gatekeeper31 ipaddr 10.101.0.35 1718
h323-gateway voip h323-id gateway31
h323-gateway voip tech-prefix 1#
!
interface Serial3/0:15
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
no cdp enable
!
interface Serial3/1:15
no ip address
no logging event link-status
isdn switch-type primary-net5

```



```
isdn incoming-voice voice
no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.102.0.1
ip http server
ip pim bidir-enable
!
call rsvp-sync
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 3/0:15
!
voice-port 3/1:15
!
mgcp ip qos dscp cs5 media
mgcp ip qos dscp cs3 signaling
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1001096 pots
destination-pattern 1001096
port 1/0/0
!
dial-peer voice 1001097 pots
destination-pattern 1001097
port 1/0/1
!
dial-peer voice 1003000 pots
destination-pattern 100300..
port 3/1:15
!
dial-peer voice 1003100 pots
destination-pattern 10031..
port 3/1:15
!
dial-peer voice 2000000 voip
destination-pattern 2.....
session protocol sipv2
session target ipv4:10.101.0.40
!
dial-peer voice 3000000 voip
destination-pattern 3.....
session protocol sipv2
session target sip-server
!
dial-peer voice 4000000 voip
destination-pattern 4.....
session target ras
!
gateway
!
sip-ua
sip-server ipv4:10.100.0.40
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
```

```

session-timeout 20
password password
login
!
end

```

SIP - Configurable PSTN Cause Code Mapping: Example

This examples shows the two commands that change the standard mappings between the SIP and PSTN networks. The **set sip-status** command and **set pstn-cause** command are highlighted in the following configuration.

```

Router# show running-config

Building configuration...

Current configuration : 1564 bytes
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3660-1
!
clock timezone GMT 0
voice-card 1
!
ip subnet-zero
!
ip domain-name example.sip.com
ip name-server 10.10.1.8
!
isdn switch-type primary-5ess
!
voice service voip
    sip
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
controller T1 1/0
    framing esf
    linecode b8zs
    ds0-group 0 timeslots 1-24 type e&m-wink-start
    ds0 busyout 2-24
!
controller T1 1/1
    framing sf
    linecode ami
!
interface FastEthernet0/0
    no ip address
    shutdown
    duplex auto
    speed auto
!
interface FastEthernet0/1

```

```
ip address 10.10.1.3 255.255.255.0
duplex auto
speed auto
ip rsvp bandwidth 75000 75000
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/1
ip http server
ip pim bidir-enable
!
call rsvp-sync
!
voice-port 1/0:0
  output attenuation 3
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 3640110 voip
  application session
  incoming called-number 3640110
  destination-pattern 3640110
  rtp payload-type nte 102
  session protocol sipv2
  session target ipv4:10.10.1.4
  dtmf-relay rtp-nte
  codec g711ulaw
!
dial-peer voice 3660110 pots
  application session
  destination-pattern 3660110
  port 2/0/0
!
sip-ua
  set sip-status 486 pstn-cause 34
  set pstn-cause 17 sip-status 503
  no oli
!
line con 0
  exec-timeout 0 0
line aux 0
line vty 0 4
  login
!
end
```

SIP Accept-Language Header Support: Examples

The following provides partial output for SIP Accept-Language Header Support configured in voice service configuration mode and dial-peer configuration mode.

```
Router# show running-config

Building configuration...
Current configuration :2791 bytes
.
.
.
voice service pots
  supported-language yo
  supported-language sd language-param 0.234
  supported-language fr language-param 0.123
.
.
.
end!
```

```
Router# show running-config

Building configuration...
Current configuration :2791 bytes
.
.
.
dial-peer voice 1 pots
  application session
  destination-pattern 36601
  port 2/0/0
  supported-language sd
  supported-language zu
  supported-language ln language-param 0.123
.
.
.
end!
```

SIP Extensions for Caller Identity and Privacy: Example

In the following example, the PSTN name is set to Company A and the PSTN number is set to 5550111.

```
Router(config-sip-ua)# calling-info sip-to-pstn name set CompanyA
Router(config-sip-ua)# calling-info sip-to-pstn number set 5550111
!
Router# show running-config

Building configuration...

Current configuration :2791 bytes
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3640
!
voice-card 2
```

```
!
ip subnet-zero
!
no ip domain lookup
ip domain name example.com
ip name-server 172.18.195.113
!
isdn switch-type primary-ni
!
fax interface-type fax-mail
mta receive maximum-recipients 0
ccm-manager mgcp
!
controller T1 2/0
  framing esf
  linecode b8zs
  pri-group timeslots 1-24
!
controller T1 2/1
  framing esf
  linecode b8zs
  pri-group timeslots 1-24
!
interface Ethernet0/0
  ip address 172.18.197.22 255.255.255.0
  half-duplex
!
interface Serial0/0
  no ip address
  shutdown
!
interface TokenRing0/0
  no ip address
  shutdown
  ring-speed 16
!
interface FastEthernet1/0
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface Serial2/0:23
  no ip address
  no logging event link-status
  isdn switch-type primary-ni
  isdn incoming-voice voice
  isdn outgoing display-ie
  no cdp enable
!
interface Serial2/1:23
  no ip address
  no logging event link-status
  isdn switch-type primary-ni
  isdn incoming-voice voice
  isdn outgoing display-ie
  no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 Ethernet0/0
no ip http server
ip pim bidir-enable
!
call rsvp-sync
```

```

!
voice-port 2/0:23
!
voice-port 2/1:23
!
voice-port 3/0/0
!
voice-port 3/0/1
!
mgcp ip qos dscp cs5 media
mgcp ip qos dscp cs3 signaling
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 voip
  incoming called-number 5552222
  destination-pattern 5552222
  session protocol sipv2
  session target ipv4:172.18.197.27
!
dial-peer voice 2 pots
  destination-pattern 5551111
  no digit-strip
  direct-inward-dial
  port 2/0:23
!
gateway
!
sip-ua
  calling-info sip-to-pstn name set CompanyA
  calling-info sip-to-pstn number set 5550111
!
line con 0
line aux 0
line vty 0 4
  login
!
end!

```

SIP Session Timer Support: Example

This example contains partial output showing that the Min-SE value has been changed from its default value. If the default value of 90 seconds remains unchanged, configuration data is not provided.

```

Router# show running-config
.
.
.
!
voice service voip
  sip
    min-se 950
!
.
.
.

```

SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion: Examples

This section provides the following configuration examples:

- [Reason Header Enabled, page 135](#)
- [Reason Header Disabled, page 139](#)
- [Buffer Calling Completion Enabled, page 143](#)
- [Buffer Calling Completion Disabled, page 147](#)

Reason Header Enabled

The following examples shows output for the **show running-config** command with reason header enabled.

```
Current configuration :4643 bytes
!
version 12.3
no parser cache
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service internal
!
hostname cartman
!
boot-start-marker
boot-end-marker
!
no logging buffered
enable secret 5 $1$exgC$6yn9bof7/cYMhpNH9DfOp/
enable password password1
!
username 4444
username 232
username username1 password 0 password2
clock timezone EST -5
aaa new-model
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
ip domain name example.sip.com
ip name-server 172.18.192.48
!
ip dhcp pool 1
  host 172.18.193.173 255.255.255.0
  client-identifier 0030.94c2.5d00
  option 150 ip 172.18.193.98
  default-router 172.18.193.98
!
no scripting tcl init
no scripting tcl encdir
!
voice call carrier capacity active
!
```

```

voice service pots
!
voice service voip
  sip
    rellxx disable
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 5 g726r16
  codec preference 6 g726r24
  codec preference 7 g726r32
  codec preference 8 g723ar53
  codec preference 9 g723ar63
!
fax interface-type fax-mail
!
translation-rule 100
!
interface FastEthernet0/0
  ip address 172.18.193.98 255.255.255.0
  duplex auto
  speed auto
  no cdp enable
  ip rsvp bandwidth 75000 75000
!
interface FastEthernet0/1
  ip address 10.1.1.98 255.0.0.0
  shutdown
  duplex auto
  speed auto
  no cdp enable
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 10.0.0.0 255.0.0.0 172.18.193.1
ip route 172.18.0.0 255.255.0.0 172.18.193.1
!
ip radius source-interface FastEthernet0/0
logging source-interface FastEthernet0/0
dialer-list 1 protocol ip permit
snmp-server engineID local 00000009020000309426F6D0
snmp-server community public RO
snmp-server community private RW
snmp-server packetsize 4096
snmp-server enable traps tty
!
tftp-server flash:XMLDefault.cnf.xml
!
radius-server host 172.18.192.108 auth-port 1645 acct-port 1646
radius-server retransmit 1
radius-server key lab
radius-server vsa send accounting
radius-server vsa send authentication
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
  station-id number 36601

```



```
    caller-id enable
  !
  voice-port 1/1/1
  !
  voice-port 2/0/0
    caller-id enable
  !
  voice-port 2/0/1
  !
  voice-port 2/1/0
  !
  voice-port 2/1/1
  !
  mgcp
  mgcp sdp simple
  !
  dial-peer cor custom
  !
  dial-peer voice 6 voip
    destination-pattern 36602
    session protocol sipv2
    session target ipv4:10.102.17.80
    session transport tcp
    incoming called-number 36601
    codec g711ulaw
  !
  dial-peer voice 5 voip
    application session
    destination-pattern 5550123
    session protocol sipv2
    session target ipv4:172.18.197.182
  !
  dial-peer voice 1 pots
    destination-pattern 36601
    port 2/0/0
  !
  dial-peer voice 38 voip
    application session
    destination-pattern 3100802
    voice-class codec 1
    session protocol sipv2
    session target ipv4:172.18.193.99
    dtmf-relay cisco-rtp
  !
  dial-peer voice 81 voip
    application session
    destination-pattern 3100801
    session protocol sipv2
    session target ipv4:172.18.193.100
    dtmf-relay rtp-nte
  !
  dial-peer voice 41 voip
    application session
    destination-pattern 777
    session protocol sipv2
    session target ipv4:172.18.199.94
    session transport udp
  !
  dial-peer voice 7 voip
    application session
    destination-pattern 999
    session protocol sipv2
    session target ipv4:172.18.193.98
    incoming called-number 999
```

```

!
dial-peer voice 2 pots
 destination-pattern 361
 port 2/1/1
!
dial-peer voice 55 voip
 destination-pattern 5678
 session protocol sipv2
 session target ipv4:192.0.2.208:5061
!
dial-peer voice 361 voip
 incoming called-number 361
!
dial-peer voice 100 voip
!
dial-peer voice 3 pots
 destination-pattern 36601
 port 2/0/1
!
dial-peer voice 111 pots
!
dial-peer voice 11 pots
 preference 5
 destination-pattern 123
 port 2/0/0
!
dial-peer voice 12 pots
 destination-pattern 456
 port 2/0/0
!
dial-peer voice 14 pots
 destination-pattern 980
 port 2/0/0
!
dial-peer voice 15 pots
 destination-pattern 789
 port 2/0/0
!
gateway
!
sip-ua
 retry invite 2
 retry response 4
 retry bye 2
 retry cancel 1
 timers expires 300000
 sip-server dns:example-srv.sip.com
 reason-header override ! reason header enabled
!
telephony-service
 mwi relay
 mwi expires 600
 max-conferences 8
!
banner motd ^Chello^C
!
line con 0
 exec-timeout 0 0
 transport preferred all
 transport output all
line aux 0
 transport preferred all
 transport output all
line vty 0 4

```

```

password password1
transport preferred all
transport input all
transport output all
!
end

```

The following shows output for the **show sip-ua status** command with reason header enabled.

```

SIP User Agent Status
SIP User Agent for UDP :ENABLED
SIP User Agent for TCP :ENABLED
SIP User Agent bind status(signaling):DISABLED
SIP User Agent bind status(media):DISABLED
SIP early-media for 180 responses with SDP:ENABLED
SIP max-forwards :70
SIP DNS SRV version:2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP:NONE
Check media source packets:DISABLED
Maximum duration for a telephone-event in NOTIFYs:2000 ms
SIP support for ISDN SUSPEND/RESUME:ENABLED
Redirection (3xx) message handling:ENABLED
Reason Header will override Response/Request Codes:ENABLED ! Reader Header Enabled

SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported:audio image
Network types supported:IN
Address types supported:IP4
Transport types supported:RTP/AVP udpt1

```

Reason Header Disabled

The following shows output for the **show running-config** command with reason header disabled.

```

Current configuration :4619 bytes
!
version 12.3
no parser cache
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service internal
!
hostname Router
!
boot-start-marker
boot-end-marker
!
no logging buffered
enable secret 5 $1$exgC$6yn9bof7/cYMhpNH9DfOp/
enable password password1
!
username 4444
username 232
username username1 password 0 password2
clock timezone EST -5
aaa new-model
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius

```

```

aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
ip domain name example.sip.com
ip name-server 172.18.192.48
!
ip dhcp pool 1
  host 172.18.193.173 255.255.255.0
  client-identifier 0030.94c2.5d00
  option 150 ip 172.18.193.98
  default-router 172.18.193.98
!
no scripting tcl init
no scripting tcl encdir
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  sip
  rel1xx disable
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 5 g726r16
  codec preference 6 g726r24
  codec preference 7 g726r32
  codec preference 8 g723ar53
  codec preference 9 g723ar63
!
fax interface-type fax-mail
!
translation-rule 100
!
interface FastEthernet0/0
  ip address 172.18.193.98 255.255.255.0
  duplex auto
  speed auto
  no cdp enable
  ip rsvp bandwidth 75000 75000
!
interface FastEthernet0/1
  ip address 10.1.1.98 255.0.0.0
  shutdown
  duplex auto
  speed auto
  no cdp enable
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 10.0.0.0 255.0.0.0 172.18.193.1
ip route 172.18.0.0 255.255.0.0 172.18.193.1
!
ip radius source-interface FastEthernet0/0
logging source-interface FastEthernet0/0
dialer-list 1 protocol ip permit
snmp-server engineID local 00000009020000309426F6D0
snmp-server community public RO
snmp-server community private RW

```

```
snmp-server packetsize 4096
snmp-server enable traps tty
!
tftp-server flash:XMLDefault.cnf.xml
!
radius-server host 172.18.192.108 auth-port 1645 acct-port 1646
radius-server retransmit 1
radius-server key lab
radius-server vsa send accounting
radius-server vsa send authentication
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
  station-id number 36601
  caller-id enable
!
voice-port 1/1/1
!
voice-port 2/0/0
  caller-id enable
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
mgcp
mgcp sdp simple
!
dial-peer cor custom
!
dial-peer voice 6 voip
  destination-pattern 36602
  session protocol sipv2
  session target ipv4:10.102.17.80
  session transport tcp
  incoming called-number 36601
  codec g711ulaw
!
dial-peer voice 5 voip
  application session
  destination-pattern 5550123
  session protocol sipv2
  session target ipv4:172.18.197.182
!
dial-peer voice 1 pots
  destination-pattern 36601
  port 2/0/0
!
dial-peer voice 38 voip
  application session
  destination-pattern 3100802
  voice-class codec 1
  session protocol sipv2
  session target ipv4:172.18.193.99
  dtmf-relay cisco-rtp
!
dial-peer voice 81 voip
```

```

application session
destination-pattern 3100801
session protocol sipv2
session target ipv4:172.18.193.100
dtmf-relay rtp-nte
!
dial-peer voice 41 voip
application session
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
session transport udp
!
dial-peer voice 7 voip
application session
destination-pattern 999
session protocol sipv2
session target ipv4:172.18.193.98
incoming called-number 999
!
dial-peer voice 2 pots
destination-pattern 361
port 2/1/1
!
dial-peer voice 55 voip
destination-pattern 5678
session protocol sipv2
session target ipv4:10.102.17.208:5061
!
dial-peer voice 361 voip
incoming called-number 361
!
dial-peer voice 100 voip
!
dial-peer voice 3 pots
destination-pattern 36601
port 2/0/1
!
dial-peer voice 111 pots
!
dial-peer voice 11 pots
preference 5
destination-pattern 123
port 2/0/0
!
dial-peer voice 12 pots
destination-pattern 456
port 2/0/0
!
dial-peer voice 14 pots
destination-pattern 980
port 2/0/0
!
dial-peer voice 15 pots
destination-pattern 789
port 2/0/0
!
gateway
!
sip-ua
retry invite 2
retry response 4
retry bye 2
retry cancel 1

```

```

timers expires 300000
sip-server dns:example-srv.sip.com
!
telephony-service
mwi relay
mwi expires 600
max-conferences 8
!
banner motd ^Chello^C
!
line con 0
exec-timeout 0 0
transport preferred all
transport output all
line aux 0
transport preferred all
transport output all
line vty 0 4
password password1
transport preferred all
transport input all
transport output all
!
end

```

The following shows output for the **show sip-ua status** command with reason header disabled.

```

SIP User Agent Status
SIP User Agent for UDP :ENABLED
SIP User Agent for TCP :ENABLED
SIP User Agent bind status(signaling):DISABLED
SIP User Agent bind status(media):DISABLED
SIP early-media for 180 responses with SDP:ENABLED
SIP max-forwards :70
SIP DNS SRV version:2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP:NONE
Check media source packets:DISABLED
Maximum duration for a telephone-event in NOTIFYs:2000 ms
SIP support for ISDN SUSPEND/RESUME:ENABLED
Redirection (3xx) message handling:ENABLED
Reason Header will override Response/Request Codes:DISABLED ! Reason Header Disabled

SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported:audio image
Network types supported:IN
Address types supported:IP4
Transport types supported:RTP/AVP udpt1

```

Buffer Calling Completion Enabled

```

Current configuration :4646 bytes
!
version 12.3
no parser cache
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service internal
!
hostname Router

```

```

!
boot-start-marker
boot-end-marker
!
no logging buffered
enable secret 5 $1$exgC$6yn9bof7/cYMhpNH9DfOp/
enable password password1
!
username 4444
username 232
username username1 password 0 password2
clock timezone EST -5
aaa new-model
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
ip domain name example.sip.com
ip name-server 172.18.192.48
!
ip dhcp pool 1
  host 172.18.193.173 255.255.255.0
  client-identifier 0030.94c2.5d00
  option 150 ip 172.18.193.98
  default-router 172.18.193.98
!
no scripting tcl init
no scripting tcl encdir
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  sip
  rel1xx disable
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 5 g726r16
  codec preference 6 g726r24
  codec preference 7 g726r32
  codec preference 8 g723ar53
  codec preference 9 g723ar63
!
fax interface-type fax-mail
!
translation-rule 100
!
interface FastEthernet0/0
  ip address 172.18.193.98 255.255.255.0
  duplex auto
  speed auto
  no cdp enable
  ip rsvp bandwidth 75000 75000
!
interface FastEthernet0/1
  ip address 10.1.1.98 255.0.0.0
  shutdown

```



```
duplex auto
speed auto
no cdp enable
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 10.0.0.0 255.0.0.0 172.18.193.1
ip route 172.18.0.0 255.255.0.0 172.18.193.1
!
ip radius source-interface FastEthernet0/0
logging source-interface FastEthernet0/0
dialer-list 1 protocol ip permit
snmp-server engineID local 00000009020000309426F6D0
snmp-server community public RO
snmp-server community private RW
snmp-server packetize 4096
snmp-server enable traps tty
!
tftp-server flash:XMLDefault.cnf.xml
!
radius-server host 172.18.192.108 auth-port 1645 acct-port 1646
radius-server retransmit 1
radius-server key lab
radius-server vsa send accounting
radius-server vsa send authentication
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
station-id number 36601
caller-id enable
!
voice-port 1/1/1
!
voice-port 2/0/0
caller-id enable
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
mgcp
mgcp sdp simple
!
dial-peer cor custom
!
dial-peer voice 6 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:10.102.17.80
session transport tcp
incoming called-number 36601
codec g711ulaw
!
dial-peer voice 5 voip
application session
destination-pattern 5550123
```

```

    session protocol sipv2
    session target ipv4:172.18.197.182
    !
dial-peer voice 1 pots
    destination-pattern 36601
    port 2/0/0
    !
dial-peer voice 38 voip
    application session
    destination-pattern 3100802
    voice-class codec 1
    session protocol sipv2
    session target ipv4:172.18.193.99
    dtmf-relay cisco-rtp
    !
dial-peer voice 81 voip
    application session
    destination-pattern 3100801
    session protocol sipv2
    session target ipv4:172.18.193.100
    dtmf-relay rtp-nte
    !
dial-peer voice 41 voip
    application session
    destination-pattern 777
    session protocol sipv2
    session target ipv4:172.18.199.94
    session transport udp
    !
dial-peer voice 7 voip
    application session
    destination-pattern 999
    session protocol sipv2
    session target ipv4:172.18.193.98
    incoming called-number 999
    !
dial-peer voice 2 pots
    destination-pattern 361
    port 2/1/1
    !
dial-peer voice 55 voip
    destination-pattern 5678
    session protocol sipv2
    session target ipv4:10.102.17.208:5061
    !
dial-peer voice 361 voip
    incoming called-number 361
    !
dial-peer voice 100 voip
    !
dial-peer voice 3 pots
    destination-pattern 36601
    port 2/0/1
    !
dial-peer voice 111 pots
    !
dial-peer voice 11 pots
    preference 5
    destination-pattern 123
    port 2/0/0
    !
dial-peer voice 12 pots
    destination-pattern 456
    port 2/0/0

```

```

!
dial-peer voice 14 pots
 destination-pattern 980
 port 2/0/0
!
dial-peer voice 15 pots
 destination-pattern 789
 port 2/0/0
!
gateway
!
sip-ua
 retry invite 2
 retry response 4
 retry bye 2
 retry cancel 1
 timers expires 300000
 timers buffer-invite 5000 ! Buffer Calling Completion enabled
 sip-server dns:example-srv.sip.com
!
!
telephony-service
 mwi relay
 mwi expires 600
 max-conferences 8
!
banner motd ^Chello^C
!
line con 0
 exec-timeout 0 0
 transport preferred all
 transport output all
line aux 0
 transport preferred all
 transport output all
line vty 0 4
 password password1
 transport preferred all
 transport input all
 transport output all
!
end

```

Buffer Calling Completion Disabled

```

Current configuration :4619 bytes
!
version 12.3
no parser cache
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
service internal
!
hostname Router
!
boot-start-marker
boot-end-marker
!
no logging buffered
enable secret 5 $1$exgC$6yn9bof7/cYMhpNH9DfOp/
enable password password1
!
username 4444

```

```

username 232
username username1 password 0 password2
clock timezone EST -5
aaa new-model
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
ip domain name example.sip.com
ip name-server 172.18.192.48
!
ip dhcp pool 1
    host 172.18.193.173 255.255.255.0
    client-identifier 0030.94c2.5d00
    option 150 ip 172.18.193.98
    default-router 172.18.193.98
!
no scripting tcl init
no scripting tcl encdir
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
    sip
    rel1xx disable
!
voice class codec 1
    codec preference 1 g729r8
    codec preference 2 g711ulaw
    codec preference 5 g726r16
    codec preference 6 g726r24
    codec preference 7 g726r32
    codec preference 8 g723ar53
    codec preference 9 g723ar63
!
fax interface-type fax-mail
!
translation-rule 100
!
interface FastEthernet0/0
    ip address 172.18.193.98 255.255.255.0
    duplex auto
    speed auto
    no cdp enable
    ip rsvp bandwidth 75000 75000
!
interface FastEthernet0/1
    ip address 10.1.1.98 255.0.0.0
    shutdown
    duplex auto
    speed auto
    no cdp enable
!
ip http server
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 10.0.0.0 255.0.0.0 172.18.193.1
ip route 172.18.0.0 255.255.0.0 172.18.193.1

```

```
!  
ip radius source-interface FastEthernet0/0  
logging source-interface FastEthernet0/0  
dialer-list 1 protocol ip permit  
snmp-server engineID local 00000009020000309426F6D0  
snmp-server community public RO  
snmp-server community private RW  
snmp-server packetsize 4096  
snmp-server enable traps tty  
!  
tftp-server flash:XMLDefault.cnf.xml  
!  
radius-server host 172.18.192.108 auth-port 1645 acct-port 1646  
radius-server retransmit 1  
radius-server key lab  
radius-server vsa send accounting  
radius-server vsa send authentication  
!  
control-plane  
!  
voice-port 1/0/0  
!  
voice-port 1/0/1  
!  
voice-port 1/1/0  
  station-id number 36601  
  caller-id enable  
!  
voice-port 1/1/1  
!  
voice-port 2/0/0  
  caller-id enable  
!  
voice-port 2/0/1  
!  
voice-port 2/1/0  
!  
voice-port 2/1/1  
!  
mgcp  
mgcp sdp simple  
!  
dial-peer cor custom  
!  
dial-peer voice 6 voip  
  destination-pattern 36602  
  session protocol sipv2  
  session target ipv4:192.0.2.80  
  session transport tcp  
  incoming called-number 36601  
  codec g711ulaw  
!  
dial-peer voice 5 voip  
  application session  
  destination-pattern 5550123  
  session protocol sipv2  
  session target ipv4:172.18.197.182  
!  
dial-peer voice 1 pots  
  destination-pattern 36601  
  port 2/0/0  
!  
dial-peer voice 38 voip  
  application session
```

```
destination-pattern 3100802
voice-class codec 1
session protocol sipv2
session target ipv4:172.18.193.99
dtmf-relay cisco-rtp
!
dial-peer voice 81 voip
application session
destination-pattern 3100801
session protocol sipv2
session target ipv4:172.18.193.100
dtmf-relay rtp-nte
!
dial-peer voice 41 voip
application session
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
session transport udp
!
dial-peer voice 7 voip
application session
destination-pattern 999
session protocol sipv2
session target ipv4:172.18.193.98
incoming called-number 999
!
dial-peer voice 2 pots
destination-pattern 361
port 2/1/1
!
dial-peer voice 55 voip
destination-pattern 5678
session protocol sipv2
session target ipv4:10.102.17.208:5061
!
dial-peer voice 361 voip
incoming called-number 361
!
dial-peer voice 100 voip
!
dial-peer voice 3 pots
destination-pattern 36601
port 2/0/1
!
dial-peer voice 111 pots
!
dial-peer voice 11 pots
preference 5
destination-pattern 123
port 2/0/0
!
dial-peer voice 12 pots
destination-pattern 456
port 2/0/0
!
dial-peer voice 14 pots
destination-pattern 980
port 2/0/0
!
dial-peer voice 15 pots
destination-pattern 789
port 2/0/0
!
```

```

gateway
!
sip-ua
  retry invite 2
  retry response 4
  retry bye 2
  retry cancel 1
  timers expires 300000
  sip-server dns:example-srv.sip.com
!
telephony-service
  mwi relay
  mwi expires 600
  max-conferences 8
!
banner motd ^Chello^C
!
line con 0
  exec-timeout 0 0
  transport preferred all
  transport output all
line aux 0
  transport preferred all
  transport output all
line vty 0 4
  password password1
  transport preferred all
  transport input all
  transport output all
!
end

```

SIP: SIP Header/URL Support and SUBSCRIBE/NOTIFY for External Triggers: Examples

SIP Header Support and Subscription

In the following example, header passing is enabled and a default server IP address is configured. The history log is configured to retain 100 history records, each of which is retained for fifteen minutes after the subscription is removed. SUBSCRIBE messages are configured to retransmit six times.

```

Router# show running-config

Building configuration...

version 12.2
no service pad
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
service internal
!
hostname Router
!
subscription asnl session history duration 15
subscription asnl session history count 100
logging buffered 1000000 debugging
!
resource-pool disable
!
ip subnet-zero

```

```

ip host server.example.com 10.7.104.88
!!
isdn switch-type primary-5ess
isdn voice-call-failure 0
!
voice call carrier capacity active
!
voice service voip
    h323
    sip
!!
The Cisco IOS VoiceXML features are enabled, and the maximum number of subscriptions to be
originated by the gateway is configured.

    header-passing
    subscription maximum originate 200
!
mta receive maximum-recipients 0
!
controller T1 0
    framing esf
    clock source line primary
    linecode b8zs
    cablelength short 133
    pri-group timeslots 1-24
!
controller T1 1
    framing sf
    clock source line secondary 1
    linecode ami
!
controller T1 2
    framing sf
    clock source line secondary 2
    linecode ami
!
controller T1 3
    framing sf
    clock source line secondary 3
    linecode ami
!
interface Ethernet0
ip address 10.7.102.35 255.255.0.0
ip helper-address 223.255.254.254
no ip mroute-cache
no cdp enable
!
interface Serial0
no ip address
no ip mroute-cache
shutdown
clockrate 2015232
no fair-queue
no cdp enable
!
interface Serial1
no ip address
no ip mroute-cache
shutdown
clockrate 2015232
no fair-queue
no cdp enable
!
interface Serial2

```



```
no ip address
no ip mroute-cache
shutdown
clockrate 2015232
no fair-queue
no cdp enable
!
interface Serial3
no ip address
no ip mroute-cache
shutdown
clockrate 2015232
no fair-queue
no cdp enable
!
interface Serial0:23
no ip address
ip mroute-cache
dialer-group 1
isdn switch-type primary-5ess
isdn incoming-voice modem
isdn disconnect-cause 1
fair-queue 64 256 0
no cdp enable
!
interface FastEthernet0
ip address 172.19.139.114 255.255.255.0
no ip mroute-cache
duplex auto
speed auto
no cdp enable
!
ip default-gateway 172.19.139.1
ip classless
ip route 172.71.56.39 255.255.255.255 172.19.139.1
ip route 10.255.254.0 255.255.255.0 10.7.104.1
no ip http server
!
ip pim bidir-enable
!
!
no cdp run
!
call application voice mwi tftp://dirt/ramsubra/cli_mwi.tcl
!
voice-port 0:D
!
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
application mwi
destination-pattern 408.....
incoming called-number 52943
port 0:D
prefix 950
!
dial-peer voice 789 voip
destination-pattern 789
session target ipv4:10.7.104.88
codec g711ulaw
```

```

!
dial-peer voice 766 voip
  application get_headers_tcl
  session protocol sipv2
  session target ipv4:10.7.102.35
  incoming uri request 766
  codec g711ulaw
!
dial-peer voice 88888 pots
  destination-pattern 767....
  port 0:D
  prefix 9767
!
sip-ua
  retry subscribe 6
!
line con 0
  exec-timeout 0 0
  logging synchronous level all
line aux 0
line vty 0 4
!
end

```

SIP: Domain Name Support in SIP Headers: Examples

This section provides the following configuration examples:

- [Configuration in Gateway-Wide Global Configuration Mode, page 154](#)
- [Configuration in Dial-Peer-Specific Dial-Peer Configuration Mode, page 155](#)

Configuration in Gateway-Wide Global Configuration Mode

The following example shows the command output when the local hostname uses the gateway-wide global configuration settings.

```

Router# show running-config

Building configuration...

Current configuration :3512 bytes
!
! Last configuration change at 14:25:20 EDT Tue Aug 31 2004
! NVRAM config last updated at 14:17:44 EDT Tue Aug 31 2004
!
version 12.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
! voice service voip
  sip
    localhost dns:example.com
!
!

```

Configuration in Dial-Peer-Specific Dial-Peer Configuration Mode

The following example shows the command output when the local hostname uses the dial-peer configuration settings.

```
Router# show running-config

Building configuration...

Current configuration :3512 bytes
!
! Last configuration change at 14:25:20 EDT Tue Aug 31 2004
! NVRAM config last updated at 14:17:44 EDT Tue Aug 31 2004
!
version 12.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
!
dial-peer voice 3301 voip
destination-pattern 9002
voice-class sip localhost dns:gw11.example.com
session protocol sipv2
session target dns:example.sip.com
session transport tcp
dtmf-relay rtp-nte
!
```

SIP Gateway Support for Permit Hostname: Example

The following example shows a configured list of hostnames.

```
router> enable
router# configure terminal
router (config)# sip-ua
router (config-sip-ua)# permit hostname dns:esample1.sip.com
router (config-sip-ua)# permit hostname dns:example2.sip.com
router (config-sip-ua)# permit hostname dns:example3.sip.com
router (config-sip-ua)# permit hostname dns:example4.sip.com
router (config-sip-ua)# permit hostname dns:example5.sip.com
router (config-sip-ua)# exit
```

Outbound-Proxy Support for the SIP Gateway: Examples

The following example shows how to configure an outbound-proxy server globally on a gateway for the specified IP address:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# outbound-proxy ipv4:10.1.1.1
```

The following example shows how to configure an outbound-proxy server globally on a gateway for the specified domain:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
```

```
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# outbound-proxy dns:sipproxy.example.com
```

The following examples shows how to configure an outbound-proxy server on a dial peer for the specified IP address:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# dial-peer voice 111 voip
gateway(conf-dial-peer)# voice-class sip
gateway(conf-dial-peer)# outbound-proxy ipv4:10.1.1.1
```

The following examples shows how to configure an outbound-proxy server on a dial peer for the specified domain:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# dial-peer voice 111 voip
gateway(conf-dial-peer)# voice-class sip
gateway(conf-dial-peer)# outbound-proxy dns:sipproxy.example.com
```

The following example shows how to disable the global outbound proxy feature for all line-side SIP phones on a Cisco Unified CME:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice register global
gateway(config-register-global)# no outbound-proxy
```

SIP: SIP Support for PAI: Examples

This section contains the following configuration examples:

- [Configuring a Privacy Header: Example, page 156](#)
- [Configuring PPI: Example, page 156](#)
- [Configuring PAI: Example, page 157](#)

Configuring a Privacy Header: Example

The following example shows how to configure a privacy header:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# privacy
```

Configuring PPI: Example

The following example shows how to configure a privacy header level for PPI:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
```

```
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# asserted-id ppi
```

Configuring PAI: Example

The following example shows how to configure a privacy header level for PAI:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# asserted-id pai
```

SIP History-Info Header Support: Examples

The following example shows how to configure history-info header support at the global level:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# history-info
```

The following example shows partial output from the **show running-config** command when history-info header support is configured at the global level:

```
gateway# show running-config

Building configuration...
Current configuration : 10198 bytes
.
.
voice service voip
  fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
  privacy user critical
  history-info
.
.
dial-peer voice 1 voip
  voice-class sip resource priority namespace drsn
  voice-class sip privacy header id critical
!
dial-peer voice 2 voip
  voice-class sip privacy header critical
.
.
end
```

The following example shows how to configure history-info header support at the dial-peer level:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# dial-peer voice 2 voip
gateway(config-dial-peer)# voice-class sip history-info
```

The following example shows partial output from the **show running-config** command when history-info header support is configured at the dial-peer level:

```
gateway# show running-config

Building configuration...
Current configuration : 10183 bytes
.
.
voice service voip
  fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    privacy user critical
.
.
dial-peer voice 1 voip
voice-class sip resource priority namespace drsn
voice-class sip privacy header id critical
!
dial-peer voice 2 voip
  voice-class sip privacy header history critical
.
.
end
```

Additional References

The following sections provide references related to the SIP message, timer, and response features.

Related Documents

Related Topic	Document Title
<i>Cisco IOS SIP Configuration Guide</i> , “Achieving SIP RFC Compliance” module	http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-rfc_comply.html
<i>Cisco IOS SIP Configuration Guide</i> , “Features Roadmap” module	http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-roadmap.html
<i>Cisco IOS SIP Configuration Guide</i> , “Overview of SIP” module	http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-overview.html
<i>Cisco IOS Tcl IVR and VoiceXML Application Guide</i>	Cisco IOS Release 12.3(14)T and later: http://www.cisco.com/en/US/docs/ios/voice/ivr/configuration/guide/tcl_c.html Cisco IOS releases prior to 12.3(14)T: http://www.cisco.com/en/US/docs/ios/voice/ivr/pre12.3_14_t/configuration/guide/ivrapp.pdf
<i>Cisco IOS Voice Command Reference</i>	http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html
<i>Cisco Unified Communications Manager Express Command Reference</i>	http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_cr.html
Cisco Unified Communications Manager Express support documentation	http://www.cisco.com/en/US/products/sw/voicesw/ps4625/tsd_products_support_series_home.html
<i>Cisco Unified SIP SRST System Administrator Guide</i>	http://www.cisco.com/en/US/docs/voice_ip_comm/cusrst/admin/siprst/configuration/guide/sprst41.html
<i>Cisco VoiceXML Programmer’s Guide</i>	http://www.cisco.com/en/US/docs/ios/voice/vxml/developer/guide/vxmlprg.html
<i>Tcl IVR API Version 2.0 Programming Guide</i>	http://www.cisco.com/en/US/docs/ios/voice/tcl/developer/guide/tclivr2.html

Standards

Standard	Title
International Organization for Standardization (ISO) specification, ISO 639	<i>Codes for Representation of Names of Languages</i>

MIBs

MIB	MIBs Link
None	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 2833	<i>RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals</i>
RFC 3261	<i>SIP: Session Initiation Protocol</i>
RFC 3262	<i>Reliability of Provisional Responses in the Session Initiation Protocol (SIP)</i>
RFC 3264	<i>An Offer/Answer Model with the Session Description Protocol (SDP)</i>
RFC 3265	<i>Session Initiation Protocol (SIP)-Specific Event Notification</i>
RFC 3312	<i>Integration of Resource Management and Session Initiation Protocol (SIP)</i>
RFC 3323	<i>A Privacy Mechanism for the Session Initiation Protocol (SIP)</i>
RFC 3325	<i>Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks</i>
RFC 3326	<i>The Reason Header Field for the Session Initiation Protocol (SIP)</i>
RFC 4028	<i>Session Timers in the Session Initiation Protocol (SIP)</i>
RFC 4244	<i>An Extension to the Session Initiation Protocol (SIP) for Request History Information</i>

Technical Assistance

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

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Configuring SIP Bind Features

This chapter describes the SIP Gateway Support for the bind Command feature. With the addition of the **bind** command, you can configure the source IP address of signaling packets or both signaling and media packets.

Feature History for SIP Gateway Support for the bind Command

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB2	This feature was implemented on an additional platform.
12.2(8)T	This feature was implemented on additional platforms.
12.2(11)T	This feature was implemented on additional platforms.
12.3(4)T	Under the name SIP Gateway Support Enhancements to the bind Command, the feature was expanded to provide the flexibility to specify different source interfaces for signaling and media, and allow network administrators a finer granularity of control on the network interfaces used for voice traffic.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Restrictions for SIP Bind Features, page 2](#)
- [Information About SIP Bind Features, page 2](#)
- [How to Configure SIP Bind Features, page 6](#)
- [Configuration Example for SIP Bind Features, page 10](#)
- [Additional References, page 11](#)



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Restrictions for SIP Bind Features

- Although the **bind all** command is an accepted configuration, it does not appear in **show running-config** command output. Because the **bind all** command is equivalent to issuing the commands **bind source** and **bind media**, those are the commands that appear in the **show running-config** command output.

Information About SIP Bind Features



Note

When you configure SIP on a router, the ports on all its interfaces are open by default. This makes the router vulnerable to malicious attackers who can execute toll fraud across the gateway if the router has a public IP address and a public switched telephone network (PSTN) connection. To eliminate the threat, you should bind an interface to an IP address so that only those ports are open to the outside world. In addition, you should protect any public or untrusted interface by configuring a firewall or an access control list (ACL) to prevent unwanted traffic from traversing the router.

Feature benefits include the following:

- SIP signaling and media paths can advertise the same source IP address on the gateway for certain applications, even if the paths used different addresses to reach the source. This eliminates confusion for firewall applications that, before to the use of binding, may have taken action on several different source address packets.
- Firewalls filter messages based on variables such as the message source, the target address, and available ports. Normally a firewall opens only certain addresses or port combination to the outside world and those addresses can change dynamically. Because VoIP technology requires the use of more than one address or port combination, the **bind** command adds flexibility by assigning a gateway to a specific interface (and therefore the associated address) for the signaling or media application.
- You can obtain a predefined and separate interface for both signaling and media traffic. Once a **bind** command is in effect, the interface it limits is bound solely to that purpose. Administrators can therefore dictate the use of one network to transport the signaling and another network to transport the media. The benefits of administrator control are:
 - Administrators know the traffic that run on specific networks, thereby making debugging easier.
 - Administrators know the capacity of the network and the target traffic, thereby making engineering and planning easier.
 - Traffic is controlled, thereby allowing QoS to be monitored.
- The **bind media** command relaxes the constraints imposed by the **bind control** and **bind all** commands, which can not be set during an active call. The **bind media** command works with active calls.

To configure SIP Gateway Support for the bind Command, you should understand the following concepts:

- [Source Address, page 3](#)
- [Voice Media Stream Processing, page 4](#)

Source Address

In early releases of Cisco IOS software with SIP functionality, the source address of a packet going out of the gateway was never deterministic. That is, the session protocols and VoIP layers always depended on the IP layer to give the *best local address*. The best local address was then used as the source address (the address showing where the SIP request came from) for signaling and media packets. Using this nondeterministic address occasionally caused confusion for firewall applications, because a firewall could not be configured with an exact address and would take action on several different source address packets.

However, the **bind** command allows you to configure the source IP address of signaling and media packets to a specific interface's IP address. Thus, the address that goes out on the packet is bound to the IP address of the interface specified with the **bind** command. Packets that are not destined to the bound address are discarded.

When you do not want to specify a bind address or if the interface is down, the IP layer still provides the best local address.

The **bind** command performs different functions based on the state of the interface (see [Table 40](#)).

Table 40 State of the Interface for Bind Command

Interface State	Result Using Bind Command
Shutdown With or without active calls	Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.) Then the sockets are opened to listen to any IP address. If the outgoing gateway has the bind command enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.
No Shutdown No Active Calls	TCP and UDP socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.) Then the sockets are opened and bound to the IP address set by the bind command. The sockets accept packets destined for the bound address only.
No Shutdown Active Calls	TCP and UDP socket listeners are initially closed. Then the sockets are opened to listen to any IP address.
Bound-interface IP address is removed	TCP and UDP socket listeners are initially closed. Then the sockets are opened to listen to any address, because the IP address has been removed. This happens even when SIP was never bound to an IP address. A message stating that the IP address has been deleted from SIP bound interface is printed. If the outgoing gateway has bind enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.

Table 40 *State of the Interface for Bind Command (continued)*

Interface State	Result Using Bind Command
The physical cable is pulled on the bound port, or the Interface layer goes down	TCP and UDP socket listeners are initially closed. Then the sockets are opened and bound to listen to any address. When the pulled cable is replaced, the result is as documented for no shutdown interfaces.
A bind interface is shutdown , or its IP address is changed , or the physical cable is pulled while SIP calls are active	The call becomes a one-way call with media flowing in only one direction. It flows from the gateway where the change or shutdown took place, to the gateway where no change occurred. Thus, the gateway with the status change no longer receives media. The call is then disconnected, but the disconnected message is not understood by the gateway with the status change, and the call is still assumed to be active.
Note	If there are active calls, the bind command does not take effect if it is issued for the first time or while another bind command is in effect. A message reminds you that there are active calls and that the change cannot take effect.

Voice Media Stream Processing

The SIP Gateway Support Enhancements to the bind Command feature extends the current capabilities of the **bind** command by supporting a deterministic network interface for the voice media stream. Before the voice media stream addition, the **bind** command supported a deterministic network interface for control (signaling) traffic or all traffic. With the SIP Gateway Support Enhancements to the bind Command feature a finer granularity of control is achieved on the network interfaces used for voice traffic.

If multiple **bind** commands are issued in sequence—that is, if one **bind** command is configured and then another **bind** command is configured—a set interaction happens between the commands. [Table 41](#) describes the expected command behavior.

Table 41 *Interaction Between Previously Set and New bind Commands*

Interface State	bind Command	Result Using bind Command
Without active calls	bind all	New bind control and bind media commands are generated to override any existing bind control and bind media commands.
	bind control	Overrides any existing bind control command.
	bind media	Overrides any existing bind media command.
With active calls	bind all or bind control	The command is blocked, and the following messages are displayed: <i>00:16:39: There are active calls</i> <i>00:16:39: configure_sip_bind_command: The bind command change will not take effect</i>
	bind media	Succeeds and overrides any existing bind media command.

The **bind all** and **bind control** commands perform different functions based on the state of the interface. Table 41 describes the actions performed based on the interface state.



Note

Table 41 applies to **bind media** only if the media interface is the same as the **bind control** interface. If the two interfaces are different, media behavior is independent of the interface state.

Table 42 *bind all and bind control Functions, Based on Interface State*

Interface State	Result Using bind all or bind control Commands
Shutdown With or without active calls	<p>TCP and User Datagram Protocol (UDP) socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.)</p> <p>Then the sockets are opened to listen to any IP address.</p> <p>If the outgoing gateway has the bind command enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.</p>
Not shutdown Without active calls	<p>TCP and UDP socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.)</p> <p>Then the sockets are opened and bound to the IP address set by the bind command.</p> <p>The sockets accept packets destined for the bound address only.</p>
Not shutdown With active calls	<p>TCP and UDP socket listeners are initially closed.</p> <p>Then the sockets are opened to listen to any IP address.</p>
Bound interface's IP address is removed.	<p>TCP and UDP socket listeners are initially closed.</p> <p>Then the sockets are opened to listen to any address because the IP address has been removed.</p> <p>A message is printed that states the IP address has been deleted from the bound SIP interface.</p> <p>If the outgoing gateway has the bind command enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.</p>
The physical cable is pulled on the bound port, or the interface layer goes down.	<p>TCP and UDP socket listeners are initially closed.</p> <p>Then the sockets are opened and bound to listen to any address.</p> <p>When the pulled cable is replaced, the result is as documented for interfaces that are not shutdown.</p>
A bind interface is shut down, or its IP address is changed, or the physical cable is pulled while SIP calls are active.	<p>The call becomes a one-way call with media flowing in only one direction. The media flows from the gateway where the change or shutdown took place to the gateway where no change occurred. Thus, the gateway with the status change no longer receives media.</p> <p>The call is then disconnected, but the disconnected message is not understood by the gateway with the status change, and the call is still assumed to be active.</p>

How to Configure SIP Bind Features

This section contains the following procedures:

- [Setting the IP Address of an Interface to Be Bound, page 6](#)
- [Configuring the bind Command, page 7](#)
- [Monitoring the bind Command, page 9](#) (optional)
- [Troubleshooting Tips, page 10](#)



Note

- Before you perform a procedure, familiarize yourself with the following information:
 - [“Restrictions for SIP Bind Features” section on page 2](#)
- For help with a procedure, see the monitoring and troubleshooting sections listed above.

Setting the IP Address of an Interface to Be Bound

To set the IP address of an interface to be bound, perform the following steps.



Note

- You must perform this procedure before you can use the **bind** command.
- The **bind media** command applies to specific interfaces.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface**
4. **ip address**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<p>interface <i>type/number</i></p> <p>Example: Router(config)# interface fastethernet0</p>	<p>Configures an interface type. The argument is as follows:</p> <ul style="list-style-type: none"> <i>type/number</i>—Type of interface to be configured and the port, connector, or interface card number. <p>To find the specific definition of this command for your router, see the Cisco IOS Voice, Video, and Fax Command Reference, Release 12.3T.</p>
Step 4	<p>ip address <i>ip-address mask [secondary]</i></p> <p>Example: Router(config-if)# ip address 192.168.200.33 255.255.255.0</p>	<p>Configures a primary or secondary IP address for an interface. Keyword and argument are as follows:</p> <ul style="list-style-type: none"> <i>ip-address mask</i>—IP address and mask for the associated IP subnet. secondary—Makes the configured address a secondary IP address. If this keyword is omitted, the configured address is the primary IP address. <p>To find the correct definition of this command for your router, see the Cisco IOS Voice, Video, and Fax Command Reference, Release 12.3T.</p>
Step 5	<p>exit</p> <p>Example: Router(config-if)# exit</p>	<p>Exits the current mode.</p>

Configuring the bind Command

To configure the bind command, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **bind**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service VoIP configuration mode.
Step 4	sip Example: Router(conf-voi-serv)# sip	Enters SIP configuration mode.

	Command or Action	Purpose
Step 5	<p>bind {control media all} source interface <i>interface-id</i></p> <p>Example: Router(conf-serv-sip)# bind {control} source-interface FastEthernet0</p>	<p>Sets a source interface for signaling and media packets. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • control—Binds signaling packets. • media—Binds media packets. • all—Binds signaling and media packets. • source interface <i>interface-id</i>—Type of interface and its ID: <ul style="list-style-type: none"> - Async—Async interface - BVI—Bridge-group virtual interface - CTunnel—CTunnel interface - Dialer—Dialer interface - Ethernet—IEEE 802.3 - FastEthernet—Fast Ethernet IEEE 802.3 - Lex—Lex interface - Loopback—Loopback interface - Multilink—Multilink-group interface - Null—Null interface - Serial—Serial interface (Frame Relay) - Tunnel—Tunnel interface - Vif—PGM multicast host interface - Virtual-Template—Virtual template interface - Virtual-TokenRing—Virtual token ring
Step 6	<p>exit</p> <p>Example: Router(conf-serv-sip)# exit</p>	Exits the current mode.

Monitoring the bind Command

To monitor the **bind** command, perform the following steps.

SUMMARY STEPS

1. **show ip sockets**
2. **show sip-ua status**

DETAILED STEPS

Step 1 **show ip sockets**

Use this command to display IP socket information and indicate whether the bind address of the receiving gateway is set.

The following sample output indicates that the bind address of the receiving gateway is set.

```
Router# show ip sockets
```

Proto	Remote	Port	Local	Port	In	Out	Stat	TTY	OutputIF
17	0.0.0.0	0	--any--	2517	0	0	9	0	
17	--listen--		172.18.192.204	1698	0	0	1	0	
17	0.0.0.0	0	172.18.192.204	67	0	0	489	0	
17	0.0.0.0	0	172.18.192.204	5060	0	0	A1	0	

Step 2 show sip-ua status

Use this command to display SIP user-agent status and indicate whether bind is enabled.

The following sample output indicates that signaling is disabled and media on 172.18.192.204 is enabled.

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

```
SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): Disabled
SIP User Agent bind status(media): ENABLED 172.18.192.204
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
Redirection (3xx) message handling: ENABLED
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Session name line (s=) required
  Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udptl
```

Troubleshooting Tips



Note

For general troubleshooting tips and a list of important **debug** commands, see the “[General Troubleshooting Tips](#)” section on page 18.

Configuration Example for SIP Bind Features



Note

IP addresses and hostnames in examples are fictitious.

This sample output shows that bind is enabled on router 172.18.192.204:

```
Router# show running-config
```

```
Building configuration...

Current configuration : 2791 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
ip subnet-zero
ip ftp source-interface Ethernet0
!
voice service voip
    sip
        bind control source-interface FastEthernet0
!
interface FastEthernet0
    ip address 172.18.192.204 255.255.255.0
    duplex auto
    speed auto
    fair-queue 64 256 1000
    ip rsvp bandwidth 75000 100
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 5550111
port 1/1/1
!
dial-peer voice 29 voip
application session
destination-pattern 5550133
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

Additional References

General SIP References

- ["SIP Feature Roadmap" on Page 1](#)—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- ["Overview of SIP" on page 1](#)—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter

- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.3T at <http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/index.htm>

Any Internet Protocol (IP) addresses used in this document are not intended to be actual addresses. Any examples, command display output, and figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses in illustrative content is unintentional and coincidental.

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Configuring SIP DTMF Features

This chapter describes the following SIP features that support dual-tone multifrequency (DTMF) signaling:

- RFC 2833 Dual-Tone Multifrequency (DTMF) Media Termination Point (MTP) Passthrough
- DTMF Events Through SIP Signaling
- DTMF Relay for SIP Calls Using Named Telephone Events
- SIP INFO Method for DTMF Tone Generation
- SIP NOTIFY-Based Out-of-Band DTMF Relay Support
- SIP KPML-Based Out-of-Band DTMF Relay Support
- SIP Support for Asymmetric SDP

Feature History for the RFC 2833 DTMF MTP Passthrough

Release	Modification
12.4(11)T	This feature was introduced.

Feature History for DTMF Events Through SIP Signaling

Release	Modification
12.2(11)T	This feature was introduced.

Feature History for DTMF Relay for SIP Calls Using NTE

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB1	This feature was implemented on an additional platform.
12.2(8)T	This feature was integrated into this release.
12.2(11)T	This feature was implemented on additional platforms.



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Feature History for SIP INFO Method for DTMF Tone Generation

Release	Modification
12.2(11)T	This feature was introduced.

Feature History for SIP NOTIFY-Based Out-of-Band DTMF Relay Support

Release	Modification
12.3(4)T	This feature was introduced.

Feature History for SIP KPML-Based Out-of-Band DTMF Relay Support

Release	Modification
12.4(9)T	This feature was introduced.

Feature History for the SIP Support for Asymmetric SDP

Release	Modification
12.4(15)T	This feature was introduced.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Restrictions for SIP DTMF, page 2](#)
- [Prerequisites for SIP DTMF, page 4](#)
- [Information About SIP DTMF, page 4](#)
- [How to Configure SIP DTMF Features, page 18](#)
- [Configuration Examples for SIP DTMF Features, page 32](#)
- [Additional References, page 38](#)

Restrictions for SIP DTMF

RFC 2833 DTMF MTP Passthrough Feature

- The RFC 2833 DTMF MTP Passthrough feature adds support for passing DTMF tones transparently between Session Initiation Protocol (SIP) endpoints that require either transcoding or use of the RSVP Agent feature. If the T38 Fax Relay feature is also configured on this IP network, configure the voice gateways to use a payload type other than PT97 or PT98 for fax relay negotiation, or depending on whether the SIP endpoints support different payload types, configure Cisco Unified CME to use a payload type other than PT97 or PT98 for DTMF.

DTMF Events Through SIP Signaling Feature

- The DTMF Events Through SIP Signaling feature adds support for sending telephone-event notifications via SIP NOTIFY messages from a SIP gateway. The events for which notifications are sent out are DTMF events from the local Plain Old Telephone Service (POTS) interface on the gateway. Notifications are not sent for DTMF events received in the Real-Time Transport Protocol (RTP) stream from the recipient user agent.

DTMF Relay for SIP Calls Using NTEs Feature

- The SIP NTE DTMF relay feature is available only for SIP calls on Cisco VoIP gateways. The SIP NTE DTMF relay feature supports only hookflash relay and does not support hookflash generation for advanced features such as call waiting and conferencing.

SIP INFO Method for DTMF Tone Generation Feature

- Minimum signal duration is 100 ms. If a request is received with a duration less than 100 ms, the minimum duration of 100 ms is used by default.
- Maximum signal duration is 5000 ms. If a request is received with a duration longer than 5000 ms, the maximum duration of 5000 ms is used by default.
- If no duration parameter is included in a request, the gateway defaults to a signal duration of 250 ms.

SIP NOTIFY-Based Out-of-Band DTMF Relay Support Feature

- To support Skinny Client Control Protocol (SCCP) IP phones, originating and terminating SIP gateways can use NOTIFY-based out-of-band DTMF relay. NOTIFY-based out-of-band DTMF relay is a Cisco proprietary function.
- You can configure support only on a SIP VoIP dial peer.

SIP KPML-Based Out-of-Band DTMF Relay Support Feature

- For incoming dial peers, if you configure multiple DTMF negotiation methods, the first value you configure takes precedence, then the second, and then the third.
- For incoming dial peers, the first out-of-band negotiation method takes precedence over other DTMF negotiation methods, except when the **dtmf-relay rtp-nte** command has precedence; in this case, the **dtmf-relay sip-kpml** command takes precedence over other out-of-band negotiation methods.
- For incoming dial peers, if both the **dtmf-relay rtp-nte** and **dtmf-relay sip-kpml** commands and notification mechanisms are enabled and negotiated, the gateway relies on RFC 2833 notification to receive digits and a SUBSCRIBE for KPML is not initiated.
- SIP KPML support complies to the IETF draft “draft-ietf-sipping-kpml-04.txt” with the following limitations:
 - The SIP gateway always initiates SUBSCRIBE in the context of an established INVITE dialog. The gateway supports receiving SUBSCRIBE in the context of an established INVITE dialog, as well as out-of-call context requests with a leg parameter in the Event header. If the request code does not match an existing INVITE dialog, the gateway sends a NOTIFY with KPML status-code 481 and sets Subscription-State to terminated.
 - The gateway does not support the Globally Routable User Agent (GRUU) requirement. The Contact header in the INVITE/200 OK message is generated locally from the gateway’s contact informationn.
 - The gateway always initiates persistent subscriptions, but the gateway receives and processes persistent and one-shot subscriptions.

- The gateway supports only single-digit reporting. There is no need for inter-digit timer support. The only regular expressions supported are those which match a single digit. For example:
 - `<regex>x</regex>`—Matches any digit 0 through 9
 - `<regex>1</regex>`—Matches digit 1
 - `<regex>[x#*ABCD]</regex>`—Matches to any digit 0 through 9, # (the pound sign), * (an asterisk), or A, B, C, or D
 - `<regex>[24]</regex>`—Matches digits 2 or 4
 - `<regex>[2-9]</regex>`—Matches on any digit 2 through 9
 - `<regex>[^2-9]</regex>`—Matches digits 0 or 1
- The gateway does not support long key presses, which are detected and reported as a single digit press.
- Digit suppression is not supported (*pre* tag for suppressing inband digits).
- Individual stream selection is not supported. A SUBSCRIBE request for KPML applies to all audio streams in the dialog (*stream* element and *reverse* are not supported).
- You can configure support only on a SIP VoIP dial peer.

Prerequisites for SIP DTMF

DTMF Relay for SIP Calls Using NTEs Feature

- Ensure that you have a working VoIP network using SIP on Cisco gateways.

Information About SIP DTMF

To configure SIP DTMF support, you should understand the following concepts:

- [RFC 2833 DTMF MTP Passthrough, page 4](#)
- [DTMF Events Through SIP Signaling, page 5](#)
- [DTMF Relay for SIP Calls Using NTEs, page 6](#)
- [SIP INFO Method for DTMF Tone Generation, page 7](#)
- [SIP NOTIFY-Based Out-of-Band DTMF Relay, page 8](#)
- [SIP KPML-Based Out-of-Band DTMF Relay, page 10](#)
- [SIP Support for Asymmetric SDP, page 14](#)

RFC 2833 DTMF MTP Passthrough

The RFC 2833 Dual-Tone Multifrequency (DTMF) Media Termination Point (MTP) Passthrough feature passes DTMF tones transparently between Session Initiation Protocol (SIP) endpoints that require either transcoding or use of the RSVP Agent feature. (An RSVP agent is a Cisco IOS-based Resource Reservation Protocol [RSVP] proxy server that registers with the call manager—Cisco Unified CallManager or Cisco Unified CallManager Express—as a media-termination point or a transcoder device.)

The MTP or transcoding module on a gateway detects RFC 2833 (DTMF) packets from an IP endpoint. You can configure whether it should do either or both of the following:

- Generate and send an out-of-band signal event to the call manager
- Pass the packets through to the other IP endpoint (default)

You can configure this instruction from the call manager, from the gateway, or both. The gateway can itself contain a call manager with an MTP or transcoder, or it can connect to another gateway that contains a call manager with an MTP or transcoder. Configuration on the call manager takes precedence over configuration on the gateway.

DTMF Events Through SIP Signaling

The DTMF Events Through SIP Signaling feature provides the following benefits:

- Provides DTMF event notification for SIP messages.
- Provides the capability of receiving hookflash event notification through the SIP NOTIFY method.
- Enables third-party call control, or other signaling mechanisms, to provide enhanced services, such as calling card and messaging services.
- Allows the user to communicate with the application outside of the media connection.

The DTMF Events Through SIP Signaling feature allows telephone event notifications to be sent through SIP NOTIFY messages, using the SIP SUBSCRIBE/NOTIFY method as defined in the Internet Engineering Task Force (IETF) draft, *SIP-Specific Event Notification*. The feature also supports sending DTMF notifications based on the IETF draft, *draft-mahy-sip-signaled-digits-01.txt, Signaled Telephony Events in the Session Initiation Protocol (SIP)*.

DTMF Dialing

DTMF dialing consists of simultaneous voice-band tones generated when a button is pressed on a telephone. The use of DTMF signaling for this feature enables support for advanced telephony services. Currently there are a number of application servers and service creation platforms that do not support media connections. To provide value-added services to the network, these servers and platforms need to be aware of signaling events from a specific participant in the call. Once the server or platform is aware of the DTMF events that are being signaled, it can use third-party call control, or other signaling mechanisms, to provide enhanced services. Examples of the types of services and platforms that are supported by this feature are various voice web browser services, Centrex switches or business service platforms, calling card services, and unified message servers. All of these applications require a method for the user to communicate with the application outside of the media connection. The DTMF Events Through SIP Signaling feature provides this signaling capability.

This feature is related to the SIP INFO Method for DTMF Tone Generation feature, which adds support for out-of-band DTMF tone generation using the SIP INFO method. Together the two features provide a mechanism to both send and receive DTMF digits along the signaling path.

NOTIFY Messages

The SIP event notification mechanism uses NOTIFY messages to signal when certain telephony events take place. In order to send DTMF signals through NOTIFY messages, the gateway notifies the subscriber when DTMF digits are signaled by the originator. The notification contains a message body with a SIP response status line.

The following sample message shows a NOTIFY message from the Notifier letting the Subscriber know that the subscription is completed. The combination of the `From`, `To`, and `Call-ID` headers identifies the call leg. The `Events` header specifies the event type being signaled, and the `Content-Type` specifies the Internet media type. The `Content-Length` header indicates the number of octets in the message body.

```
NOTIFY sip:subscriber@example1.com SIP/2.0
Via: SIP/2.0/UDP example2.com:5060
From: Notifier <sip:notifier@example2.com>;tag=5678-EFGH
To: Subscriber <sip:subscriber@example1.com>;tag=1234-ABCD
Call-ID: 12345@example2.com
CSeq: 104 NOTIFY
Contact: Notifier <sip:notifier@example2.com>
Events: telephone-event;rate=1000
Content-Type: audio/telephone-event
Content-Length: 4
```

DTMF Relay for SIP Calls Using NTEs

Feature benefits include the following:

- Reliable DTMF digit relay between Cisco VoIP gateways when low-bandwidth codecs are used
- Ability to communicate with SIP phone software that uses NTE packets to indicate DTMF digits

The SIP NTE DTMF relay feature is used for the following applications:

- [Reliable DTMF Relay, page 6](#)
- [SIP IP Phone Support, page 7](#)



Note

This feature is implemented only for SIP calls on VoIP gateways.

Reliable DTMF Relay

The SIP NTE DTMF relay feature provides reliable digit relay between Cisco VoIP gateways when a low-bandwidth codec is used. Using NTE to relay DTMF tones provides a standardized means of transporting DTMF tones in Real-Time Transport Protocol (RTP) packets according to section 3 of RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, developed by the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group. RFC 2833 defines formats of NTE RTP packets used to transport DTMF digits, hookflash, and other telephony events between two peer endpoints.

DTMF tones are generated when a button on a touch-tone phone is pressed. When the tone is generated, it is compressed, transported to the other party, and decompressed. If a low-bandwidth codec, such as a G.729 or G.723 is used without a DTMF relay method, the tone may be distorted during compression and decompression.

With the SIP NTE DTMF relay feature, the endpoints perform per-call negotiation of the DTMF relay method. They also negotiate to determine the payload type value for the NTE RTP packets.

In a SIP call, the gateway forms a Session Description Protocol (SDP) message that indicates the following:

- If NTE will be used
- Which events will be sent using NTE
- NTE payload type value

The SIP NTE DTMF relay feature can relay hookflash events in the RTP stream using NTP packets.

**Note**

The SIP NTE DTMF relay feature does not support hookflash generation for advanced features such as call waiting and conferencing.

SIP IP Phone Support

The SIP NTE DTMF relay feature adds SIP phone support. When SIP IP phones are running software that does not have the capability to generate DTMF tones, the phones use NTE packets to indicate DTMF digits. With the SIP NTE DTMF relay feature, Cisco VoIP gateways can communicate with SIP phones that use NTE packets to indicate DTMF digits. The Cisco VoIP gateways can relay the digits to other endpoints.

SIP INFO Method for DTMF Tone Generation

This section describes the SIP INFO Method for DTMF Tone Generation feature, which uses the SIP INFO method to generate dual-tone multifrequency (DTMF) tones on the telephony call leg. SIP methods or request message types, request a specific action be taken by another user agent or proxy server. The SIP INFO message is sent along the signaling path of the call. With the feature, upon receipt of a SIP INFO message with DTMF relay content, the gateway generates the specified DTMF tone on the telephony end of the call.

The SIP INFO Method for DTMF Tone Generation feature is always enabled, and is invoked when a SIP INFO message is received with DTMF relay content. This feature is related to the SIP NOTIFY-Basec Out-of-Band DTMF Relay Support feature, which provides the ability for an application to be notified about DTMF events using SIP NOTIFY messages. Together, the two features provide a mechanism to both send and receive DTMF digits along the signaling path.

**Note**

For information on sending DTMF event notification using SIP NOTIFY messages, see the [“DTMF Events Through SIP Signaling”](#) section on page 5.

SIP INFO Messages

The SIP INFO method is used by a user agent to send call signaling information to another user agent with which it has an established media session. The following example shows a SIP INFO message with DTMF content:

```
INFO sip:2143302100@172.17.2.33 SIP/2.0
Via: SIP/2.0/UDP 172.80.2.100:5060
From: <sip:9724401003@172.80.2.100>;tag=43
To: <sip:2143302100@172.17.2.33>;tag=9753.0207
Call-ID: 984072_15401962@172.80.2.100
CSeq: 25634 INFO
Supported: 100rel
Supported: timer
Content-Length: 26
Content-Type: application/dtmf-relay

Signal= 1
Duration= 160
```

This sample message shows a SIP INFO message received by the gateway with specifics about the DTMF tone to be generated. The combination of the `From`, `To`, and `Call-ID` headers identifies the call leg. The `signal` and `duration` headers specify the digit, in this case 1, and duration, 160 milliseconds in the example, for DTMF tone play.

SIP NOTIFY-Based Out-of-Band DTMF Relay

SCCP IP phones do not support in-band DTMF digits; they are capable of sending only out-of-band DTMF digits. To support SCCP devices, originating and terminating SIP gateways can use Cisco proprietary NOTIFY-based out-of-band DTMF relay. In addition, NOTIFY-based out-of-band DTMF relay can also be used by analog phones attached to analog voice ports (FXS) on the router.

NOTIFY-based out-of-band DTMF relay sends messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, NOTIFY-based out-of-band DTMF relay takes precedence.

The originating gateway sends an Invite message with a SIP Call-Info header to indicate the use of NOTIFY-based out-of-band DTMF relay. The terminating gateway acknowledges the message with an 18x or 200 Response message, also using the Call-Info header. The Call-Info header for NOTIFY-based out-of-band relay appears as follows:

```
Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
```



Note

Duration is the interval between NOTIFY messages sent for a single digit and is set by means of the `notify telephone-event` command.

First, the NOTIFY-based out-of-band DTMF relay mechanism is negotiated by the SIP Invite and 18x/200 Response messages. Then, when a DTMF event occurs, the gateway sends a SIP NOTIFY message for that event. In response, the gateway expects to receive a 200 OK message.

The NOTIFY-based out-of-band DTMF relay mechanism is similar to the DTMF message format described in RFC 2833. NOTIFY-based out-of-band DTMF relay consists of 4 bytes in a binary encoded format. The message format is shown in Figure 74; field descriptions are listed in Table 43.

Figure 74 Message Format of NOTIFY-Based Out-of-Band DTMF Relay

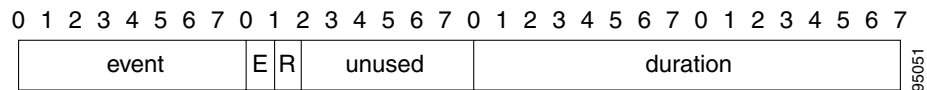


Table 43 Fields in NOTIFY-based out-of-band DTMF relay Message

Field	Description
event	The DTMF event that is between 0-9, A, B, C, D, #, * and flash.
E	E signifies the end bit. If E is set to a value of 1, the NOTIFY message contains the end of the DTMF event. Thus, the duration parameter in this final NOTIFY message measures the complete duration of the event.
R	Reserved.

Table 43 *Fields in NOTIFY-based out-of-band DTMF relay Message*

Field	Description
unused	In RFC 2833, unused corresponds to the volume field, but is not used in NOTIFY-based out-of-band DTMF relay.
duration	Duration of this DTMF event, in milliseconds.

Sending NOTIFY Messages

As soon as the DTMF event is recognized, the gateway sends out an initial NOTIFY message for this event with the duration negotiated in the Invite's Call-Info header. For the initial NOTIFY message, the end bit is set to zero. Afterward, one of the following actions can occur:

- If the duration of the DTMF event is less than the negotiated duration, the originating gateway sends an end NOTIFY message for this event with the duration field containing the exact duration of the event and the end bit set to 1.
- If the duration of the DTMF event is greater than the negotiated duration, the originating gateway sends another NOTIFY message for this event after the initial timer runs out. The updated NOTIFY message has a duration of twice the negotiated duration. The end bit is set to 0 because the event is not yet over. If the event lasts beyond the duration specified in the first updated NOTIFY message, another updated NOTIFY message is sent with three times the negotiated duration.
- If the duration of the DTMF event is exactly the negotiated duration, either of the above two actions occurs, depending on whether the end of the DTMF event occurred before or after the timer ran out.

For example, if the negotiated duration is 600 ms, as soon as a DTMF event occurs, the initial NOTIFY message is sent with duration as 600 ms. Then a timer starts for this duration.

- If the DTMF event lasts only 300 ms, the timer stops and an end NOTIFY message is sent with the duration as 300 ms.
- If the DTMF event lasts longer than 600 ms (1000 ms), when the timer expires an updated NOTIFY message is sent with the duration as 1200 ms and the timer restarts. When the DTMF event ends, an end NOTIFY message is sent with the duration set to 1000 ms.

Every DTMF event corresponds to at least two NOTIFYs: an initial NOTIFY message and an end NOTIFY message. There might also be some update NOTIFYs involved, if the total duration of the event is greater than the negotiated max-duration interval. Because DTMF events generally last for less than 1000 ms, setting the duration using **notify telephone-event** command to more than 1000 ms reduces the total number of NOTIFY messages sent. The default value of **notify telephone-event** command is 2000 ms.

Receiving NOTIFY Messages

Once a NOTIFY message is received by the terminating gateway, the DTMF tone plays and a timer is set for the value in the duration field. Afterward, one of the following actions can occur:

- If an end NOTIFY message for a DTMF event is received, the tone stops.
- If an update is received, the timer is updated according to the duration field.
- If an update or end NOTIFY message is not received before the timer expires, the tone stops and all subsequent NOTIFY messages for the same DTMF event or DTMF digit are ignored until an end NOTIFY message is received.

- If a NOTIFY message for a different DTMF event is received before an end NOTIFY message for the current DTMF event is received (which is an unlikely case), the current tone stops and the new tone plays. This is an unlikely case because for every DTMF event there needs to be an end NOTIFY message, and unless this is successfully sent and a 200 OK is received, the gateway cannot send other NOTIFY messages.

**Note**

In-band tones are not passed while NOTIFY-based out-of-band DTMF relay is used as the DTMF relay method.

Two commands allow you to enable or disable NOTIFY-based out-of-band DTMF relay on a dial peer. The functionality is advertised to the other end using Invite messages if it is enabled by the commands, and must be configured on both the originating and terminating SIP gateways. A third command allows you to verify DTMF relay status.

- **dtmf-relay (VoIP)**
- **notify telephone-event**
- **show sip-ua status**

SIP KPML-Based Out-of-Band DTMF Relay

KPML support is required on SIP gateways for non-conferencing calls, and for interoperability between SIP products and SIP phones. If you configure KPML on the dial peer, the gateway sends INVITE messages with “kpml” in the Allow-Events header. Currently, all configured DTMF methods are recognized and sent in the outgoing INVITE. If you configure rtp-nte (RFC 2833), sip-notify, and sip-kpml, the outgoing INVITE contains a call-info header, an Allow-Events header with KPML, and an sdp with rtp-nte payload.

DTMF negotiation is performed based on the matching inbound dial-peer configuration. The gateway negotiates to either just cisco-rtp, just rtp-nte, rtp-nte + kpml, just kpml, or just sip-notify. If you configure more than one out-of-band DTMF method, preference goes from highest to lowest in the order they were configured. Whichever DTMF negotiation method you configure first takes precedence.

A gateway negotiates both rtp-nte and KPML if both are supported and advertised in the incoming INVITE. However, in this case, the gateway relies on the rtp-nte DTMF method to receive digits and a SUBSCRIBE for KPML is not initiated, however the gateway still accepts SUBSCRIBES for KPML. This prevents double-digit reporting problems at the gateway.

The following example shows the INVITE and SUBSCRIBE sequence for KPML.

```
Sent:
INVITE sip:8888@172.18.193.250:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bK1ECC
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>
Date: Fri, 01 Mar 2002 00:15:59 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
Cisco-Guid: 1424162198-736104918-2148455531-3036263926
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1014941759
Contact: <sip:172.18.193.251:5060>
```



```

Expires: 180
Allow-Events: kpml, telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 221

v=0
o=CiscoSystemsSIP-GW-UserAgent 1438 8538 IN IP4 172.18.193.251
s=SIP Call
c=IN IP4 172.18.193.251
t=0 0
m=audio 17576 RTP/AVP 0 19
c=IN IP4 172.18.193.251
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20

//-1/xxxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bKC1ECC
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Date: Fri, 01 Mar 2002 01:02:34 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
Timestamp: 1014941759
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
Require: 100rel
RSeq: 3482
Allow-Events: kpml, telephone-event
Contact: <sip:8888@172.18.193.250:5060>
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 221

v=0
o=CiscoSystemsSIP-GW-UserAgent 9384 6237 IN IP4 172.18.193.250
s=SIP Call
c=IN IP4 172.18.193.250
t=0 0
m=audio 17468 RTP/AVP 0 19
c=IN IP4 172.18.193.250
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20

//-1/xxxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bKC1ECC
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Date: Fri, 01 Mar 2002 01:02:38 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
Timestamp: 1014941759
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
Allow-Events: kpml, telephone-event
Contact: <sip:8888@172.18.193.250:5060>

```

```

Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 221

v=0
o=CiscoSystemsSIP-GW-UserAgent 9384 6237 IN IP4 172.18.193.250
s=SIP Call
c=IN IP4 172.18.193.250
t=0 0
m=audio 17468 RTP/AVP 0 19
c=IN IP4 172.18.193.250
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000

//-1/xxxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:8888@172.18.193.250:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bKKB8B
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Date: Fri, 01 Mar 2002 00:16:00 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: kpml, telephone-event
Content-Length: 0

//-1/xxxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SUBSCRIBE sip:8888@172.18.193.250:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bKFF36
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 103 SUBSCRIBE
Max-Forwards: 70
Date: Fri, 01 Mar 2002 00:16:15 GMT
User-Agent: Cisco-SIPGateway/IOS-12.x
Event: kpml
Expires: 7200
Contact: <sip:172.18.193.251:5060>
Content-Type: application/kpml-request+xml
Content-Length: 327

<?xml version="1.0" encoding="UTF-8"?><kpml-request
xmlns="urn:ietf:params:xml:ns:kpml-request"
xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
xsi:schemaLocation="urn:ietf:params:xml:ns:kpml-request kpml-request.xsd"
version="1.0"><pattern persist="persist"><regex
tag="dtmf">[x*#ABCD]</regex></pattern></kpml-request>

//-1/xxxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SUBSCRIBE sip:172.18.193.251:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.250:5060;branch=z9hG4bK5FE3
From: <sip:8888@172.18.193.250>;tag=39497C-2EA
To: <sip:172.18.193.251>;tag=EA330-F6
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 101 SUBSCRIBE
Max-Forwards: 70
Date: Fri, 01 Mar 2002 01:02:46 GMT
User-Agent: Cisco-SIPGateway/IOS-12.x
Event: kpml
Expires: 7200

```

```
Contact: <sip:172.18.193.250:5060>
Content-Type: application/kpml-request+xml
Content-Length: 327
```

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bKFF36
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Date: Fri, 01 Mar 2002 01:02:51 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 103 SUBSCRIBE
Content-Length: 0
Contact: <sip:172.18.193.250:5060>
Expires: 7200
```

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.250:5060;branch=z9hG4bK5FE3
From: <sip:8888@172.18.193.250>;tag=39497C-2EA
To: <sip:172.18.193.251>;tag=EA330-F6
Date: Fri, 01 Mar 2002 00:16:24 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 101 SUBSCRIBE
Content-Length: 0
Contact: <sip:172.18.193.251:5060>
Expires: 7200
```

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
NOTIFY sip:172.18.193.250:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bK101EA4
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 104 NOTIFY
Max-Forwards: 70
Date: Fri, 01 Mar 2002 00:16:24 GMT
User-Agent: Cisco-SIPGateway/IOS-12.x
Event: kpml
Subscription-State: active
Contact: <sip:172.18.193.251:5060>
Content-Length: 0
```

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
NOTIFY sip:172.18.193.251:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.250:5060;branch=z9hG4bK6111
From: <sip:8888@172.18.193.250>;tag=39497C-2EA
To: <sip:172.18.193.251>;tag=EA330-F6
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 102 NOTIFY
Max-Forwards: 70
Date: Fri, 01 Mar 2002 01:02:51 GMT
User-Agent: Cisco-SIPGateway/IOS-12.x
Event: kpml
Subscription-State: active
Contact: <sip:172.18.193.250:5060>
Content-Length: 0
```

```
//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
```

```

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.250:5060;branch=z9hG4bK6111
From: <sip:8888@172.18.193.250>;tag=39497C-2EA
To: <sip:172.18.193.251>;tag=EA330-F6
Date: Fri, 01 Mar 2002 00:16:32 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 102 NOTIFY
Content-Length: 0

//-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
NOTIFY sip:172.18.193.250:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bK1117DE
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 105 NOTIFY
Max-Forwards: 70
Date: Fri, 01 Mar 2002 00:37:33 GMT
User-Agent: Cisco-SIPGateway/IOS-12.x
Event: kpml
Subscription-State: active
Contact: <sip:172.18.193.251:5060>
Content-Type: application/kpml-response+xml
Content-Length: 113

<?xml version="1.0" encoding="UTF-8"?><kpml-response version="1.0" code="200" text="OK"
digits="1" tag="dtmf"/>

/-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.251:5060;branch=z9hG4bK1117DE
From: <sip:172.18.193.251>;tag=EA330-F6
To: <sip:8888@172.18.193.250>;tag=39497C-2EA
Date: Fri, 01 Mar 2002 01:24:08 GMT
Call-ID: 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
CSeq: 105 NOTIFY
Content-Length: 0

...

```

SIP Support for Asymmetric SDP

The SIP Support for Asymmetric SDP feature allows a SIP gateway to receive an offer or answer with a different payload than it prefers, and treat the payloads as asymmetric, as long as both the payloads are not in use. The gateway interprets the payload that it receives and generates RTP packets for DTMF events and dynamic codecs. The gateway expects to receive the DTMF events and dynamic codec payload value it originally sent in the request/response message. In this way, the gateway advertises its preferred payload in the answer to the offer that it receives.

For delayed media cases, if the gateway receives an INVITE message with no SDP, the corresponding response or offer carries the preferred payload in the SDP. The gateway expects that the remote end will use the payload offered in RTP packets that it receives. The gateway uses the payload type received in the answer in all the RTP packets that it generates, as long as this payload is not in use.

The following sections describe the behavior of a Universal Agent Client (UAC) and Universal Agent Server (UAS) gateway for asymmetric payload handling:

- [UAC Behavior for Asymmetric DTMF Payloads, page 15](#)

- [UAS Behavior for Asymmetric DTMF Payloads, page 15](#)
- [UAC Behavior for Asymmetric Dynamic Codec Payloads, page 16](#)
- [UAS Behavior for Asymmetric Dynamic Codec Payloads, page 17](#)

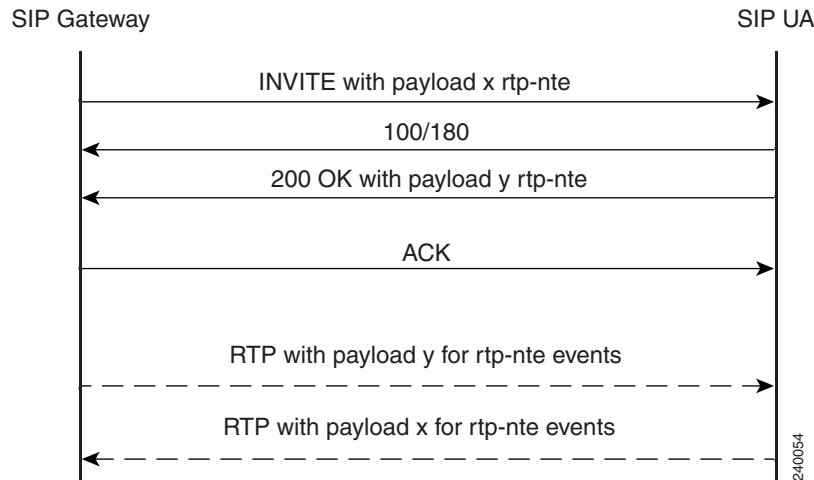
UAC Behavior for Asymmetric DTMF Payloads

When a UAC gateway originates an INVITE message, it advertises the preferred DTMF relay payload and method. However, when the gateway receives an 18x/2xx response with a different DTMF payload, the gateway first checks if the associated DTMF relay method can successfully negotiate with it. If negotiation is successful, the gateway checks if the new payload is available for use. If it is available, the gateway uses the received payload in all RTP (RFC 2833) events packets it generates. At the same time, the gateway expects the remote end to send packets with the preferred payload that the gateway advertised in its original request.

If the gateway is unable to use the new payload that the remote end requests in the answer, then DTMF relay negotiation fails for the requested DTMF relay method.

Figure 75 summarizes the call behavior for a Universal Access Controller (UAC) when using asymmetric DTMF payloads.

Figure 75 UAC Behavior for Asymmetric DTMF Payloads



UAS Behavior for Asymmetric DTMF Payloads

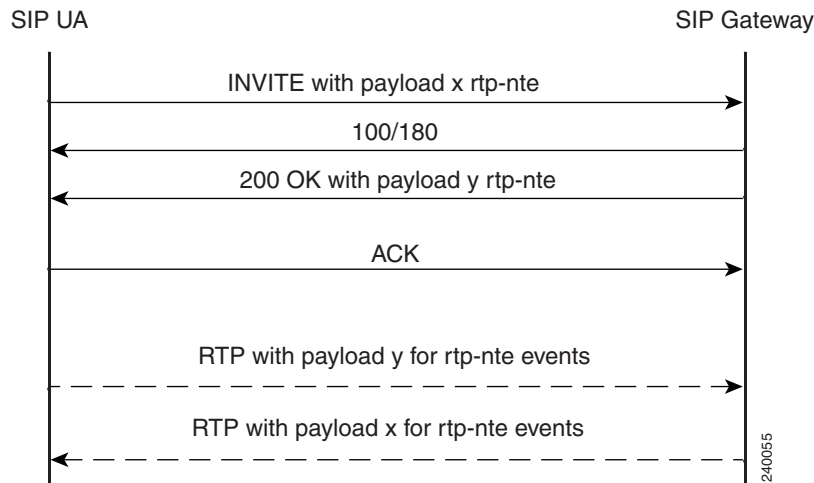
When a gateway receives an INVITE message that has a preferred DTMF relay method and payload, and the requested DTMF relay is negotiated successfully, the gateway checks if the requested DTMF payload is available for use. If it is available, the gateway checks if you configured a preferred payload at the matching dial peer.

If you did not configure a preferred payload, the gateway uses its default value for that particular DTMF relay method. If the preferred payload is different from the one in the original request, the payload in the original request is used in all RTP (RFC 2833) events packets that the gateway generates. The gateway responds with the preferred payload in the SDP in the response to the original request. The gateway expects the remote end to send packets with the preferred payload that the gateway advertised in its response.

When the gateway receives a delayed media request, the gateway offers the preferred DTMF relay and payload in its response. If the remote end responds with the same DTMF relay and different payload, this new payload is used to generate RTP RFC 2833 events packets. The gateway expects the remote end to send packets with the preferred payload that the gateway advertised in its response.

Figure 76 summarizes the call behavior for a Universal Access Server (UAS) when using asymmetric DTMF payloads.

Figure 76 UAS Behavior for Asymmetric DTMF Payloads



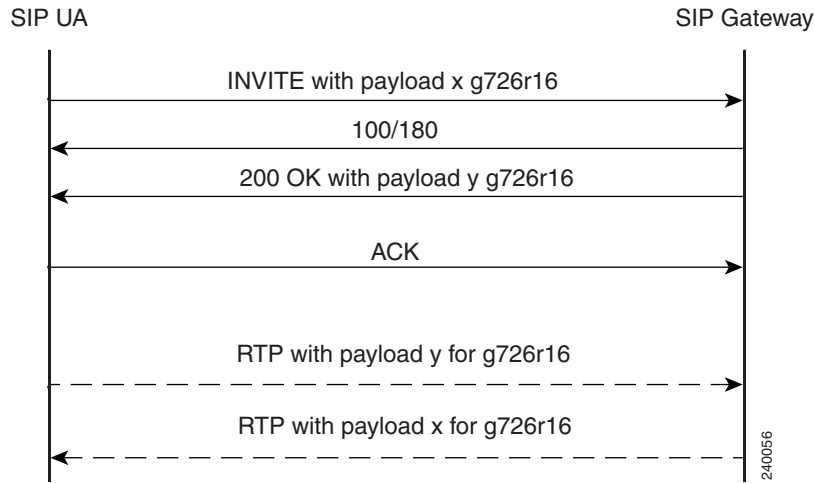
UAC Behavior for Asymmetric Dynamic Codec Payloads

When the gateway originates an INVITE message, the gateway advertises the preferred dynamic codec. However, when the gateway receive an 18x/2xx response with a different dynamic payload, the gateway first checks whether the new payload, in the answer SDP message, is being used for the same dynamic codec that was advertised. If the new payload is the same codec, and if the dynamic payload is available for use, the received payload is used in all RTP dynamic codec packets that are generated by the gateway. At the same time, the gateway expects the remote end to send packets with the preferred payload that the gateway advertised in its original request.

If the gateway is unable to use the new payload that the remote end requests in the answer, then codec negotiation for that particular dynamic codec fails.

Figure 77 summarizes the call behavior for a UAC when using asymmetric dynamic codec payloads.

Figure 77 UAC Behavior for Asymmetric Dynamic Codec Payloads



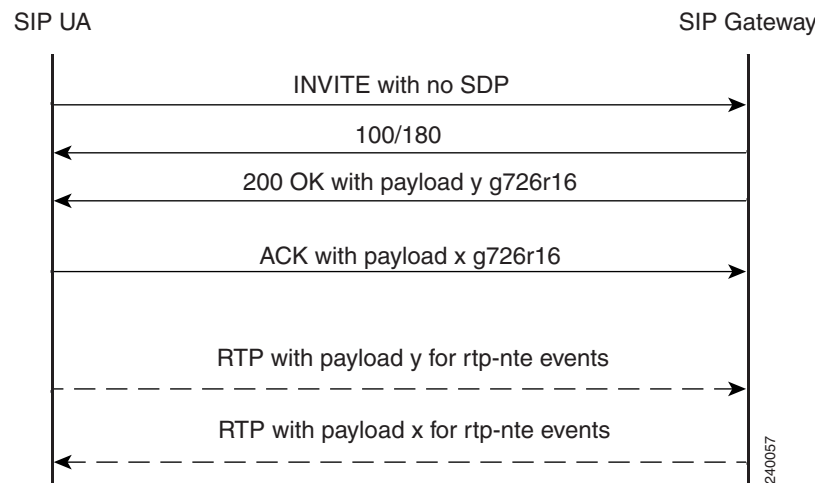
UAS Behavior for Asymmetric Dynamic Codec Payloads

When a gateway receives an INVITE message which has a dynamic codec and payload, and the requested dynamic codec and payload is negotiated successfully, the gateway uses the same payload in its response to the INVITE message. The gateway cannot generate a different dynamic payload in a response message.

Beginning with Cisco Software Release 12.4(15)T, when a gateway receives a delayed media request, you can configure the gateway to offer the preferred dynamic codec and payload in its response. If the remote end responds with the same dynamic codec and a different payload, this new payload is used to generate the codec RTP packets. The gateway expects the remote end to send packets with the preferred payload that the gateway advertised in its response.

Figure 78 summarizes the call behavior for UAS when using asymmetric dynamic codec payloads.

Figure 78 UAS Behavior for Asymmetric Dynamic Codec Payloads



How to Configure SIP DTMF Features

This section contains the following procedures:

- [Configuring Passthrough on a Gateway That Connects to an MTP or Transcoder Gateway](#), page 18
- [Configuring DTMF Events Through SIP Signaling](#), page 19
- [Configuring DTMF Relay for SIP Calls Using NTEs](#), page 20
- [Configuring SIP INFO Method for DTMF Tone Generation](#), page 22
- [Configuring SIP NOTIFY-Based Out-of-Band DTMF Relay](#), page 22
- [Configuring SIP KPML-Based Out-of-Band DTMF Relay](#), page 24
- [Configuring SIP Support for SDP](#), page 29
- [Troubleshooting Tips](#), page 32

**Note**

-
- Before you perform a procedure, familiarize yourself with the following information:
 - [“Restrictions for SIP DTMF”](#) section on page 2
 - [“Prerequisites for SIP DTMF”](#) section on page 4
 - For help with a procedure, see the verification and troubleshooting sections listed above.
-

Configuring Passthrough on a Gateway That Connects to an MTP or Transcoder Gateway

To configure RFC 2833 DTMF MTP Passthrough on a gateway that does not itself contain an MTP or transcoder but connects to another gateway that does, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *tag* voip**
4. **dtmf-relay rtp-nte**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 103 voip	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	dtmf-relay rtp-nte Example: Router(config-dial-peer-voice)# dtmf-relay rtp-nte	Specifies that the gateway should relay DTMF tones between telephony interfaces and an IP network by using RTP with the Named Telephone Event (NTE) payload type.
Step 5	exit Example: Router(config-dial-peer-voice)# exit	Exits the current configuration mode.

**Note**

If you plan to employ both this feature and Cisco Fax Relay—both of which use the same default payload type (96 or 97)—in the same RTP session, you may need to configure the gateway that negotiates fax relay to use a different payload type.

To configure a gateway to use a different payload type, use the **rtp payload-type** command as in the following example:

```
Router(config-dial-peer)# rtp payload-type nte 105
```

Configuring DTMF Events Through SIP Signaling

To configure the DTMF Events Through SIP Signaling feature, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers notify**

5. **retry notify**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers notify number Example: Router(config-sip-ua)# timers notify 100	Sets the amount of time that the user agent waits before retransmitting the Notify message. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Time, in ms, to wait before retransmitting. Range: 100 to 1000. Default: 500.
Step 5	retry notify number Example: Router(config-sip-ua)# retry notify 6	Sets the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Number of retries. Range: 1 to 10. Default: 10.
Step 6	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring DTMF Relay for SIP Calls Using NTEs

This section includes the following procedures:

- [Configure DTMF Relay for SIP Calls Using NTEs, page 21](#)
- [Configuring Passthrough on a Gateway That Connects to an MTP or Transcoder Gateway, page 18](#)

Configure DTMF Relay for SIP Calls Using NTEs

To configure the DTMF Relay for SIP Calls Using NTEs feature, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice**
4. **session protocol sipv2**
5. **dtmf-relay rtp-nte**
6. **rtp payload-type nte comfort-noise**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> voip Example: Router(config)# dial-peer voice 10 voip	Enters dial-peer VoIP configuration mode for the specified dial peer.
Step 4	session protocol sipv2 Example: Router(config-dial-peer)# session protocol sipv2	Specifies a session protocol for calls between local and remote routers using the packet network. The keyword is as follows: <ul style="list-style-type: none"> • sipv2—Dial peer uses the IETF SIP. Use this keyword with the SIP option.
Step 5	dtmf-relay rtp-nte Example: Router(config-dial-peer)# dtmf-relay rtp-nte	Specifies how an H.323 or SIP gateway relays DTMF tones between telephone interfaces and an IP network. The keyword is as follows: <ul style="list-style-type: none"> • rtp-nte—Forwards tones by using RTP with the NTE payload type.

	Command or Action	Purpose
Step 6	<pre>rtp payload-type nte <i>number</i> comfort-noise [13 19]</pre> <p>Example: Router(config-dial-peer)# rtp payload-type nte 100 comfort-noise 13</p>	<p>Identifies the payload type of a RTP packet. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> nte <i>number</i>—Named telephone event (NTE). Range: 96 to 127. Default: 101. comfort-noise—RTP payload type of comfort noise. If you are connected to a gateway that complies with the <i>RTP Payload for Comfort Noise July 2001</i> draft, use 13. If you are connected to an older Cisco gateway that uses DSPware before version 3.4.32, use 19.
Step 7	<pre>exit</pre> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configuring SIP INFO Method for DTMF Tone Generation

You cannot configure, enable, or disable this feature. You can display SIP statistics, including SIP INFO method statistics, by using the **show sip-ua statistics** and **show sip-ua status** commands in privileged EXEC mode. See the following fields for SIP INFO method statistics:

- OkInfo 0/0, under SIP Response Statistics, Success, displays the number of successful responses to an INFO request.
- Info 0/0, under SIP Total Traffic Statistics, displays the number of INFO messages received and sent by the gateway.



Note

- To see sample output of these show commands, see the [“Configuring Passthrough on a Gateway That Connects to an MTP or Transcoder Gateway”](#) section on page 18.
- To reset the counters for the **show sip-ua statistics** command, use the **clear sip-ua statistics** command.

Configuring SIP NOTIFY-Based Out-of-Band DTMF Relay

To configure the SIP NOTIFY-Based Out-of-Band DTMF Relay feature, perform the following steps.



Note

Cisco proprietary NOTIFY-based out-of-band DTMF relay adds support for devices that do not support in-band DTMF. This configuration must be done on both originating and terminating gateways. With this configuration, DTMF tones are forwarded by using SIP NOTIFY messages in SIP Invites or 18x or 200 Response messages.

SUMMARY STEPS

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

3. **dial-peer voice**
4. **dtmf-relay sip-notify**
5. **exit**
6. **sip-ua**
7. **notify telephone-event max-duration**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the designated dial peer.
Step 4	dtmf-relay sip-notify Example: Router(config-dial-peer)# dtmf-relay sip-notify	Forwards DTMF tones using SIP NOTIFY messages.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.
Step 6	sip-ua Example: Router (config)# sip-ua	Enters SIP user-agent configuration mode.
Step 7	notify telephone-event max-duration time Example: Router(config-sip-ua)# notify telephone-event max-duration 2000	Sets the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. Keyword and argument are as follows: <ul style="list-style-type: none"> • max-duration time—Time, in ms, between consecutive NOTIFY messages for a single DTMF event. Range: 500 to 3000. Default: 2000.
Step 8	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP KPML-Based Out-of-Band DTMF Relay

To configure the SIP KPML Out-of-Band DTMF Relay feature, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice**
4. **dtmf-relay sip-kpml**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the designated dial peer.
Step 4	dtmf-relay sip-kpml Example: Router(config-dial-peer)# dtmf-relay sip-kpml	Forwards DTMF tones using SIP KPML messages.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Verifying SIP DTMF Support

To verify SIP DTMF support, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show running-config**
2. **show sip-ua retry**

3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**
6. **show voip rtp connections**
7. **show sip-ua calls**

DETAILED STEPS

Step 1 **show running-config**

Use this command to show dial-peer configurations.

The following sample output shows that the **dtmf-relay sip-notify** command is configured in dial peer 123:

```
Router# show running-config
.
.
.
dial-peer voice 123 voip
 destination-pattern [12]...
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
```

The following sample output shows that DTMF relay and NTE are configured on the dial peer.

```
Router# show running-config
!
dial-peer voice 1000 pots
 destination-pattern 4961234
 port 1/0/0
!
dial-peer voice 2000 voip
 application session
 destination-pattern 4965678
 session protocol sipv2
 session target ipv4:192.0.2.34
 dtmf-relay rtp-nte
! RTP payload type value = 101 (default)
!
dial-peer voice 3000 voip
 application session
 destination-pattern 2021010101
 session protocol sipv2
 session target ipv4:192.0.2.34
 dtmf-relay rtp-nte
 rtp payload-type nte 110
! RTP payload type value = 110 (user assigned)
!
```

Step 2 **show sip-ua retry**

Use this command to display SIP retry statistics.

```
Router# show sip-ua retry

SIP UA Retry Values
```

```
invite retry count = 6 response retry count = 1
bye retry count = 1 cancel retry count = 1
prack retry count = 10 comet retry count = 10
reliable lxx count = 6 notify retry count = 10
```

Step 3 show sip-ua statistics

Use this command to display response, traffic, and retry SIP statistics.



Tip To reset counters for the **show sip-ua statistics** display, use the **clear sip-ua statistics** command.

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
Trying 4/2, Ringing 2/1,
Forwarded 0/0, Queued 0/0,
SessionProgress 0/0
Success:
OkInvite 1/2, OkBye 0/1,
OkCancel 1/0, OkOptions 0/0,
OkPrack 2/0, OkPreconditionMet 0/0,
OkNotify 1/0, 202Accepted 0/1
Redirection (Inbound only):
MultipleChoice 0, MovedPermanently 0,
MovedTemporarily 0, SeeOther 0,
UseProxy 0, AlternateService 0
Client Error:
BadRequest 0/0, Unauthorized 0/0,
PaymentRequired 0/0, Forbidden 0/0,
NotFound 0/0, MethodNotAllowed 0/0,
NotAcceptable 0/0, ProxyAuthReqd 0/0,
ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
LengthRequired 0/0, ReqEntityTooLarge 0/0,
ReqURITooLarge 0/0, UnsupportedMediaType 0/0,
BadExtension 0/0, TempNotAvailable 0/0,
CallLegNonExistent 0/0, LoopDetected 0/0,
TooManyHops 0/0, AddrIncomplete 0/0,
Ambiguous 0/0, BusyHere 0/0
RequestCancel 1/0, NotAcceptableMedia 0/0
Server Error:
InternalError 0/1, NotImplemented 0/0,
BadGateway 0/0, ServiceUnavail 0/0,
GatewayTimeout 0/0, BadSipVer 0/0,
PreCondFailure 0/0
Global Failure:
BusyEverywhere 0/0, Decline 0/0,
NotExistAnywhere 0/0, NotAcceptable 0/0
SIP Total Traffic Statistics (Inbound/Outbound) /* Traffic Statistics
Invite 3/2, Ack 3/2, Bye 1/0,
Cancel 0/1, Options 0/0,
Prack 0/2, Comet 0/0,
Notify 0/1, Refer 1/0
Retry Statistics /* Retry Statistics
Invite 0, Bye 0, Cancel 0, Response 0,
Prack 0, Comet 0, Reliablelxx 0, Notify 0
```

Following is sample output verifying configuration of the SIP INFO Method for DTMF Tone Generation feature

```
Router# show sip-ua statistics
```



```

SIP Response Statistics (Inbound/Outbound)
Informational:
Trying 1/1, Ringing 0/0,
Forwarded 0/0, Queued 0/0,
SessionProgress 0/1
Success:
OkInvite 0/1, OkBye 1/0,
OkCancel 0/0, OkOptions 0/0,
OkPrack 0/0, OkPreconditionMet 0/0
OkSubscribe 0/0, OkNotify 0/0,
OkInfo 0/0, 202Accepted 0/0
Redirection (Inbound only):
MultipleChoice 0, MovedPermanently 0,
MovedTemporarily 0, SeeOther 0,
UseProxy 0, AlternateService 0
Client Error:
BadRequest 0/0, Unauthorized 0/0,
PaymentRequired 0/0, Forbidden 0/0,
NotFound 0/0, MethodNotAllowed 0/0,
NotAcceptable 0/0, ProxyAuthReqd 0/0,
ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
LengthRequired 0/0, ReqEntityTooLarge 0/0,
ReqURITooLarge 0/0, UnsupportedMediaType 0/0,
BadExtension 0/0, TempNotAvailable 0/0,
CallLegNonExistent 0/0, LoopDetected 0/0,
TooManyHops 0/0, AddrIncomplete 0/0,
Ambiguous 0/0, BusyHere 0/0,
BadEvent 0/0
Server Error:
InternalServerError 0/0, NotImplemented 0/0,
BadGateway 0/0, ServiceUnavail 0/0,
GatewayTimeout 0/0, BadSipVer 0/0
Global Failure:
BusyEverywhere 0/0, Decline 0/0,
NotExistAnywhere 0/0, NotAcceptable 0/0
SIP Total Traffic Statistics (Inbound/Outbound)
  Invite 0/0, Ack 0/0, Bye 0/0,
  Cancel 0/0, Options 0/0,
  Prack 0/0, Comet 0/0,
  Subscribe 0/0, Notify 0/0,
  Refer 0/0, Info 0/0
Retry Statistics
Invite 0, Bye 0, Cancel 0, Response 0, Notify 0

```

Step 4 **show sip-ua status**

Use this command to display status for the SIP user agent.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Session name line (s=) required
  Timespec line (t=) required
  Media supported: audio image

```

```

Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1

```

The following sample output shows that the time interval between consecutive NOTIFY messages for a telephone event is the default of 2000 ms:

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED

SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1

```

The following sample output shows configuration of the SIP INFO Method for DTMF Tone Generation feature.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
SDP application configuration:
Version line (v=) required
Owner line (o=) required
Session name line (s=) required
Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1

```

Step 5 show sip-ua timers

Use this command to display the current settings for SIP user-agent timers.

```

Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 300000, connect 500, disconnect 500
comet 500, prack 500, rellxx 500, notify 500

```

Step 6 show voip rtp connections

Use this command to show local and remote Calling ID and IP address and port information.

Step 7 show sip-ua calls

Use this command to ensure the DTMF method is SIP-KPML.

The following sample output shows that the DTMF method is SIP-KPML.

```
router# show sip-ua calls
SIP UAC CALL INFO

Call 1
SIP Call ID           : 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
  State of the call    : STATE_ACTIVE (7)
  Substate of the call : SUBSTATE_NONE (0)
  Calling Number       :
  Called Number        : 8888
  Bit Flags            : 0xD44018 0x100 0x0
  CC Call ID          : 6
  Source IP Address (Sig) : 172.18.193.251
  Destn SIP Req Addr:Port : 172.18.193.250:5060
  Destn SIP Resp Addr:Port: 172.18.193.250:5060
  Destination Name     : 172.18.193.250
  Number of Media Streams : 1
  Number of Active Streams: 1
  RTP Fork Object      : 0x0
  Media Mode           : flow-through
Media Stream 1
  State of the stream   : STREAM_ACTIVE
  Stream Call ID        : 6
  Stream Type           : voice-only (0)
  Negotiated Codec      : g711ulaw (160 bytes)
  Codec Payload Type    : 0
  Negotiated Dtmf-relay : sip-kpml
  Dtmf-relay Payload Type : 0
  Media Source IP Addr:Port: 172.18.193.251:17576
  Media Dest IP Addr:Port : 172.18.193.250:17468
  Orig Media Dest IP Addr:Port : 0.0.0.0:0

Number of SIP User Agent Client(UAC) calls: 1

SIP UAS CALL INFO

Number of SIP User Agent Server(UAS) calls: 0
```

Configuring SIP Support for SDP

This section describes the procedures for configuring and associating the SIP Support for Asymmetric SDP feature. These procedures include the following:

- [How to Configure a SIP Support for Asymmetric SDP Globally for a Gateway, page 30](#)
- [How to Configure SIP Support for Asymmetric SDP on a Dial Peer, page 31](#)

How to Configure a SIP Support for Asymmetric SDP Globally for a Gateway

To configure an asymmetric payload globally on a SIP network, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service {pots | voatm | vofr | voip}**
4. **sip**
5. **asymmetric payload [dtmf | dynamic-codecs | full]**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service {pots voatm vofr voip} Example: Router(config)# voice service voip	Enters voice-service voip configuration mode.
Step 4	sip Example: Router(conf-voi-ser)# sip	Enters SIP parameters mode.
Step 5	asymmetric payload [dtmf dynamic-codecs full] Example: Router(conf-ser-sip)# asymmetric payload dtmf	Enables the gateway to send and receive DTMF and dynamic codec RTP packets with different payloads. In this example, the asymmetric payload is configured for DTMF.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

How to Configure SIP Support for Asymmetric SDP on a Dial Peer

To configure an asymmetric payload on a dial peer, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice <tag> {pots | vof r| voip}**
4. **voice-class sip**
5. **asymmetric payload [dtmf | dynamic-codecs | full | system]**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <tag> {pots vofr voip} Example: Router(config)# dialpeer voice 111 voip	Enters dial-peer voip configuration mode.
Step 4	voice-class sip Example: Router(conf-dial-peer)# voice-class sip	Enters SIP parameters mode.
Step 5	asymmetric payload [dtmf dynamic-codecs full] Example: Router(conf-dial-peer)# asymmetric payload dynamic-codecs	Enables the gateway to send and receive DTMF and dynamic codec RTP packets with different payloads.
Step 6	exit Example: Router (conf-ser-sip)# exit	Exits the current mode.

Troubleshooting Tips



Note

For general troubleshooting tips and a list of important **debug** commands, see the “[Basic Troubleshooting Procedures](#)” section on page 1.

- To enable debugging for RTP named event packets, use the **debug voip rtp** command.
- To enable KPML debugs, use the **debug kpml** command.
- To enable SIP debugs, use the **debug ccsip** command.
- Collect debugs while the call is being established and during digit presses.
- If an established call is not sending digits though KPML, use the **show sip-ua calls** command to ensure SIP-KPML is included in the negotiation process.

Configuration Examples for SIP DTMF Features

This section provides the following configuration examples:

- [DTMF Relay for SIP Calls Using NTEs: Examples, page 32](#)
- [SIP NOTIFY-Based Out-of-Band DTMF Relay: Example, page 32](#)
- [SIP KPML-Based Out-of-Band DTMF Relay: Example, page 34](#)
- [RFC 2833 DTMF MTP Passthrough: Example, page 37](#)
- [SIP Support for Asymmetric SDP: Example, page 37](#)

DTMF Relay for SIP Calls Using NTEs: Examples

DTMF Relay using RTP-NTE

The following is an example of DTMF relay using RTP-NTE:

```
Router(config)# dial-peer voice 62 voip
Router(config-dial-peer)# session protocol sipv2
Router(config-dial-peer)# dtmf-relay rtp-nte
```

RTP Using Payload Type NTE

The following is an example of RTP Using Payload Type NTE with the default value of 101:

```
Router(config)# dial-peer voice 62 voip
Router(config-dial-peer)# rtp payload-type nte 101
```

SIP NOTIFY-Based Out-of-Band DTMF Relay: Example

```
Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
memory-size iomem 15
```

```
ip subnet-zero
!
no ip domain lookup
!
voice service voip
  redirect ip2ip
sip
  redirect contact order best-match

ip dhcp pool vespa
  network 192.168.0.0 255.255.255.0
  option 150 ip 192.168.0.1
  default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
  codec preference 2 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
  ip address 10.8.17.22 255.255.0.0
  half-duplex
!
interface FastEthernet0/0
  ip address 192.168.0.1 255.255.255.0
  speed auto
  no cdp enable
  h323-gateway voip interface
  h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
  network 10.0.0.0
  network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
  destination-pattern 5100
  port 1/0
!
dial-peer voice 2 pots
```

```

destination-pattern 9998
port 1/1
!
dial-peer voice 123 voip
destination-pattern [12]...
session protocol sipv2
session target ipv4:10.8.17.42
dtmf-relay sip-notify
!
gateway
!
sip-ua
retry invite 3
retry register 3
timers register 150
registrar dns:myhost3.example.com expires 3600
registrar ipv4:10.8.17.40 expires 3600 secondary
!
telephony-service
max-dn 10
max-conferences 4
!
ephone-dn 1
number 4001
!
ephone-dn 2
number 4002
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
login
line vty 5 15
login
!
no scheduler allocate
end

```

SIP KPML-Based Out-of-Band DTMF Relay: Example

```

router(config-dial-peer)#dtmf
router(config-dial-peer)#dtmf-relay ?
  cisco-rtp          Cisco Proprietary RTP
  h245-alphanumeric  DTMF Relay via H245 Alphanumeric IE
  h245-signal        DTMF Relay via H245 Signal IE
  rtp-nte            RTP Named Telephone Event RFC 2833
  sip-kpml           DTMF Relay via KPML over SIP SUBSCRIBE/NOTIFY
  sip-notify         DTMF Relay via SIP NOTIFY messages

router(config-dial-peer)#dtmf-relay sip-kpml
router(config-dial-peer)#end

%SYS-5-CONFIG_I: Configured from console by console
router#sh run

Building configuration...Current configuration : 2430 bytes
!
version 12.4
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec

```



```
no service password-encryption
!
hostname mahoney
!
boot-start-marker
boot-end-marker
!
logging buffered 5000000 debugging
!
no aaa new-model
!
resource policy
!
clock timezone EST 0
ip cef
ip name-server 192.0.2.21
ip name-server 192.0.2.22
!
voice-card 0
!
voice service voip
  sip
    min-se 90
    registrar server
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g729br8
  codec preference 4 g711alaw
  codec preference 5 g726r16
  codec preference 6 g726r24
  codec preference 7 g726r32
  codec preference 8 g723ar53
  codec preference 9 g723ar63
!
!
voice register pool 1
  id ip 192.0.2.168 mask 0.0.0.0
  dtmf-relay rtp-nte
!
!
interface FastEthernet0/0
  ip address 172.18.193.250 255.255.255.0
  no ip proxy-arp
no ip mroute-cache
  duplex auto
  speed auto
!
interface FastEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
!
ip default-gateway 172.20.118.129
ip route 0.0.0.0 0.0.0.0 192.0.2.1
ip route 0.0.0.0 0.0.0.0 172.18.193.1
!
ip http server
!
control-plane
!
voice-port 2/0
```

```
!
voice-port 2/1
!
voice-port 2/2
.
.
voice-port 2/22
!
voice-port 2/23
!
!
dial-peer voice 1 pots
 destination-pattern 8888
 port 2/1
!
dial-peer voice 9999 voip
 destination-pattern 9999
 session protocol sipv2
 session target ipv4:192.0.2.228
 dtmf-relay sip-kpml
 codec g711ulaw
!
dial-peer voice 5555555 voip
 destination-pattern 5555555
 session protocol sipv2
 session target ipv4:172.18.193.97
 codec g711ulaw
!
dial-peer voice 36 voip
 destination-pattern 36601
 session protocol sipv2
 session target ipv4:172.18.193.98
 codec g711ulaw
!
dial-peer voice 444 voip
 destination-pattern 444
 session protocol sipv2
 session target ipv4:172.18.197.154
 codec g711ulaw
!
dial-peer voice 333 voip
 destination-pattern 333
 session protocol sipv2
 session target ipv4:172.18.199.95
!
!
sip-ua
 retry invite 3
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
end
```

RFC 2833 DTMF MTP Passthrough: Example

The following example shows a sample configuration of the RFC 2833 DTMF MTP Passthrough feature on a SIP gateway.

```
dial-peer voice 1000 voip
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.120.70.10
 incoming called-number .T
 dtmf-relay rtp-nte
!
sip-ua
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
!
end
```

SIP Support for Asymmetric SDP: Example

This section contains the following configuration examples:

- [Configuring SIP Support for Asymmetric SDP Globally on a Gateway: Example, page 37](#)
- [Configuring SIP Support for Asymmetric SDP on a Dial Peer: Example, page 37](#)

Configuring SIP Support for Asymmetric SDP Globally on a Gateway: Example

The following example shows how to configure asymmetric SDP globally on a gateway:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# voice service voip
gateway(conf-voi-serv)# sip
gateway(conf-serv-sip)# asymmetric payload dtmf
```

Configuring SIP Support for Asymmetric SDP on a Dial Peer: Example

The following examples shows how to configure asymmetric SDP on a dial peer:

```
gateway> enable
gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
gateway(config)# dial-peer voice 111 voip
gateway(conf-dial-peer)# voice-class sip
gateway(conf-dial-peer)# asymmetric payload dtmf
```

Additional References

General SIP References

- [“SIP Features Roadmap” on page 1](#)—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- [“Basic SIP Configuration” on page 1](#)—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

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Configuring SIP Support for SRTP

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This module contains information about configuring Session Initiation Protocol (SIP) support for the Secure Real-time Transport Protocol (SRTP). SRTP is an extension of the Real-time Transport Protocol (RTP) Audio/Video Profile (AVP) and ensures the integrity of RTP and Real-Time Control Protocol (RTCP) packets that provide authentication, encryption, and the integrity of media packets between SIP endpoints.

SIP support for SRTP was introduced in Cisco IOS Release 12.4(15)T. In this and later releases, you can configure the handling of secure RTP calls on both a global level and on an individual dial peer basis on Cisco IOS voice gateways. You can also configure the gateway (or dial peer) either to fall back to (nonsecure) RTP or to reject (fail) the call for cases where an endpoint does not support SRTP.

The option to allow negotiation between SRTP and RTP endpoints was added for Cisco IOS Release 12.4(20)T and later releases, as was interoperability of SIP support for SRTP on Cisco IOS voice gateways with Cisco Unified Communications Manager. In Cisco IOS Release 12.4(22)T and later releases, you can configure SIP support for SRTP on Cisco Unified Border Elements (Cisco UBEs), as well.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the [“Feature Information for Configuring SIP Support for SRTP”](#) section on page 16.

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



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Prerequisites for Configuring SIP Support for SRTP

- Establish a working IP network and configure VoIP.

**Note**

For information about configuring VoIP, see *Enhancements to the Session Initiation Protocol for VoIP on Cisco Access Platforms* (http://www.cisco.com/en/US/docs/ios/12_2t/12_2t11/feature/guide/ftsipgv1.html).

- Ensure that the gateway has voice functionality configured for SIP.
- Ensure that your Cisco router has adequate memory.
- As necessary, configure the router to use Greenwich Mean Time (GMT). SIP requires that all times be sent in GMT. SIP INVITE messages are sent in GMT. However, the default for routers is to use Coordinated Universal Time (UTC). To configure the router to use GMT, issue the **clock timezone** command in global configuration mode and specify GMT.

Restrictions for Configuring SIP Support for SRTP

- The SIP gateway does not support codecs other than those listed in the table titled “SIP Codec Support by Platform and Cisco IOS Release” in the “Enhanced Codec Support for SIP Using Dynamic Payloads” section of the *Configuring SIP QoS Features* module (http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-qos.html).
- SIP requires that all times be sent in GMT.

Information About Configuring SIP Support for SRTP

The SIP Support for SRTP features use encryption to secure the media flow between two SIP endpoints. Cisco IOS voice gateways and Cisco Unified Border Elements use the Digest method for user authentication and, typically, they use Transport Layer Security (TLS) for signaling authentication and encryption.

**Note**

To provide more flexibility, TLS signaling encryption is no longer required for SIP support of SRTP in Cisco IOS Release 12.4(22)T and later releases. Secure SIP (SIPS) is still used to establish and determine TLS but TLS is no longer a requirement for SRTP, which means calls established with SIP only (and not SIPS) can still successfully negotiate SRTP without TLS signaling encryption. This also means you could configure encryption using a different protocol, such as IPsec.

However, Cisco does not recommend configuring SIP support for SRTP without TLS signaling encryption because doing so compromises the intent of forcing media encryption (SRTP).

When TLS is used, the cryptographic parameters required to successfully negotiate SRTP rely on the cryptographic attribute in the Session Description Protocol (SDP). To ensure the integrity of cryptographic parameters across a network, SRTP uses the SIPS schema (*sips:example.com*). If the Cisco IOS voice gateway or Cisco Unified Border Element is configured to use TLS encryption and sends an invite to an endpoint that cannot provide TLS support, that endpoint rejects the INVITE message. For cases like these, you can configure the gateway or Cisco UBE either to fall back to an RTP-only call or to reject the call.

The SIP support for SRTP features provide the following security benefits:

- Confidentiality of RTP packets—protects packet-payloads from being read by unapproved entities but does so without authorized entities having to enter a secret encryption key.
- Message authentication of RTP packets—protects the integrity of the packet against forgery, alteration, or replacement.
- Replay protection—protects the session address against denial of service attacks.

[Table 1](#) describes the security level of SIP INVITE messages according to which of the four possible combinations of TLS and SRTP is configured.

Table 1 *TLS-SRTP Combinations*

TLS	SRTP	Description
On	On	Signaling and media are secure.
Off	On	Signaling is insecure: <ul style="list-style-type: none"> • If you use the srtp fallback command, the gateway sends an RTP-only SDP. • If you do not configure the srtp fallback command, the call fails and the gateway does not send an INVITE message. <p>Note In Cisco IOS Release 12.4(20)T and later releases (and, for Cisco UBEs, in Cisco IOS Release 12.4(22)T and later releases), calls established with SRTP only (and not SIPS) will succeed even if the srtp fallback command is not configured.</p>
On	Off	RTP-only call.
Off	Off	Signaling and media are not secure.

Cryptographic Parameters

RFC 3711 defines the SRTP cryptographic parameters, including valid syntax and values for attribute *a=crypto* (see [Table 2](#)). Some of these parameters are declarative and apply only to the send direction of the declarer, while others are negotiable and apply to both send and receive directions.

The following shows the cryptographic attribute syntax:

```
a=crypto:<tag> <crypto-suite> <key-params> [<session-params>]
```

Table 2 summarizes the syntax for the cryptographic attribute.

Table 2 Cryptographic Attribute Syntax

Attribute	Optional	Description
tag	No	The tag attribute is a unique decimal number used as an identifier for a particular cryptographic attribute to determine which of the several offered cryptographic attributes was chosen by the answerer.
crypto-suite	No	The crypto-suite attribute defines the encryption and authentication algorithm. Cisco IOS voice gateways and Cisco UBEs support default suite AES_CM_128_HMAC_SHA1_32 (AES-CM encryption with a 128-bit key and HMAC-SHA1 message authentication with a 32-bit tag).
key-params	No	“inline:” <key salt> [“l” lifetime] [“l” MKI “:” length] key salt is base64 encoded contacted master key and salt.
session-params	Yes	The session-params attribute is specific to a given transport and is optional. The gateway does not generate any session-params in an outgoing INVITE message, nor will the SDP library parse them.

SDP Negotiation

To allow calls to succeed between endpoints that do not support SRTP, you can enable SIP support for SRTP by configuring the **srtp** command but you can also enable fall back to the RTP mechanism in cases where SRTP is not supported by one or both endpoints. To configure a Cisco IOS voice gateway or Cisco Unified Border Element to fall back to RTP when SRTP is not supported, configure the **srtp fallback** command either globally or for an individual dial peer.

When the **srtp** command is enabled, the offer SDP in the INVITE message has only one m line for the RTP/SAVP (RTP/Secure AVP) transport type. If a called endpoint does not support SRTP, the call fails with a 4xx error. However, if you configure the **srtp fallback** command, the gateway generates another INVITE message with an RTP-only offer instead of simply allowing the call to fail.



Note In Cisco IOS Release 12.4(20)T and later releases (and, for Cisco UBEs, in Cisco IOS Release 12.4(22)T and later releases), calls established with SRTP only (and not SIPS) will succeed even if the **srtp fallback** command is not configured.

When the gateway or Cisco Unified Border Element is the called endpoint, it will accept an offer with an m line value of SRTP-only, RTP-only, or both SRTP and RTP. For calls with two m lines (SRTP and RTP), the negotiation depends on the configuration of the inbound dial peer or global configuration. Only one m line negotiates—the port number in the other m line is set to 0.

Table 3 summarizes the behavior of the gateway during negotiation.

Table 3 Gateway Behavior During Negotiations

Dial Peer Configuration	INVITE Message Received with SRTP	INVITE Message Received with SRTP and RTP	INVITE Message Received with RTP
SRTP only	SRTP call.	SRTP call, port number in m line of RTP set to 0.	488 (unacceptable media).
SRTP with fallback	SRTP call.	SRTP call, port number in m line of RTP set to 0.	RTP call, port number in m line of SRTP set to 0.
No SRTP	488 (unacceptable media).	RTP call, port number in m line of SRTP set to 0.	RTP call.

The following example shows an offer SDP with two m lines that use the cryptographic attribute for the RTP/SAVP media transport type:

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 7826 3751 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 1789 RTP/AVP 0
a=rtptime:0 PCMU/8000
m=audio 51372 RTP/SAVP 0
a=rtptime:0 PCMU/8000
a=crypto:1 AES_CM_128_HMAC_SHA1_32
inline:d0RmdmcmVCspeEc3QGZiNWpVLFJhQX1cfHAWJSoj|2^20|1:32
```

The following example shows the corresponding answer SDP with SRTP supported:

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 7826 3751 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 0 RTP/AVP 0
a=rtptime:0 PCMU/8000
m=audio 49170 RTP/SAVP 0
a=crypto:1 AES_CM_128_HMAC_SHA1_32
inline:NzB4d1BINUAvLEw6UzF3WSJ+PSdFcGdUJShpX1Zj|2^20|1:32
```

Call Control and Signaling

SIP uses the SRTP library to receive cryptographic keys. If you configure SRTP for the call and cryptographic context is supported, SDP offers the cryptographic parameters. If the cryptographic parameters are negotiated successfully, the parameters are downloaded to the DSP, which encrypts and decrypts the packets. The sender encrypts the payload by using the AES algorithm and builds an authentication tag, which is encapsulated to the RTP packet. The receiver verifies the authentication tag and then decrypts the payload.

Default and Recommended SRTP Settings

Table 4 lists the default and recommended SRTP settings.

Table 4 *Default and Recommended SRTP Settings*

Parameter	Default	Recommended Value
Key derivation rate	0	0—Rekeying is supported
Master key length	128 bits	128 bits
Master salt key length	112 bits	112 bits
MKI indicator	0	0
MKI length	0	0
PRF	AES_CM	128
Session authentication key length	128	128
Session encryption key length	128 bits	128 bits
Session salt key length	112	112
SRTP authentication	HMAC-SHA1	HMAC-SHA1
SRTCP authentication	HMAC-SHA1	HMAC-SHA1
SRTP cipher	AES_CM	AES_CM
SRTCP cipher	AES_CM	NULL
SRTP HMAC tag length	80	32 (voice)—Supported 80 (other)—Not supported
SRTCP HMAC tag length	80	80
SRTP packets maximum lifetime	2 ⁴⁸ packets	2 ⁴⁸ packets
SRTCP packets maximum lifetime	2 ³¹ packets	2 ³¹ packets
SRTP replay-window size	64	64—Not supported
SRTCP replay-window size	64	64—Not supported

Before an SRTP session can be established on a Cisco IOS voice gateway or Cisco UBE, the following cryptographic information must be exchanged in SDP between the two endpoints:

- Crypto suite—crypto algorithm {AES_CM_128_HMAC_SHA1_32} and the supported codec list {g711, G729, G729a}. There could be one or more crypto suites. Cisco IOS Release 12.4(15)T supports only one crypto suite.
- Crypto context—16-byte master key and a 14-byte master salt.

Generating Master Keys

The SRTP library provides an application program interface (API), `srtp_generate_master_key`, to generate a random master key. For encryption and authentication purposes, the key length is 128 bits (master key and session keys). Additionally, RFC 3711 introduces “salting keys”—master salts and sessions salts—and strongly recommends the use of a master salt in the key derivation of session keys. The salting keys (salts) are used to fight against pre-computation and time-memory tradeoff attacks.

The master salt (also known as the n-bit SRTP key) prevents off-line key-collision attacks on the key derivation and, when used, must be random (but can be public). The master salt is derived from the master key and is used in the key derivation of session keys. Session salts, in turn, are used in encryption to counter various attacks against additive stream ciphers. All salting keys (master salt and session salts) are 112 bits.

SRTP Offer and Answer Exchange

If you configure the gateway for SRTP (globally or on an individual dial peer) and end-to-end TLS, an outgoing INVITE message has cryptographic parameters in the SDP.

If you use the **srtp fallback** command and the called endpoint does not support SRTP (offer is rejected with a 4xx class error response), the gateway or Cisco Unified Border Element sends an RTP offer SDP in a new INVITE request. If you do not configure the **srtp fallback** command, the call fails.



Note In Cisco IOS Release 12.4(20)T and later releases (and, for Cisco UBEs, in Cisco IOS Release 12.4(22)T and later releases), calls established with SRTP only (and not SIPS) will succeed even if the **srtp fallback** command is not configured.

When a gateway or Cisco Unified Border Element receives an SRTP offer, negotiation is based on the inbound dial peer if specified and, if not, the global configuration. If multiple cryptographic attributes are offered, the gateway selects an SRTP offer it supports (AES_CM_128_HMAC_SHA1_32). The cryptographic attribute will include the following:

- The tag and same crypto suite from the accepted cryptographic attribute in the offer.
- A unique key the gateway generates from the SRTP library API.
- Any negotiated session parameters and its own set of declarative parameters, if any.

If this cryptographic suite is not in the list of offered attributes, or if none of the attributes are valid, the SRTP negotiation fails. If the INVITE message contains an alternative RTP offer, the gateway (or Cisco UBE) negotiates and the call falls back to (nonsecure) RTP mode. If there is no alternative offer and the SRTP negotiation fails, the INVITE message is rejected with a 488 error (Not Acceptable Media).

Rekeying Rules

There is no rekeying on an SRTP stream. A REINVITE/UPDATE message is used in an established SIP call to update media-related information (codec, destination address, and port number) or other features, such as call-hold. A new key need only be generated if the offer SDP has a new connection address or port. Because the source connection address and port do not change, the gateway or Cisco UBE will not generate a new master key after a key has been established for an SRTP session.

Call-Feature Interactions

This section describes call-feature interactions when SIP Support for SRTP features are configured.

Call Hold

If a gateway receives a call hold REINVITE message after an initial call setup is secured, the gateway places the existing SRTP stream on hold, and its answer in the 200 OK message depends on the offer SDP. If there is a cryptographic attribute in the offer, the gateway responds with a cryptographic attribute in its answer.

Signaling Forking

A proxy can fork an INVITE message that contains an SRTP offer, which can result in multiple SRTP streams until a 200 OK message is received. Because the gateway always honors the last answer, the gateway deletes previous SRTP streams and creates a new stream to the latest endpoint. Other endpoints might also stream to the gateway, but because the DSP knows only the last streams's cryptographic suite and key, authentication on these packets fails, and the packets are dropped.

Call Redirection

A gateway redirects a call when an INVITE message, sent to a proxy or redirect server, results in a 3xx response with a list of redirected contact addresses. The gateway handles a 3xx response based on the schema in the contact of a 3xx message. If the message is SIP, and you configure the call for SRTP with fallback, the gateway offers an SRTP-only redirected INVITE message. If you configure for SRTP only, the offer is SRTP only.

If the schema is SIP, and you use the **srtp fallback** command to configure the call for RTP with fallback, the INVITE message has an RTP offer. If you do not configure the **srtp fallback** command, the call fails.



Note In Cisco IOS Release 12.4(20)T and later releases (and, for Cisco UBEs, in Cisco IOS Release 12.4(22)T and later releases), calls established with SRTP only (and not SIPS) will succeed even if the **srtp fallback** command is not configured.

Call Transfer

The SIP Support for SRTP feature interaction with call transfer depends on your outbound dial peer or global configuration. During a call transfer, the gateway sends an INVITE message to establish the connection to the transfer target. The gateway includes an SRTP offer in the INVITE message if the outbound dial peer or global configuration includes the SRTP offer.

T.38 Fax

The T.38 transport supported is User Datagram Protocol (UDP). A T.38 call is initiated as a voice call, which can be RTP or SRTP, and when it switches to T.38 fax mode, the fax call is not secure. When the fax is switched back to voice, the call returns to its initial voice state.

Conferencing Calls

For conferencing calls, the incoming INVITE message does not match any inbound dial peer and the message body is sent to the application in a container. The conferencing application performs the necessary negotiation and replies through PROGRESS or CONNECT events.

How to Configure SIP Support for SRTP

Before configuring SIP support for SRTP on a gateway or Cisco Unified Border Element, it is strongly recommended you first configure SIPS either globally or on an individual dial peer basis. The configuration on a dial peer overrides the global configuration.

This section contains the following configurations:

- [Configuring SIPS Globally, page 9](#) (optional)
- [Configuring SIPS on a Dial Peer, page 10](#) (optional)
- [Configuring SRTP and SRTP Fallback Globally, page 11](#) (required)
- [Configuring SRTP and SRTP Fallback on a Dial Peer, page 12](#) (optional)

Configuring SIPS Globally

To configure secure SIP (SIPS) globally on a Cisco IOS voice gateway or Cisco Unified Border Element, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service {pots | voatm | vofr | voip}**
4. **sip**
5. **url sips**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service {pots voatm vofr voip} Example: Router(config)# voice service voip	Enters voice service configuration mode.
Step 4	sip Example: Router(conf-voi-serv)# sip	Enters SIP configuration mode.

	Command or Action	Purpose
Step 5	<code>url sips</code> Example: Router(conf-serv-sip)# url sips	Specifies generation of URLs in SIPS format for VoIP calls for all dial peers on the voice gateway or Cisco UBE.
Step 6	<code>exit</code> Example: Router(conf-serv-sip)# exit	Exits the current mode.

Configuring SIPS on a Dial Peer

To configure secure SIP (SIPS) on an individual dial peer, perform the following steps.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `dial-peer voice tag {pots | vofr | voip}`
4. `voice-class sip url sips`
5. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.
Step 3	<code>dial-peer voice tag {pots vofr voip}</code> Example: Router(config)# dial-peer voice 111 voip	Enters dial peer voice configuration mode.

	Command or Action	Purpose
Step 4	voice-class sip url sips Example: Router(config-dial-peer)# voice-class sip url sips	Specifies configuration of URLs in SIPS format for VoIP calls for a specific dial peer.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring SRTP and SRTP Fallback Globally

To configure SRTP and SRTP fallback behavior globally on a Cisco IOS voice gateway or Cisco Unified Border Element, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service {pots | voatm | vofr | voip}**
4. **srtp**
5. **srtp fallback**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service {pots voatm vofr voip} Example: Router(config)# voice service voip	Enters voice service configuration mode.
Step 4	srtp Example: Router(conf-voi-serv)# srtp	Configures secure RTP calls.

	Command or Action	Purpose
Step 5	srtp fallback Example: Router(conf-voi-serv)# srtp fallback	Configures a fallback to RTP calls in case secure RTP calls fail due to lack of support from an endpoint.
Step 6	exit Example: Router(conf-voi-serv)# exit	Exits the current mode.

Configuring SRTP and SRTP Fallback on a Dial Peer

To configure SRTP and SRTP fallback behavior on an individual dial peer that overrides the global SRTP configuration, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag {pots | vofr | voip}**
4. **srtp**
5. **srtp fallback**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag {pots vofr voip} Example: Router(config)# dial-peer voice 111 voip	Enters dial peer voice configuration mode.
Step 4	srtp Example: Router(config-dial-peer)# srtp	Configures secure RTP calls.

	Command or Action	Purpose
Step 5	srtp fallback Example: Router(config-dial-peer)# srtp fallback	Configures a fallback to RTP calls in case secure RTP calls fail due to lack of support from an endpoint.
Step 6	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuration Examples for Configuring SIP Support for SRTP

The following example shows how to configure SIPS globally on a Cisco IOS voice gateway or Cisco Unified Border Element:

```
Router> enable
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# url sips
Router(conf-serv-sip)# exit
```

The following example shows how to configure SIPS on dial peer 111:

```
Router> enable
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# dial-peer voice 111 voip
Router(config-dial-peer)# voice-class sip url sips
Router(config-dial-peer)# exit
```

The following example shows how to configure for SRTP with fallback to RTP globally on a Cisco IOS voice gateway or Cisco Unified Border Element:

```
Router> enable
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# voice service voip
Router(conf-voi-serv)# srtp
Router(conf-voi-serv)# srtp fallback
Router(conf-voi-serv)# exit
```

The following example shows how to configure for SRTP with fallback to RTP on dial peer 111:

```
Router> enable
Router# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)# dial-peer voice 111 voip
Router(config-dial-peer)# srtp
Router(config-dial-peer)# srtp fallback
Router(config-dial-peer)# exit
```

Additional References

The following sections provide references related to configuring the SIP Support for SRTP features.

Related Documents

Related Topic	Document Title
Cisco IOS dial peer overview	Dial Peer Overview
Cisco IOS dial technologies command information	Cisco IOS Dial Technologies Command Reference
Cisco IOS SIP features, listed by release	SIP Features Roadmap
Cisco IOS SIP overview and related documents	Overview of SIP
Cisco IOS software configuration guides	<ul style="list-style-type: none"> Cisco IOS Dial Technologies Configuration Guide Cisco IOS SIP Configuration Guide <p>Note To locate the configuration guide specific to your Cisco IOS software release, choose the Cisco IOS and NX-OS Software category on the Product Support page and navigate according to your release (http://www.cisco.com/web/psa/products/index.html).</p>
Cisco IOS voice command information	Cisco IOS Voice Command Reference
Cisco IOS voice configuration information	Cisco IOS Voice Configuration Library
Cisco Unified Border Element configuration information	Cisco Unified Border Element Configuration Guide
Cisco Unified CME command information	Cisco Unified Communications Manager Express Command Reference
Cisco Unified CME configuration information	Cisco Unified CME Support Documentation Home Page

RFCs

RFC	
draft-ietf-mmusic-sdescriptions-08.txt	Session Description Protocol Security Descriptions for Media Streams
RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers
RFC 3711	The Secure Real-time Transport Protocol (SRTP)

MIBs

MIB	MIBs Link
None	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

Technical Assistance

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

Feature Information for Configuring SIP Support for SRTP

Table 5 lists the features in this module and provides links to specific configuration information. Only features that were introduced or modified in Cisco IOS Release 12.2(1) or Cisco IOS Releases 12.2(1) or 12.0(3)S or a later release appear in the table.

For information on a feature in this technology that is not documented here, see the *Cisco IOS SIP Features Roadmap* module at http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-roadmap.html.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

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


Note

Table 5 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release. Unless noted otherwise, subsequent releases of that Cisco IOS software release also support that feature.

Table 5 Feature Information for Configuring SIP Support for SRTP

Feature Name	Releases	Feature Information
SIP Support for SRTP	12.4(15)T	This feature introduces SIP support for supplementary services features, such as call hold, call transfer, call waiting, and call conference (3WC) using hook flash (HF) for FXS phones on Cisco IOS voice gateways. The following commands were introduced or modified: srtp, srtp fallback.
SIP SRTP Fallback to Nonsecure RTP	12.4(15)XY 12.4(20)T	This feature extends the existing SRTP to RTP fallback on Cisco IOS voice gateways to support a delayed offer and adds support for SRTP over SIP. The following commands were introduced or modified: srtp negotiate, voice-class sip srtp negotiate.
Interworking of Secure RTP calls for SIP and H323	12.4(20)T	This feature provides an option for a Secure RTP (SRTP) call to be connected from H323 to SIP and from SIP to SIP. Additionally, this feature extends SRTP fallback support from the Cisco IOS voice gateway to the Cisco Unified Border Element. This feature uses no new or modified commands.

Table 5 *Feature Information for Configuring SIP Support for SRTP (continued)*

Feature Name	Releases	Feature Information
SIP SRTP Fallback to Nonsecure RTP for Cisco Unified Border Elements	12.4(22)T	This feature adds support for both SRTP to RTP fallback with a delayed offer and SRTP over SIP to the Cisco Unified Border Element. This feature uses no new or modified commands.
Cisco Unified Border Element Support for SRTP-RTP Internetworking	12.4(22)YB	This feature provides the ability to support interworking between SRTP on one IP leg and RTP on another IP leg of a Cisco Unified Border Element. The following command was introduced or modified: tls .

Glossary

AVP—Audio/Video Profile.

CAC—Call Admission Control.

CME—Communications Manager Express.

CVP—Customer Voice Portal.

GW—gateway.

ISDN—Integrated Services Digital Network.

MIME—Multipurpose Internet Mail Extensions.

m line—The media-level section of an SDP session begins and ends with an “m” line that confines the information about the media stream.

MOH—music on hold.

OGW—originating gateway (ingress gateway).

PBX—Private Branch Exchange.

PINX—private integrated services network exchange.

PISN—private integrated services network.

QoS—quality of service.

QSIG—Q Signaling protocol.

RSVP—Resource Reservation Protocol.

RTP—Real-time Transport Protocol.

SDP—Session Description Protocol.

SIP—Session Initiation Protocol.

SRTP—Secure Real-time Transport Protocol.

TDM—time-division multiplexing.

TGW—terminating gateway (egress gateway).

UA—user agent.

UDP—User Datagram Protocol.

URI—uniform resource identifier.

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Configuring SIP Support for Hookflash

This chapter contains information about the SIP Support for Hookflash feature that allows you to configure IP Centrex supplementary services on SIP-enabled, Foreign Exchange Station (FXS) lines. Supplementary services for the SIP Support for Hookflash feature include the following:

- Call hold
- Call waiting
- Call transfer
- 3-Way conferencing

Use the **service dsapp** command to configure supplementary Centrex-like features on FXS phones to interwork with SIP-based softswitches. The SIP Support for Hookflash feature supports the concept of a dual-line (ACTIVE and STANDBY for active and held calls) for FXS calls to support supplementary services. Hookflash triggers supplementary services based on the current state of the call.

You can configure the **service dsapp** command on individual dial peers, or configure globally for all calls entering the gateway.

Feature History for SIP Support for Hookflash

Release	Modification
12.4(11)T	This feature was introduced.

Contents

- [Prerequisites for SIP Support for Hookflash, page 2](#)
- [Information About SIP Support for Hookflash, page 2](#)
- [How to Configure and Associate SIP Support for Hookflash, page 11](#)
- [Configuration Examples for SIP Support for Hookflash, page 20](#)
- [Additional References, page 24](#)



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Prerequisites for SIP Support for Hookflash

All Hookflash Features for FXS Ports

- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network. For information on configuring IP, see the [Cisco IOS IP Configuration Guide](#), Release 12.3.
- Configure VoIP.

Information About SIP Support for Hookflash

Use the **service dsapp** command to configure supplementary Centrex-like services on FXS phones to interwork with SIP-based softswitches. Hookflash triggers supplementary features based on the current state of the call and provides a simulation of dual-line capability for analog phones to allow one line to be active while the other line is used to control supplementary IP Centrex services. Supplementary services for the SIP Support for Hookflash feature include the following:

- [Call Hold, page 2](#)
- [Call Waiting, page 4](#)
- [Call Transfers, page 5](#)
- [3-Way Conference, page 9](#)

Call Hold

With the Call Hold feature, you can place a call on hold. When you are active with a call and you press hookflash, and there is no call that is waiting, you hear a dial tone.

If there is a call on hold, the hookflash switches between two calls; the call on hold becomes active while the active call is put on hold.

If you have a call on hold and the call hangs up, the call on hold is disconnected.

Call Holding Flows

The sequence of placing a call on hold is summarized in the following steps:

1. User A and user B are active with a call.
2. By pressing hookflash, user A initiates a call hold.
3. SIP sends a call hold indication to user B.
4. User A can now initiate another active call (user C), transfer the active call (call transfer), or respond to a call-waiting indication.

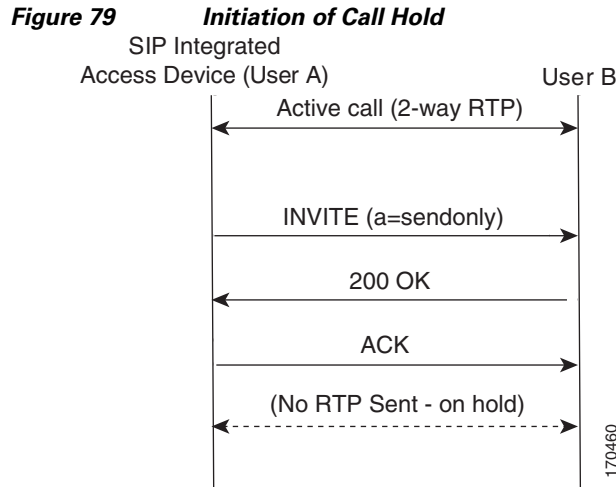


Note

Use the **offer call-hold** command in sip-ua configuration mode to configure the method of hold used on the gateway. For detailed information on the **offer call-hold** command, see the [Cisco IOS Voice Command Reference Guide](#).

5. User A receives a second dial tone and presses hookflash.

Figure 79 shows the initiation of the calls hold sequence.



6. User A and User B reconnect.

Figure 80 shows the calls on hold resume sequence.

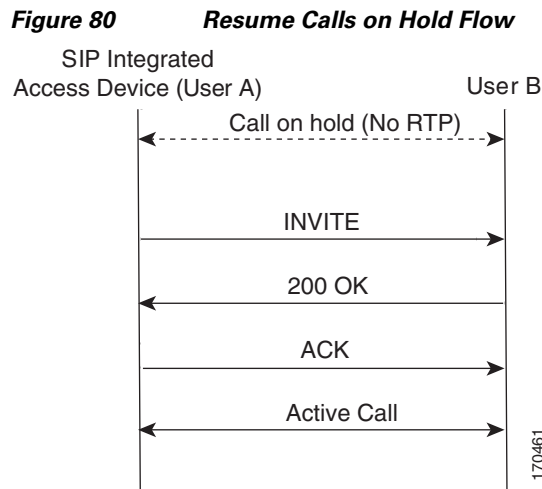


Table 48 summarizes the hookflash support for Call Hold services.

Table 48 Call Hold Hookflash Services

State	Action	Result	Response to FXS Line
Active call	Hookflash	Call placed on hold for remote party.	Second dial tone for FXS phone.
Call on hold	Hookflash	Active call.	FXS line connects to call.

Table 48 Call Hold Hookflash Services

State	Action	Result	Response to FXS Line
Call on hold and active call	Hookflash	Active and call on hold are swapped.	FXS line connects to previous held call.
	On hook	Active call is dropped.	Held call still active. Reminder ring on FXS line.
	Call on hold goes on hook	Call on hold is dropped.	None.
	Active call goes on hook	Active call is dropped.	Silence. Reconnects to held call after the value you specify for disc-toggle-time expires. See How to Configure Disconnect Toggle Time , page 16.

Call Waiting

With the Call Waiting feature, you can receive a second call while you are on the phone with another call. When you receive a second call, you hear a call-waiting tone (a tone with a 300 ms duration). Caller ID appears on phones that support caller ID. You can use hookflash to answer a waiting call and place the previously active call on hold. By using hookflash, you can toggle between the active and a call that is on hold. If the Call Waiting feature is disabled, and you hang up the current call, the second call will hear a busy tone.

The call-waiting sequence is summarized in the following steps:

1. User A is active with a call to user B
2. User C calls user B
3. User B presses hookflash.
The call between user A and user B is held.
4. User B connects to user C.

[Figure 81](#) shows the call waiting sequence.

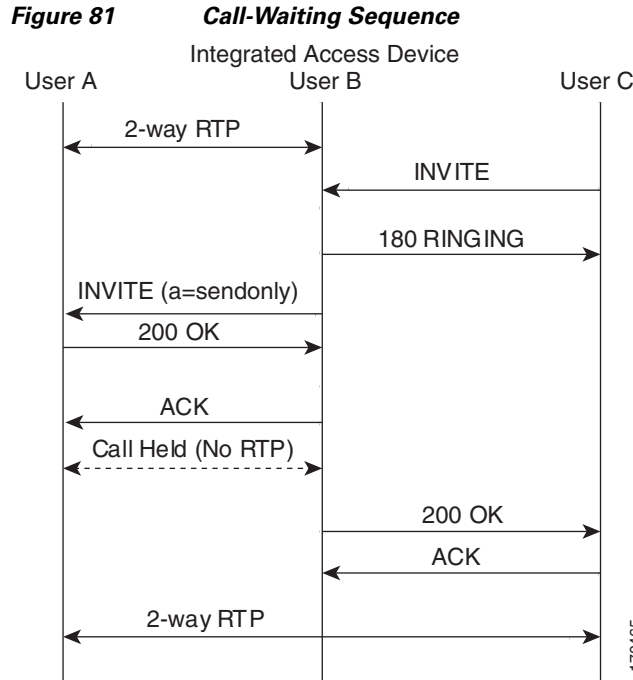


Table 49 summarizes hookflash support for Call Waiting services.

Table 49 Call Waiting Hookflash Services

State	Action	Result	Response to FXS Line
Active call and waiting call	Hookflash	Swap active call and waiting call.	FXS line connects to waiting call.
	Active call disconnects	Active call is disconnected.	Silence.
	Waiting call goes disconnects	Stay connected to active call.	None.
	Call disconnects	Active call is dropped.	Reminder ring on FXS line.

Call Transfers

Call transfers are when active calls are put on hold while a second call is established between two users. After you establish the second call and terminate the active call, the call on hold will hear a ringback. The Call Transfer feature supports all three types of call transfers—blind, semi-attended, and attended.

Blind Call Transfer

The following describes a typical Blind call-transfer scenario:

1. User A calls user B.

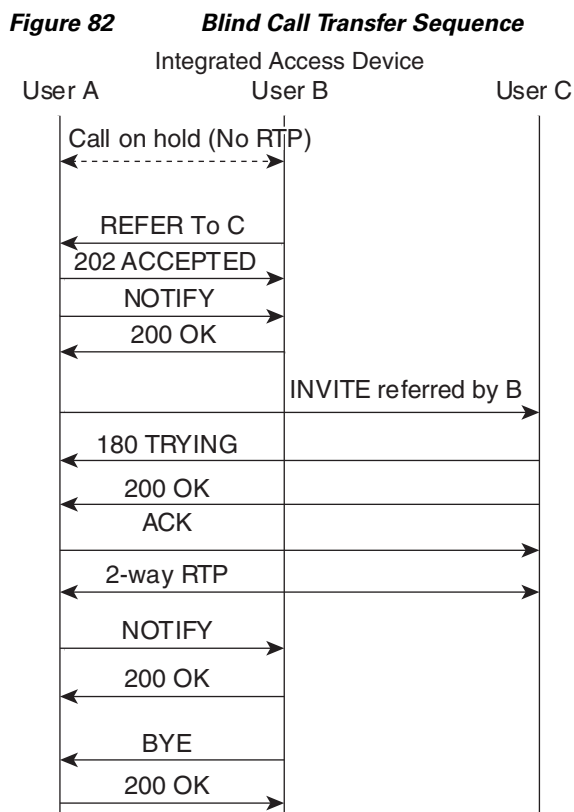
- User B (transferer) presses hookflash, places user A (transferee) on hold, and dials user C (transfer-to).



Note User B will not hear alerting for the time you configure. See [How to Configure Blind Transfer Wait Time](#), page 17.

- Before the Blind call transfer trigger timer expires, user B disconnects, and the call between user A and user B is terminated.
- User A is transferred to user C and hears a ringback if user C is available. If user C is busy, user A hears a busy tone; if user C is not busy and answers, user A and user C connect.

Figure 82 shows the call sequence for a Blind call transfer.



Semi-Attended Transfers

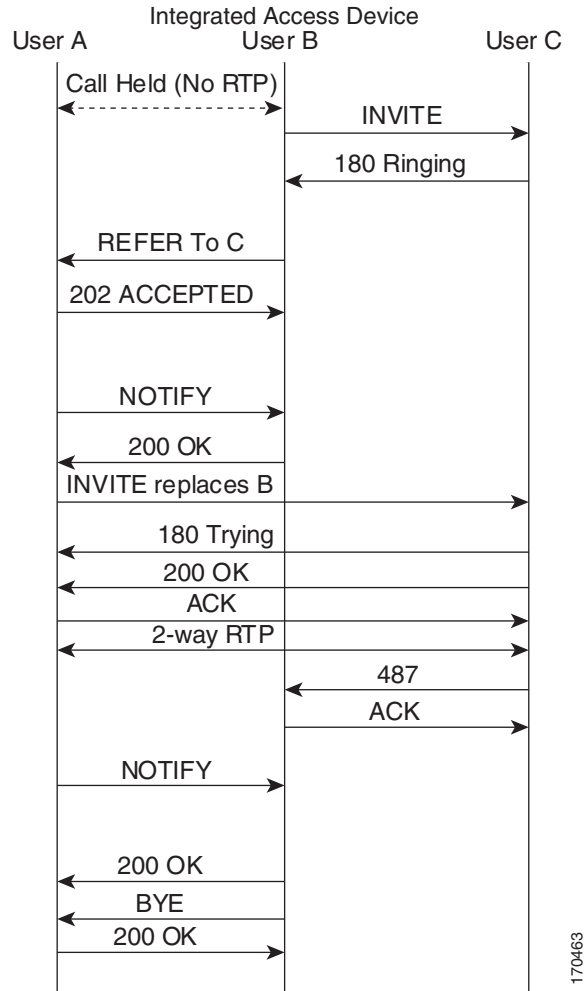
The following is a typical semi-attended transfer scenario:

- User A calls user B.
- User B places user A on hold and dials user C.
- After user B hears a ringback and user C rings, user B initiates a transfer, and the call between user A and user B is terminated.
- User A is transferred to user C and hears a ringback if user C is available. If user C is busy, user A hears a busy tone.

- If user C is not busy and answers, user A and user C connect.

Figure 83 shows the call details for a semi-attended transfer.

Figure 83 *Initiation of Semi-Attended Transfer*



Attended Transfers

The following describes a typical attended transfer:

- User A calls user B.
- User B places user A on hold and dials user C.
- After user C answers, user B goes on-hook to initiate a transfer, and the call between user A and user B is terminated.
- User A is transferred to user C. If user C is busy when user B calls, user A hears a busy tone.
- If user C is not busy and answers, user A and user C connect.

Figure 84 shows the call details for an attended transfer.

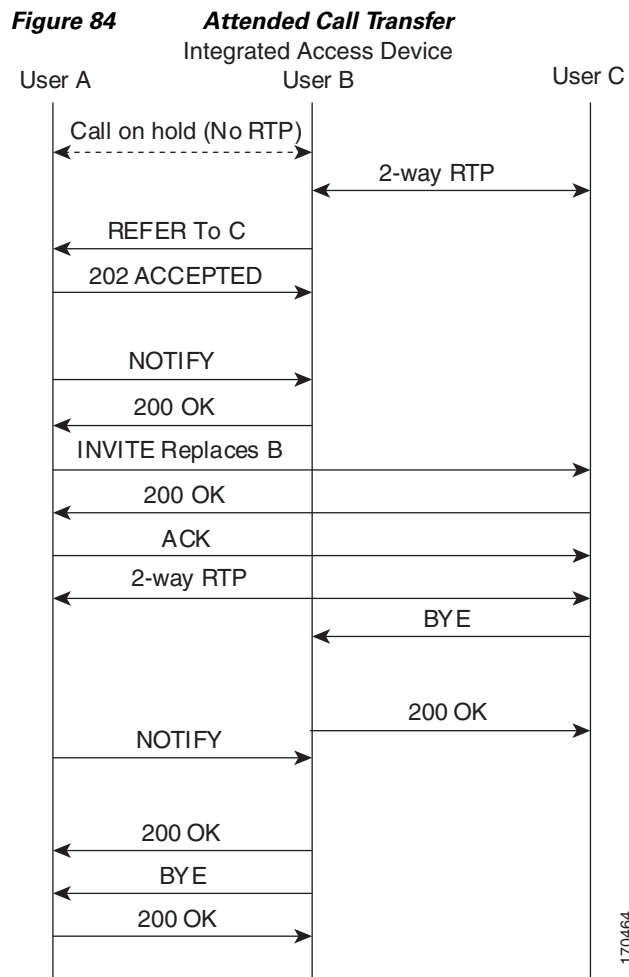


Table 50 summarizes the hookflash support for Call Transfer services.

Table 50 *Call Transfer Hook Flash Services*

State	Action	Result	Response to FXS Line
Active call	Hookflash.	Call placed on hold.	Second dial tone.
Call on hold and outgoing dialed or alerting, or active call	On hook.	Call on hold and active call transferred.	—
Call on hold and outgoing alerting call	Hookflash	Active call dropped.	FXS line connects to call on hold.

3-Way Conference

You can use the 3-Way Conference feature to establish two calls with a single connection so that all three parties can talk together. If the 3-Way Conference feature is disabled, a second hookflash will toggle between the two calls.



Note

The 3-Way Conference feature only supports those SIP calls using the g711 codex. This feature also supports specification GR-577-CORE.

Setting Up a 3-Way Conference

The following describes a typical 3-way conference scenario:

1. User A is talking with user B (a second-party call).
2. User A presses hookflash, receives a dial-tone, and dials user C.
3. User C answers. User A and user C are active in a second-party call.
4. User A presses hookflash to activate a 3-way conference.

In other terminology, user A is the host or controller; user B is the original call, and user C is the add-on.



Note

The 3-Way Conference feature is available when the second-party call is outgoing. If the second-party call is incoming and you press hookflash, the phone toggles between the two calls.

Figure 85 shows the call details for 3-way conferencing.

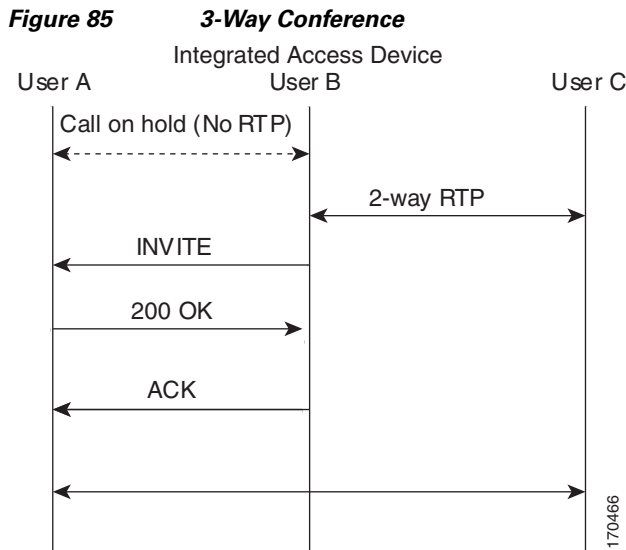


Table 51 summarizes the hookflash support for 3-way conferencing services.

Table 51 3-Way Conference Hookflash Services

State	Action	Result	Response to FXS Line
Active call	Hookflash	Call place on hold.	Second dial tone
Call on hold and active call		Join call on hold and active call.	Media mixing of both calls

Terminating a 3-Way Conference

Table 52 summarizes the termination of a 3-way conference:

Table 52 3-Way Conference Termination

State	Action	Result	Response to FXS Line
Active 3-way conference	User A disconnects first	3-Way conference terminates; all users are disconnected.	Dial tone
	User B disconnects first	User A and user C establish a second-party call.	FXS line connects user A and user C.
	User C disconnects first	User A and user B establish a second-party call.	FXS line connects user A and user B.
	User A presses hookflash	User C disconnects and user A and user B establish a second-party call.	FXS line connects user A and user B.

How to Configure and Associate SIP Support for Hookflash

This section describes the procedures for configuring and associating the SIP Support for Hookflash feature. These procedures include the following:

1. Configuring supplementary service by using the **service dsapp** command.
2. Associating the supplementary services with configured dial peers.

or

Associating the supplementary services as the global default application on a gateway.

This section provides configurations for the following supplementary services and provides configuration for associating supplementary services with dial peers:

- [How to Configure Call Hold, page 11](#)
- [How to Configure Call Waiting, page 13](#)
- [How to Configure Call Transfer, page 14](#)
- [How to Configure 3-Way Conferencing, page 15](#)
- [How to Configure Disconnect Toggle Time, page 16](#)
- [How to Configure Blind Transfer Wait Time, page 17](#)
- [How to Associate Services with a Fixed Dial Peer, page 18](#)
- [How to Associate Services Globally on a Gateway, page 19](#)

How to Configure Call Hold

To configure the Call Hold feature, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. application
4. service dsapp
5. param callHold
6. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router(config)# application	Enters SIP gateway-application configuration mode.
Step 4	service dsapp Example: Router(config-app)# service dsapp	Enters DSAPP parameters mode.
Step 5	param callHold TRUE Example: Router(config-app-param)# param callHold TRUE	Enables call hold.
Step 6	exit Example: Router (config-app-param)# exit	Exits the current mode.

How to Configure Call Waiting

To enable call waiting for a DSAPP, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **service dsapp**
5. **param callWaiting**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router(config)# application	Enters SIP gateway-application configuration mode.
Step 4	service dsapp Example: Router(config-app)# service dsapp	Enters DSAPP parameters mode.
Step 5	param callWaiting TRUE Example: Router(config-app-param)# param callWaiting TRUE	Enables call waiting.
Step 6	exit Example: Router (config-app-param)# exit	Exits the current mode.

How to Configure Call Transfer

To configure call transfer, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **service dsapp**
5. **param callTransfer**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router (config)# application	Enters SIP gateway-application configuration mode.
Step 4	service dsapp Example: Router(config-app)# service dsapp	Enters DSAPP parameters mode.
Step 5	param callTransfer TRUE Example: Router(config-app-param)# param callTransfer TRUE	Enables call transfer.
Step 6	exit Example: Router(config-app-param)# exit	Exits the current mode.

How to Configure 3-Way Conferencing

To configure 3-way conferencing, follow these steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **service dsapp**
5. **param callConference**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router (config)# application	Enters SIP gateway-application configuration mode.
Step 4	service dsapp Example: Router(config-app)# service dsapp	Enters DSAPP parameters mode.
Step 5	param callConference TRUE Example: Router(config-app-param)# param callConference TRUE	Enables 3-way conferencing.
Step 6	exit Example: Router(config-app-param)# exit	Exits the current mode.

How to Configure Disconnect Toggle Time

You can configure the time to wait before switching to a call on hold if an active call disconnects (commonly known as disconnect toggle time). You can configure a time-to-wait range between 10 (default) and 30 seconds.

To configure disconnect toggle time, proceed with the following steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **service dsapp**
5. **param disc-toggle-time**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router (config)# application	Enters SIP gateway-application configuration mode.
Step 4	service dsapp Example: Router(config-app)# service dsapp	Enters DSAPP parameters mode.
Step 5	param disc-toggle-time 20 Example: Router(config-app-param)# param disc-toggle-time 20	Sets the time to wait before switching to a call on hold, if the active call disconnects (disconnect toggle time). You can specify a disconnect toggle time between 10 (default) and 30 seconds.
Step 6	exit Example: Router(config-app-param)# exit	Exits the current mode.

How to Configure Blind Transfer Wait Time

To configure the time the system waits before establishing a call, so that you can transfer a call by placing the phone on hook, proceed with the following steps.

**Note**

The transferrer will not hear alerting for the time you configure because the system delays the call in case blind transfer is initiated.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **service dsapp**
5. **param blind-xfer-wait-time**

6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> <code>enable</code>	Enables privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# <code>configure terminal</code>	Enters global configuration mode.
Step 3	<code>application</code> Example: Router(config)# <code>application</code>	Enters SIP gateway-application configuration mode.
Step 4	<code>service dsapp</code> Example: Router(config-app)# <code>service dsapp</code>	Enters DSAPP parameters mode.
Step 5	<code>param blind-xfer-wait-time</code> Example: Router(config-app-param)# <code>param blind-xfer-wait-time 10</code>	Enables call waiting.
Step 6	<code>exit</code> Example: Router (config-app-param)# <code>exit</code>	Exits the current mode.

How to Associate Services with a Fixed Dial Peer

After you have enabled and customized your services on a gateway by using the **service dsapp** command, you must associate these services with configured dial peers. You can associate individual dial peers, or alternately, you can configure these services globally on the gateway (see [How to Associate Services Globally on a Gateway, page 19](#)). If you associate these services globally, all calls entering from the FXS line side and from the SIP trunk side invoke the **service dsapp** services.

To configure a fixed dial peer used by DSAPP to set up a call to the SIP server (trunk) side, proceed with the following steps:

SUMMARY STEPS

- `enable`
- `configure terminal`
- `application`

4. `service dsapp`
5. `param dialpeer dialpeer tag`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> <code>enable</code>	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# <code>configure terminal</code>	Enters global configuration mode.
Step 3	<code>application</code> Example: Router(config)# <code>application</code>	Enters SIP gateway-application configuration mode.
Step 4	<code>service dsapp</code> Example: Router (config-app)# <code>service dsapp</code>	Enters DSAPP parameters mode.
Step 5	<code>param dialpeer <i>dialpeer tag</i></code> Router(config-app-param)# <code>param dialpeer 5000</code>	Configures a fixed dial peer used by DSAPP to set up a call to the SIP server (trunk) side, where <i>dial peer tag</i> is the tag of the dial peer used to place an outgoing call on the IP trunk side. The <i>dial peer tag</i> must be the same tag as the dial peer configured to the SIP server.
Step 6	<code>exit</code> Example: Router(config-app-param)# <code>exit</code>	Exits the current mode.

How to Associate Services Globally on a Gateway

After you have enabled and customized your services on a gateway by using the **service dsapp** command, you must associate these services with configured dial peers. You can associate individual dial peers (see [How to Associate Services with a Fixed Dial Peer, page 18](#)), or alternately, you can configure these services globally on the gateway. If you associate these services globally, all calls entering from the FXS line side and from the SIP trunk side will invoke the **service dsapp** services.

To associate services globally on a gateway, proceed with the following steps:

SUMMARY STEPS

1. `enable`
2. `configure terminal`

3. application
4. global
5. service default dsapp
6. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	application Example: Router(config)# application	Enters SIP gateway-application configuration mode.
Step 4	global Example: Router (config-app)# global	Enters SIP gateway-application-global configuration mode.
Step 5	service default dsapp Example: Router (config-app-global)# service default dsapp	Globally sets dsapp as the default application. All calls entering the gateway (from the FXS line side and the SIP trunk side) invoke the dsapp application.
Step 6	exit Example: Router(config-app-global)# exit	Exits the current mode.

Configuration Examples for SIP Support for Hookflash

This section contains the following configuration examples:

- [Configuring Call Hold: Example, page 21](#)
- [Configuring Call Waiting: Example, page 21](#)
- [Configuring Call Transfer: Example, page 21](#)
- [Configuring 3-Way Conferencing: Example, page 21](#)
- [Configuring Disconnect Toggle Time: Example, page 21](#)
- [Configuring Blind Transfer Wait Time: Example, page 22](#)

- [Configuring a Fixed Dial Peer Used for Outgoing Calls on SIP Trunk Side: Example, page 22](#)
- [Associating Services with a Fixed Dial Peer: Example, page 22](#)
- [Associating Services Globally on a Gateway: Example, page 23](#)

Configuring Call Hold: Example

The following example shows how to enable the Call Hold feature:

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param callHold TRUE
```

Configuring Call Waiting: Example

The following example shows the configuration to enable the Call Waiting feature.

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param callWaiting TRUE
```

Configuring Call Transfer: Example

The following example shows how to enable the Call Transfer feature:

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param callTransfer TRUE
```

Configuring 3-Way Conferencing: Example

The following example shows how to enable the 3-Way Conferencing feature:

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param callConference TRUE
```

Configuring Disconnect Toggle Time: Example

In this example, a disconnect toggle time is configured; the toggle time specifies the amount of time in seconds the system waits before committing the call transfer, after the originating call is placed on hook.

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
```

```
Gateway(config-app-param)# param disc-toggle-time 10
```

Configuring Blind Transfer Wait Time: Example

In this example, a blind transfer wait time is configured that specifies the amount of time in seconds the system waits before committing the call transfer after the originating call is placed on hook.

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param blind-xfer-wait-time 10
```

Configuring a Fixed Dial Peer Used for Outgoing Calls on SIP Trunk Side: Example

In this example, a fixed dial peer is configured to set up the call to the SIP server (trunk) side.

```
Gateway# configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
Gateway(conf)# application
Gateway(config-app)# service dsapp
Gateway(config-app-param)# param dialpeer 5000
```

Associating Services with a Fixed Dial Peer: Example

In this example, a fixed dial peer is configured to set up the call to the SIP server (trunk) side. The line in bold shows the dial peer statement.

```
Gateway# show running log
.
!
application
  service dsapp
  param dialpeer 1234
  param disc-toggle-time 15
  param callWaiting TRUE
  param callConference TRUE
  param blind-xfer-wait-time 10
  param callTransfer TRUE
!

voice-port 1/0/0
  station-id-name Example1
  station-id number 1234567890
!
voice-port 1/0/1
  station-id-name Example2
  station-id number 1234567891
!
voice-port 1/0/2
  station-id-name Example31
  station-id number 1234567892
!
dial-peer voice 1234 voip
  service dsapp
```

```

destination-pattern.T
session protocol sipv2
session target ipv4:10.1.1.1
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 100 pots
service dsapp
destination-pattern.1234567890
port 1/0/0
prefix 1234567890
!
dial-peer voice 101 pots
service dsapp
destination-pattern.1234567891
port 1/0/1
prefix 1234567891
!
dial-peer voice 102 pots
service dsapp
destination-pattern.1234567892
port 1/0/2
prefix 1234567892
!
!
sip-ua
registrar ipv4:10.1.1.1 expires 3600
!

```

Associating Services Globally on a Gateway: Example

In this example, the gateway is associated globally with supplementary services. The lines in bold show the dial peer statement.

```

Gateway# show running log
.
!
application
service dsapp
param disc-toggle-time 15
param callWaiting TRUE
param callConference TRUE
param blind-xfer-wait-time 10
param callTransfer TRUE
!

voice-port 1/0/0
station-id-name Example1
station-id number 1234567890
!
voice-port 1/0/1
station-id-name Example2
station-id number 1234567891
!
voice-port 1/0/2
station-id-name Example31
station-id number 1234567892
!
dial-peer voice 1234 voip
service dsapp
destination-pattern 1800T
session protocol sipv2

```

```
session target ipv4:10.1.1.1
dtmf-relay rtp-nte
codec g729r8
!
dial-peer voice 9753 voip
service dsapp
destination-pattern.T
session protocol sipv2
session target ipv4:10.1.1.1
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 100 pots
preference 8
service dsapp
destination-pattern.6234567890
port 1/0/0
prefix 6234567890
!
dial-peer voice 101 pots
preference 8
service dsapp
destination-pattern.6234567892
port 1/0/1
prefix 6234567892
!
dial-peer voice 102 pots
preference 8
service dsapp
destination-pattern.6234567893
port 1/0/2
prefix 6234567893
!
!
dial-peer hunt 2
!
sip-ua
registrar ipv4:10.1.1.1 expires 3600
!
```

Additional References

The following sections provide references related to the SIP Support for Hookflash feature.

MIBs

MIB	MIBs Link
None	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

Technical Assistance

Description	Link
The Cisco Technical Support website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

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Configuring SIP ISDN Features

This chapter discusses the following SIP features that support ISDN:

- ISDN Calling Name Display
- Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks
- SIP Carrier Identification Code (CIC)
- SIP: CLI for Caller ID When Privacy Exists
- SIP: ISDN Suspend/Resume Support
- SIP PSTN Transport Using the Cisco Generic Transparency Descriptor (GTD)

Feature History for ISDN Calling Name Display

Release	Modification
12.3(4)T	This feature was introduced.

Feature History for Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks

Release	Modification
12.3(7)T	This feature was introduced.

Feature History for SIP Carrier Identification Code

Release	Modification
12.2(11)T	This feature was introduced.

Feature History for SIP: CLI for Caller ID When Privacy Exists Feature

Release	Modification
12.4(4)T	This feature was introduced.

Feature History for SIP: ISDN Suspend/Resume Support

Release	Modification
12.2(15)T	This feature was introduced.



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Feature History for SIP PSTN Transport Using the Cisco Generic Transparency Descriptor

Release	Modification
12.3(1)	This feature was introduced.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Prerequisites for SIP ISDN Support, page 2](#)
- [Restrictions for SIP ISDN Support, page 3](#)
- [Information About SIP ISDN Support, page 4](#)
- [How to Configure SIP ISDN Support Features, page 15](#)
- [Configuration Examples for SIP ISDN Support Features, page 31](#)
- [Additional References, page 51](#)

Prerequisites for SIP ISDN Support

ISDN Calling Name Display Feature

- Configure Generic Transparency Descriptor (GTD) on your SIP network.



Note For information on SIP support for communicating ISDN information using GTD bodies, see the “[SIP PSTN Transport Using the Cisco Generic Transparency Descriptor](#)” section on [page 12](#).

- Enable the Remote-Party-ID header on your SIP network. In general, Remote-Party-ID is enabled by default and no configuration is necessary. The Remote-Party-ID header provides translation capabilities for ISDN screening and presentation indicators in call setup messages.



Note For information on the Remote-Party-ID header, see the “[SIP Extensions for Caller Identity and Privacy](#)” section on [page 13](#).

- Use this feature in a uni-directional deployment beginning with an originating gateway. For example, the flow must be from a gateway to a phone or gateway to an application server.

Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks Feature

- Configure the SIP protocol.

SIP: CLI for Caller ID When Privacy Exists

- Establish a working IP network.

- Configure VoIP.
- Ensure that the gateway has voice functionality configured for SIP.



Note For information about configuring voice functionality, see the [Cisco IOS Voice Configuration Library](#).

SIP ISDN Suspend/Resume Support Feature

- Configure ISDN switch types on the gateway to support Suspend and Resume messages.

SIP PSTN Transport Using the Cisco Generic Transparency Descriptor Feature

- Configure your VoIP network, including the following components:
 - Cisco PGW 2200 signaling controller (SC) in Cisco MGC Software Release 9.2(2)



Note The Cisco PGW 2200 SC is formerly known as the Cisco Media Gateway Controller (MGC) and the Cisco SC 2200 signaling controller.

- Cisco Signaling Link Terminal (Cisco SLT), which performs Signaling System 7 (SS7) signal preprocessing for a Cisco PGW 2200 SC
- Cisco IOS gateways to allow sending and processing of SS7 ISUP messages in GTD format: Cisco IOS Release 12.3(1)
- Cisco SS7 Interconnect for Voice Gateways solution

Restrictions for SIP ISDN Support

SIP Carrier Identification Code Feature

- SIP gateways receive the CIC parameter in SIP INVITE or 302 REDIRECT messages only.
- SIP gateways do not add or configure CIC parameters.
- The TNS IE in the ISDN SETUP message does not map to the CIC parameter in a SIP INVITE request. It is only the CIC parameter that maps to the TNS IE in the outgoing ISDN SETUP message.



Note The workaround created in Cisco IOS Release 12.3(2)XB is no longer supported with the release of this feature. The workaround handled the CIC parameter by including it in the called-party number. The To header in the SIP INVITE message that contained the called-party number was prefixed with 101xxx, where xxx was the CIC parameter. The number was then sent to the ISDN in the SETUP message. When the ISDN received the number, for example, 101032119193921234

the ISDN ignored the 101 and then routed the call to carrier 0321, as if 0321 was in the TNS IE of the outgoing SETUP message. The rest of the number, formatted as the called-party number, was forwarded to the carrier.

- Support for the CIC parameter is addressed by the expired IETF draft-yu-tel-url-02.txt. The SIP Carrier Identification Code feature does not encompass all areas that are addressed in the draft.

SIP: ISDN Suspend/Resume Support Feature

- SIP ISDN Suspend/Resume support is available only for ISDN PRI trunks connected at the gateway.

SIP PSTN Transport Using the Cisco Generic Transparency Descriptor Feature

- Redundant Link Manager (RLM) is a requirement for the SIP PSTN Transport Using the Cisco Generic Transparency Descriptor feature. As a result, only the following platforms that use RLM are supported: Cisco AS5300, Cisco AS5350, and Cisco AS5400.



Note For information on RLM, see [Redundant Link Manager \(RLM\)](#).

- SIP-T also transparently transmits ISUP messages across a SIP network, but the process is not supported in this feature.

Information About SIP ISDN Support

To configure SIP ISDN support features, you should understand the following concepts:

- [ISDN Calling Name Display, page 4](#)
- [Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks, page 6](#)
- [SIP Carrier Identification Code, page 7](#)
- [SIP: CLI for Caller ID When Privacy Exists, page 7](#)
- [SIP: ISDN Suspend/Resume Support, page 11](#)
- [SIP PSTN Transport Using the Cisco Generic Transparency Descriptor, page 12](#)

ISDN Calling Name Display

With releases earlier than Cisco IOS Release 12.2(15)ZJ, when a call came in from the ISDN network to a SIP gateway, the calling name as presented in ISDN Q.931 messages (Setup and/or Facility) was not transported end-to-end over the VoIP cloud to a SIP endpoint (a SIP IP phone). With this feature, SIP signaling on Cisco IOS gateways has been enhanced to update the calling name and number information in SIP headers as per the recommended SIP standards. Also included is the complete translation of ISDN screening and presentation indicators, allowing SIP customers basic caller ID privileges.

Caller ID in ISDN Networks

In ISDN networks, caller ID (sometimes called CLID or ICLID for incoming calling line identification) is an analog service offered by a central office (CO) to supply calling party information to subscribers. Caller ID allows the calling party number and name to appear on a device such as a telephone display.

ISDN messages signal call control and are composed of information elements (IEs) that specify screening and presentation indicators. ISDN messages and their IEs are passed in GTD format. GTD format enables transport of signaling data in a standard format across network components and applications. The standard format enables other devices to scan and interpret the data. The SIP network extracts the calling name from the GTD format and sends the calling name information to the SIP customer.

ISDN and SIP Call Flows Showing the Remote-Party-ID Header

Figure 86 shows the SIP gateway receiving an ISDN Setup message that contains a Display (or Facility) IE indicating the calling name. Receiving the message initiates call establishment.

The Remote-Party-ID header sent by the SIP gateway identifies the calling party and carries presentation and screening information. The Remote-Party-ID header, which can be modified, added, or removed as a call session is being established, enables call participant privacy indication, screening, and verification.

Figure 86 Calling Name in Display or Facility IE of an ISDN Setup Message

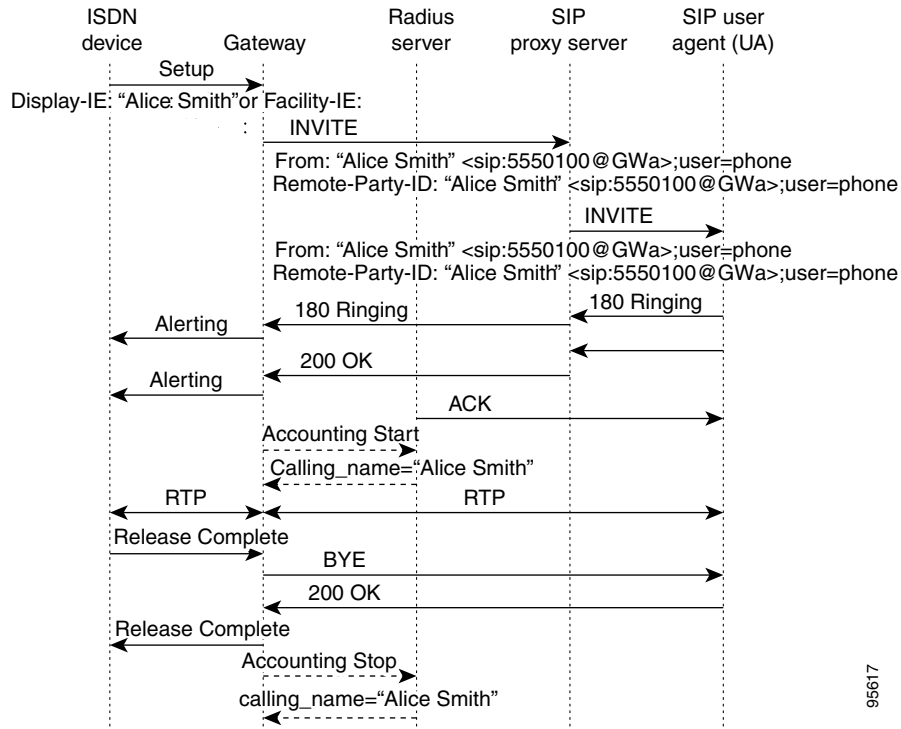
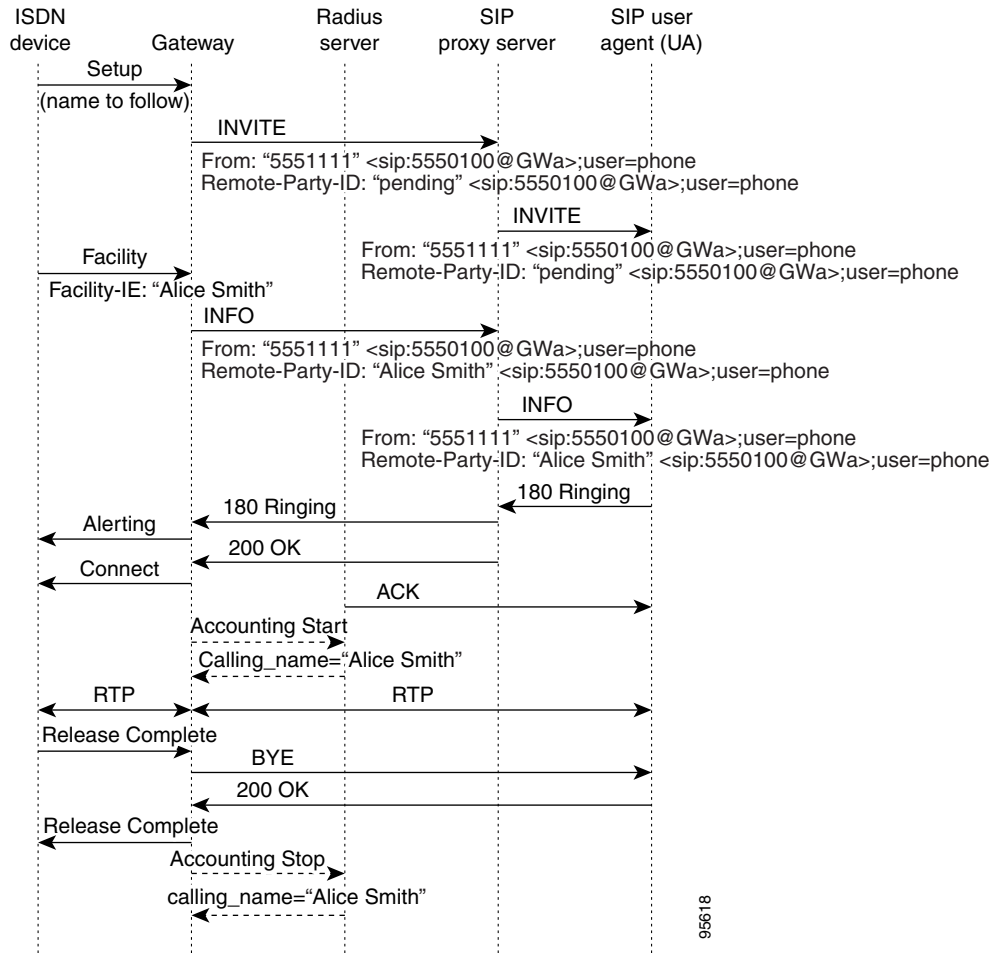


Figure 87 shows that the original ISDN Setup message sent by the ISDN device does not contain a Facility IE. The SIP gateway receives the ISDN Setup message indicating that the calling name is to be delivered in a subsequent ISDN Facility message. The SIP gateway then sets the display name of the Remote-Party-ID to *pending*. The presence of *pending* in a calling Remote-Party-ID of an INVITE denotes that the display name is to follow.

The functionality of a calling name sent in a subsequent message requires that:

- The ISDN switch type has the ability to indicate that the name follows in the next Facility message after the initial ISDN Setup message.
- The SIP gateway has the ability to interpret the subsequent Facility message into a SIP message. The SIP INFO message is used to interpret the Facility received from the ISDN device.

Figure 87 Calling Name in Facility IE of an ISDN Facility Message



Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks

The Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks feature enables call management applications to identify specific ISDN bearer (B) channels used during a voice gateway call for billing purposes. With the identification of the B channel, SIP gateways can enable port-specific features such as voice recording and call transfer.

In Cisco IOS releases prior to 12.3(7)T, fields used to store call leg information regarding the telephony port do not include B channel information. B channel information is used to describe incoming ISDN call legs. The Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks feature allows SIP and H.323 gateways to receive B-channel information from incoming ISDN calls. The acquired B channel information can be used during call transfer or to route a call.

SIP gateways use the **ds0-num** command to enable receiving the B channel of a telephony call leg. H.323 gateways use a different command, which allows users to run the two protocols on one gateway simultaneously.

**Note**

For information on using this feature on H.323 gateways, see [Configuring H.323 Gateways](#).

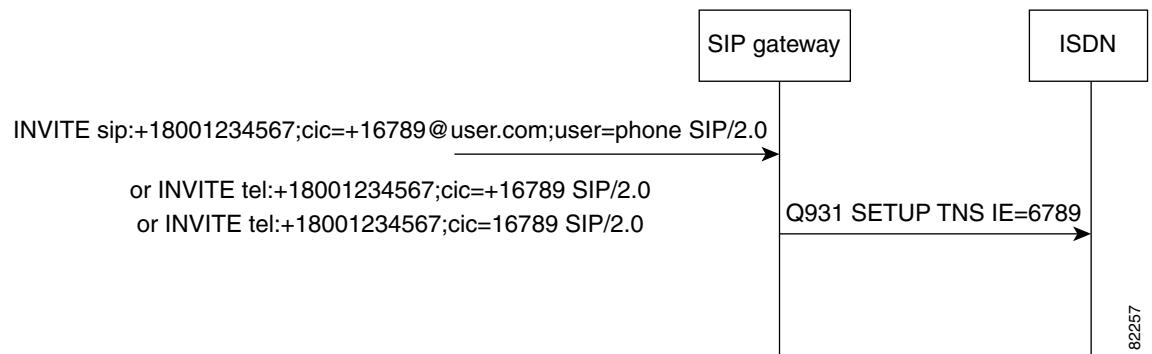
For SIP, if the **ds0-num** command is configured, the ISDN B-channel information is carried in the Via header of outgoing SIP requests.

SIP Carrier Identification Code

SIP gateways can receive and transmit the carrier identification code (CIC) parameter, allowing equal access support over many different networks. CIC enables transmission of the CIC parameter from the SIP network to the ISDN.

The CIC parameter is used in routing tables to identify the network that serves the remote user when a call is routed over many different networks. The parameter is carried in SIP INVITE requests and 302 REDIRECTs, and maps to the ISDN Transit Network Selection Information Element (TNS IE) in the outgoing ISDN SETUP message (see [Figure 88](#)). The TNS IE identifies the requested transportation networks and allows different providers equal access support based on customer choice.

Figure 88 Path of INVITE request with CIC Parameter to SIP Gateway Receiving and to ISDN



The CIC parameter is supported in SIP URLs, which identify a user's address and appear similar to e-mail addresses: *user@host*. It is also supported in the telephone-subscriber part of a TEL URL, which takes the basic form of *tel:telephone subscriber number*, where *tel* requests the local entity to place a voice call, and *telephone subscriber number* is the number to receive the call.

The CIC parameter can be a three-digit or a four-digit code. However, if it is a three-digit code, it is prefixed by a zero as in the following example:

```
cic=+1234 = TNS IE 0234.
```

SIP: CLI for Caller ID When Privacy Exists

The SIP: CLI for Caller ID When Privacy Exists feature is comprised of three main components, as follows:

- [SIP: Caller ID Removable to Improve Privacy, page 8](#)
- [SIP: Calling Number Substitution for the Display Name When the Display Name is Unavailable, page 9](#)
- [SIP: Calling Number Passing as Network-Provided or User-Provided, page 10](#)

SIP: Caller ID Removable to Improve Privacy

The caller ID information is passed through from the ISDN-to-SIP by copying the number in the Calling Party Number information element (IE) in an ISDN Setup message into the Calling Number field of the SIP Remote-Party-ID and From headers.

The Calling Name from the ISDN Display IE is copied into the SIP Display Name field in the SIP Remote-Party-ID and From headers. The Calling Party Number IE contains a Presentation Indicator field that is set to presentation allowed, presentation restricted, number not available due to interworking, or reserved. Presentation allowed and presentation restricted are translated into privacy set to off or privacy set to null, respectively, in the SIP Remote-Party-ID header field.

However, for added privacy, the SIP: CLI for Caller ID When Privacy Exists feature introduces CLI to completely remove the Calling Number and Display Name from an outgoing message's From header if presentation is prohibited. This prohibits sending the SIP Remote Party ID header, because the Cisco gateway does not send SIP Remote-Party ID headers without both a Display Name and Calling Number.



Note

The [SIP: Caller ID Removable to Improve Privacy](#) option is available both globally and at the dial-peer level.

See [Figure 89](#) for call flows and [Table 53](#) and [Table 54](#) for additional presentation mapping.

Figure 89 Call Flow for Blocking Caller ID Information When Privacy Exists

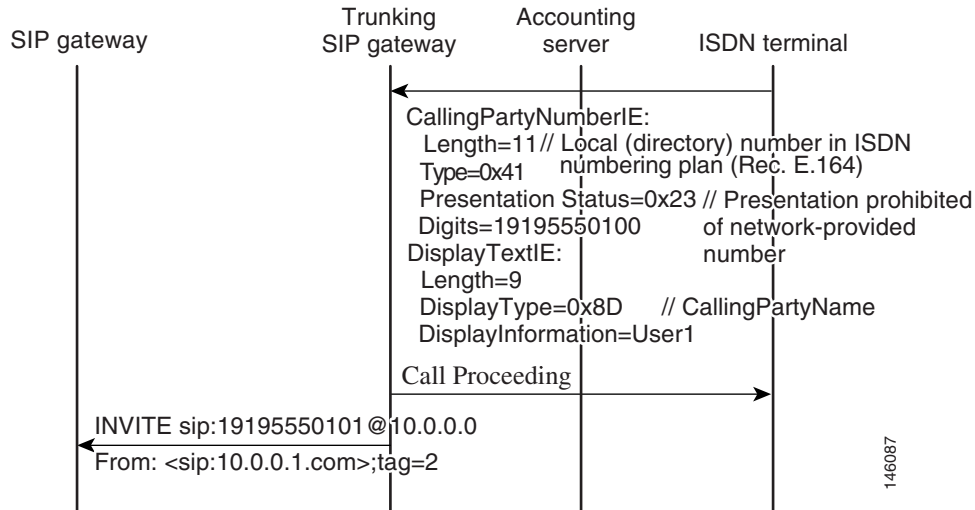


Table 53 *Presentation to Privacy Mapping with CLI Disabled*

Presentation Indicator	From Remote Party ID (RPID)
Presentation Allowed	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1" <sip:19195550100@10.0.0.0>;party=calling;privacy=off
Presentation Prohibited	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1"<sip:19195550100@10.0.0.0>;party=calling;privacy=full

Table 54 *Presentation to Privacy Mapping with CLI Enabled*

Presentation Indicator	From RPID
Presentation Allowed	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1"<sip:19195550100@10.0.0.0>;party=calling;privacy=off
Presentation Prohibited	From: <sip:10.0.0.0>;tag=1 Remote Party ID not sent

SIP: Calling Number Substitution for the Display Name When the Display Name is Unavailable

When the Display information element (IE) in a PSTN-to-SIP call is not available with a Setup message, the Cisco gateway leaves the Display Name field in the SIP Remote-Party-ID and From headers blank.

When presentation is allowed, the SIP: CLI for Caller ID When Privacy Exists feature enables the substitution of the Calling Number for the missing Display Name in the SIP Remote-Party-ID and From headers. Upon receipt of a Setup message where a name to follow is indicated, the Calling Number is not copied into the Display Name.

Also, the SIP Extensions for Caller Identity and Privacy on SIP gateway feature added the ability to hardcode calling name and number in the SIP Remote-Party-ID and From headers. The SIP Extensions for Caller Identify and Privacy feature settings take precedence over the SIP: CLI for Caller ID When Privacy Exists feature settings.

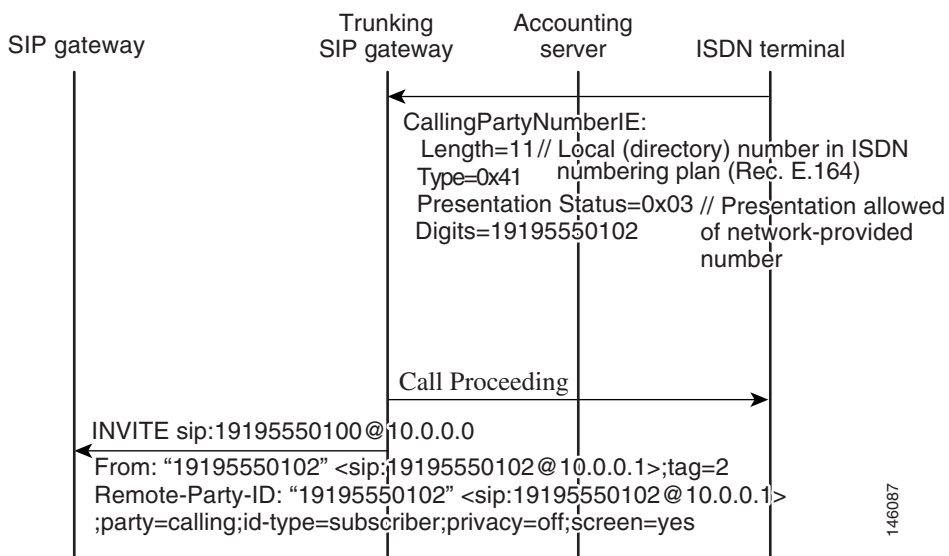


Note

The [SIP: Calling Number Substitution for the Display Name When the Display Name is Unavailable](#) option is available both globally and at the dial-peer level.

See [Figure 90](#) for the call flow where the Calling Number is substituted for the Display Number.

Figure 90 Call Flow for Substituting the Calling Number for the Display Name When the Display Name is Unavailable



SIP: Calling Number Passing as Network-Provided or User-Provided

ISDN numbers can be passed along as network-provided or user-provided in an ISDN Calling Party information element (IE) Screening Indicator field. The Cisco gateway automatically sets the Screening Indicator to user-provided in SIP-to-ISDN calls.

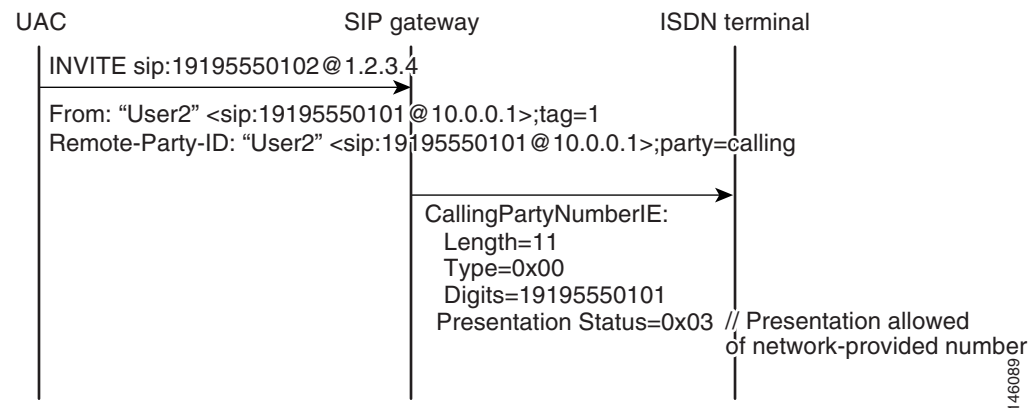
The SIP: CLI for Caller ID When Privacy Exists feature allows toggling between user-provided and network-provided ISDN numbers for the screening indicator. Therefore, after bits 1 and 2 are set to reflect network-provided, any existing screening information is lost. However, presentation information in bits 6 and 7 is preserved.



Note

The [Call Flow for Passing Through the Calling Number as Network-Provided](#) option is available both globally and at the dial-peer level.

See [Figure 91](#) for the call flow when the calling number is passed along as network-provided.

Figure 91 Call Flow for Passing Through the Calling Number as Network-Provided

SIP: ISDN Suspend/Resume Support

Suspend and Resume are basic functions of ISDN and ISDN User Part (ISUP) signaling procedures and now are a part of SIP functionality. Suspend is described in ITU Q.764 as a message that indicates a temporary cessation of communication that does not release the call. A Suspend message can be accepted during a conversation. A Resume message is received after a Suspend message and is described in ITU Q.764 as a message that indicates a request to recommence communication. If the calling party requests to release the call, the Suspend and Resume sequence is overridden.

SIP Call-Hold Process

When a SIP originating gateway receives an ISDN Suspend message, the originating gateway informs the terminating gateway that there is a temporary cessation of media; that is, the call is placed on hold. There are two ways that SIP gateways receive notice of a call hold. The first way is for the originating gateway to use a connection IP address of 0.0.0.0 (c=0.0.0.0) in the Session Description Protocol (SDP). The information in the SDP is sent in a re-Invite to the terminating gateway. The second way is for the originating gateway to use a=sendonly in the SDP of a re-Invite.



Note

Earlier than Cisco IOS Release 12.3(8)T, a SIP gateway could initiate call hold only by using c=0.0.0.0. As of Cisco IOS Release 12.3(8)T, a gateway can initiate call hold by using either c=0.0.0.0 or a=sendonly.

The purpose of the c=0.0.0.0 line is to notify the terminating gateway to stop sending media packets. When the hold is cancelled and communication is to resume, an ISDN Resume message is sent. The SIP originating gateway takes the call off hold by sending out a re-Invite with the actual IP address of the remote SIP entity in the c= line (in place of 0.0.0.0).

Multiple media fields (m-lines) in the SDP of a re-Invite message are used to indicate media forking, with each m-line representing one media destination. SIP gateways negotiate multiple media streams by using multiple m- and/or c-lines. When an originating gateway receives an ISDN Suspend on a gateway that has negotiated multiple media streams, all of the media streams are placed on hold. The originating gateway sends out a re-Invite that has a c= line that advertises the IP address as 0.0.0.0 on all streams.

The originating gateway also mutes the SIP calls for each media stream so that no media is sent to the terminating gateway. When the originating gateway receives an ISDN Resume, it initiates a re-Invite with the original SDP and takes the call off hold.

If the media inactivity timer is configured on the network, the timer is stopped for all active streams. The purpose of the media inactivity timer is to monitor and disconnect calls if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period. However, on initiating the call hold, the originating gateway disables the media inactivity timer for that particular call, so the call remains active. The terminating gateway behaves in the same way when it receives the call-hold re-Invite from the originating gateway. When the call resumes, the originating gateway re-enables the Media Inactivity Timer.

**Note**

For information on the timer, see the [“SIP Media Inactivity Timer” section on page 17](#).

All billing and accounting procedures are unaffected by the SIP: ISDN Suspend/Resume Support feature.

SIP PSTN Transport Using the Cisco Generic Transparency Descriptor

This section contains the following information:

- [SIP ISUP Transparency Using GTD Overview, page 12](#)
- [SIP INFO Message Generation and Serialization, page 13](#)
- [Transporting ISDN Messages in GTD Format, page 13](#)
- [SIP Generation of Multiple Message Bodies, page 14](#)
- [ISUP-to-SIP Message Mapping, page 14](#)

The SIP PSTN Transport Using the Cisco Generic Transparency Descriptor feature adds SIP support for ISDN User Part (ISUP) Transport using Generic Transparency Descriptor (GTD). The ISUP data received on the originating gateway (OGW) is preserved and passed in a common text format to the terminating gateway (TGW).

Feature benefits include the following:

- The ISUP data is reconstructed on the basis of the protocol at the egress side of the network, without any concern for the ISDN or ISUP variant on the ingress side of the network.
- By providing the ISDN or ISUP information in text format, the information can also be used by applications inside the core SIP network. An example of one such application is a route server that can use certain ISDN or ISUP information to make routing decisions.
- The transport of ISUP encapsulated in GTD maintains compatibility with the H.323 protocol.

SIP ISUP Transparency Using GTD Overview

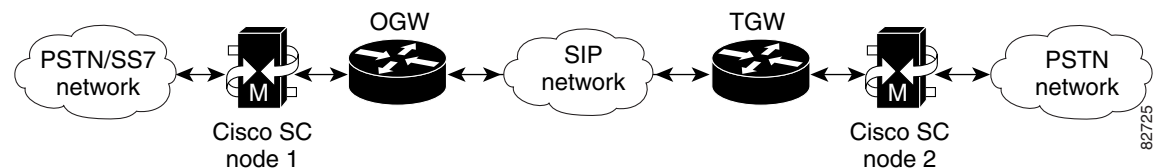
The SIP PSTN Transport Using the Cisco Generic Transparency Descriptor feature adds SIP support for ISUP transport using GTD. That is, ISUP data received on the OGW is preserved and presented in a common ASCII format to the TGW.

GTD objects can be used to represent ISUP messages, parameters, and R2 signals. These GTD objects are encapsulated into existing signaling protocols, such as SIP, facilitating end-to-end transport. The transport of ISUP encapsulated in GTD ASCII format already exists for H.323; SIP PSTN Transport

Using the Cisco Generic Transparency Descriptor provides feature parity. Using GTD as a transport mechanism for signaling data in Cisco IOS software provides a common format for sharing signaling data between various components in a network and for interworking various signaling protocols.

To attain ISUP transparency in VoIP Networks, the gateway needs to externally interface with the Cisco SC node. The Cisco SC node is the combination of hardware (Cisco PGW 2200 and Cisco SLTs) and signaling controller software that provides the signaling controller function. The Cisco SC node transports the signaling traffic between the SC hosts and the SS7 signaling network. A brief example of the process of an ISDN message containing an ISUP GTD message that comes into the Cisco OGW from a Cisco SC node is described below and shown in [Figure 92](#).

Figure 92 ISUP Transparency Implementation



The process in [Figure 92](#) is as follows:

1. Cisco SC node 1 receives an ISUP message from the public switched telephone network (PSTN). This node is now responsible for mapping the ISUP message into a GTD format and encapsulating this GTD body within the ISDN message that is sent to the OGW.
2. The SIP user agent on the OGW extracts the GTD body from the Q931 message and encapsulates it into a corresponding SIP message as a multipart MIME attachment.



Note For information on ISUP- to-SIP mapping, see [Table 55 on page 14](#).

3. The SIP message is sent by the OGW over the SIP network to the TGW.
4. The TGW encapsulates the GTD in the outgoing ISDN message which is sent to SC node 2. The SC then remaps the GTD to ISUP before passing it to the PSTN.

SIP INFO Message Generation and Serialization

The SIP PSTN Transport Using the Cisco Generic Transparency Descriptor (GTD) feature adds client and server support for the SIP INFO message in all phases of a call. INFO messages are used to carry ISUP messages that were encapsulated into GTD format, but that do not have a specific mapping to any SIP response or request. These ISUP messages can be received in any phase of the call.



Note For specific mapping messages, see the [“ISUP-to-SIP Message Mapping”](#) section on page 14.

The gateway does not support sending out overlapping SIP INFO messages. For example, a second INFO message cannot be sent out while one is still outstanding. Multiple PSTN messages that map to SIP INFO messages are sent out serially.

Transporting ISDN Messages in GTD Format

Support for ISDN messages in GTD format is limited to the ISDN Setup message. Only the following parameters are encoded and decoded:

- Originating-line information
- Bearer capability
- Calling-party number
- Called-party number
- Redirecting number

Whereas ISDN to GTD parameter mapping is enabled by default, you must configure the gateway to transport ISUP messages through SIP signaling.

The ISDN parameters can be transported using either GTD or SIP headers. Before the SIP PSTN Transport Using the Cisco Generic Transparency Descriptor feature, only SIP headers provided ISDN parameters. For instance, the user portion of a SIP From header can carry the ISDN Calling Party information element.

SIP headers generally contain the same information that is provided by GTD, because the headers are built on the OGW using information gained from the PSTN. However, there are situations in which the data may be in conflict. The inconsistent data occurs if the header was updated by an intermediate proxy or application server. In cases of conflict, the SIP header is used to construct the ISDN parameters on the TGW, because it generally contains the most recent information.

SIP Generation of Multiple Message Bodies

Before this feature, the SIP gateway handled only SDP as a message body type. With SIP PSTN Transport Using the Cisco Generic Transparency Descriptor, it is now possible for the gateway to generate and properly format messages that contain both SDP and GTD message body types.

Any SIP message that contains both SDP and GTD bodies may be large enough to require link-level fragmentation when User Datagram Protocol (UDP) transport is used, which could result in excessive retransmissions. TCP transport can be used if fragmentation becomes a performance issue.

ISUP-to-SIP Message Mapping

SIP PSTN Transport Using the Cisco Generic Transparency Descriptor attempts to map particular ISUP messages to an equivalent SIP message. This mapping is defined in [Table 55](#).

Table 55 Mapping of Supplemental ISUP Messages to SIP Messages

ISUP Message Type	ISDN (NI2C) Message Type	SIP Message Type
ACM	Alerting	180/183 Progress messages
ANM	Connect	200 OK to the INVITE request
CON	Connect	200 OK to the INVITE request
CPG	Progress	180/183 Progress messages
IAM	Setup	INVITE request
REL	Disconnect	BYE/CANCEL/4xx/5xx/6xx
RES	Resume	INVITE request
SUS	Suspend	INVITE

**Note**

There are many other PSTN or SS7 messages that are mapped into GTD formats within an ISDN message by the SC node. If the mapping is not listed in [Table 55](#), the message is treated with the SIP INFO method.

How to Configure SIP ISDN Support Features

This section contains the following procedures:

- [Configuring ISDN Calling Name Display](#), page 15
- [Configuring Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks](#), page 17
- [Configuring SIP Carrier Identification Code](#), page 18
- [Configuring SIP: CLI for Caller ID When Privacy Exists](#), page 19
- [Configuring SIP: ISDN Suspend/Resume Support](#), page 27
- [Configuring SIP PSTN Transport Using the Cisco Generic Transparency Descriptor](#), page 28
- [Verifying SIP ISDN Support Features](#), page 29
- [Troubleshooting Tips](#), page 30

**Note**

- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP ISDN Support”](#) section on page 2
 - [“Restrictions for SIP ISDN Support”](#) section on page 3
- For help with a procedure, see the troubleshooting section listed above.

Configuring ISDN Calling Name Display

To enable SIP IP phones to display caller-name identification for calls that originate on an ISDN network, perform the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **signaling forward**
5. **exit**
6. **interface serial**
7. **isdn supp-service name calling**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service configuration mode.
Step 4	signaling forward {none unconditional} Example: Router(conf-voi-serv)# signaling forward unconditional	Specifies whether or not the originating gateway (OGW) forwards the signaling payload to the terminating gateway (TGW). Keywords are as follows: <ul style="list-style-type: none"> • none—Prevent the gateway from passing the signaling payload to the TGW. • unconditional—Forward the signaling payload received in the OGW to the TGW, even if the attached external route server has modified the GTD payload.
Step 5	exit Example: Router(conf-voi-serv)# exit	Exits the current mode.
Step 6	interface serial slot/port:timeslot Example: Router(config)# interface serial 1/0:23	Specifies a serial interface created on a channelized E1 or channelized T1 controller. You must explicitly specify a serial interface. Arguments are as follows: <ul style="list-style-type: none"> • <i>slotport</i>—Slot and port where the channelized E1 or T1 controller is located. The slash is required. • <i>:time-slot</i>—For ISDN, the D-channel time slot, which is the 23 channel for channelized T1 and the 15 channel for channelized E1. The colon is required.
Step 7	isdn supp-service name calling Example: Router(config-if)# isdn supp-service name calling	Sets the calling-name display parameters sent out an ISDN serial interface.
Step 8	exit Example: Router(config-if)# exit	Exits the current mode.

Configuring Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks

To configure the Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks feature, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **ds0-num**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters VoIP voice-service configuration mode.
Step 4	sip Example: Router(conf-voi-serv) # sip	Enters SIP configuration mode.
Step 5	ds0-num Example: Router(conf-serv-sip) # ds0-num	Adds B-channel information to outgoing SIP messages.
Step 6	exit Example: Router(conf-serv-sip) # exit	Exits the current mode.

Configuring SIP Carrier Identification Code

SUMMARY STEPS

1. **debug ccsip messages**
2. **debug isdn q931**

DETAILED STEPS

Step 1 **debug ccsip messages**

Use this command to show all SIP SPI message tracing. Use it on a terminating gateway to verify the incoming CIC parameter.

Examples:

This example shows output of an INVITE request that uses a SIP URL and contains a CIC parameter:

```
Router# debug ccsip messages

00:03:01: Received:
INVITE sip:5550101;cic=+16789@172.18.202.60:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.202.62:5060
From: <sip:4440001@172.18.202.62>;tag=24176150-1A11
To: <sip:5550101@172.18.202.60;user=phone>
Date: Mon, 08 Mar 1993 00:11:51 GMT
Call-ID: 590F6480-1A7011CC-80B5CC57-1D726644@172.18.202.62
Supported: 100rel
Cisco-Guid: 1494180992-443552204-2159266903-494036548
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 731549511
Contact: <sip:4440001@172.18.202.62:5060;user=phone>
Expires: 180
Allow-Events: telephone-event, x-com-cisco-telephone-event,
x-com-cisco-fail-telephone-event
Content-Type: application/sdp
Content-Length: 160
```

The following shows output of an INVITE request that uses a TEL URL and contains a CIC parameter:

```
Router# debug ccsip messages

00:01:00: Received:
INVITE tel:+5550101;cic=+16789 SIP/2.0
Via: SIP/2.0/UDP 172.18.202.62:5060
From: <sip:4440001@172.18.202.62>;tag=24158B04-1D45
To: <sip:5550101@172.18.202.60;user=phone>
Date: Mon, 08 Mar 1993 00:09:51 GMT
Call-ID: 114C6D4C-1A7011CC-80B0CC57-1D726644@172.18.202.62
Supported: 100rel
Cisco-Guid: 290221388-443552204-2158939223-494036548
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 731549391
Contact: <sip:4440001@172.18.202.62:5060;user=phone>
Expires: 180
Allow-Events: telephone-event, x-com-cisco-telephone-event,
x-com-cisco-fail-telephone-event
```

```
Content-Type: application/sdp
Content-Length: 160
```

Step 2 **debug isdn q931**

Use this command to display information about call setup and teardown of ISDN network connections (layer 3) between the local router (user side) and the network. Use it to verify the contents of the CIC parameter and the TNS IE.

Example:

This example shows output of an outgoing call SETUP that contains the TNS IE. Output is the same for either a SIP or TEL URL.

```
Router# debug isdn q931
```

```
00:01:00: ISDN Se2/0:23: TX -> SETUP pd = 8 callref = 0x0001
00:01:00: Bearer Capability i = 0x8090A2
00:01:00: Channel ID i = 0xA98397
00:01:00: Calling Party Number i = 0x0081, '4440001', Plan:Unknown, Type:Unknown
00:01:00: Called Party Number i = 0xA8, '5550101', Plan:National, Type:National
00:01:00: Transit Net Select i = 0xA1, '6789'
```

Configuring SIP: CLI for Caller ID When Privacy Exists

This section contains the following procedures:

- [Configuring SIP: Blocking Caller ID Information Globally When Privacy Exists, page 19](#) (optional)
- [Configuring Dial-Peer Level SIP: Blocking of Caller ID Information When Privacy Exists, page 21](#) (optional)
- [Configuring Globally the SIP: Calling Number for Display Name Substitution When Display Name Is Unavailable, page 21](#) (optional)
- [Configuring Dial-Peer-Level SIP: Substitution of the Calling Number for Display Name When the Display Name Is Unavailable, page 22](#) (optional)
- [Configuring Globally the SIP: Pass-Through of the Passing Calling Number as Network-Provided, page 23](#) (optional)
- [Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as Network-Provided, page 24](#) (optional)
- [Configuring Globally the SIP: Pass-Through of the Passing Calling Number as User-Provided, page 25](#) (optional)
- [Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as User-Provided, page 26](#) (optional)

Configuring SIP: Blocking Caller ID Information Globally When Privacy Exists

The Call-ID information is private information. In ISDN there is a private setting that can be set to protect this information. However, whenever SIP gets the Call-ID information, it does not hide the private information, rather, it just sets a field to reflect that it is private and not to display it on a Call-ID display. But, the data is still viewable in the SIP message requests. This option allows the Cisco gateway to delete the Call-ID information from the SIP message requests so it cannot be read on the network.

Upon receiving an ISDN Setup message with the calling-party information element, the Cisco gateway translates the presentation indicator to set privacy to full for restricted presentation or to set privacy to off for unrestricted presentation in the Remote-Party-ID header field. The SIP: CLI for Caller ID When Privacy Exists feature introduces a CLI switch that either allows stripping the Calling Number and Display Name from the From and Remote-Party-ID fields in the SIP message requests or passes on the information. However, in cases of unrestricted presentation, the gateway passes the caller ID information, regardless of the CLI setting.

The global commands to strip the Calling Name and Calling Number from the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid strip pi-restrict all**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid strip pi-restrict all Example: Router(config-voip-serv)# clid strip pi-restrict all	Enters block call ID information when privacy exists in global configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring Dial-Peer Level SIP: Blocking of Caller ID Information When Privacy Exists

The dial-peer specific command to strip the Calling Number from the Remote-Party-ID and From headers is as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *dial-peer-number* voip**
4. **clid strip pi-restrict all**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid strip pi-restrict all Example: Router(config-dial-peer)# clid strip pi-restrict all	Enters block call ID information when privacy exists in dial-peer configuration mode.
Step 5	exit Example: Router# exit	Exits the current mode.

Configuring Globally the SIP: Calling Number for Display Name Substitution When Display Name Is Unavailable

When this is enabled, if there is no Display Name field but there is a number, it copies the number into the Display Name field, so the number is displayed on the recipient's Call-ID display.

The Cisco gateway omits the Display Name field if no display information is received. This feature also introduces a CLI switch that allows the Calling Number to be copied into the Display Name field, as long as presentation is not prohibited.

The steps for substituting the Calling Number for the Display Name when it is unavailable in the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid substitute name**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid substitute name Example: Router(config-voip-serv)# clid substitute name	Substitutes the calling number for the display name when the display name is unavailable in the global configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring Dial-Peer-Level SIP: Substitution of the Calling Number for Display Name When the Display Name Is Unavailable

The dial-peer-specific steps for substituting the Calling Number for the Display Name when it is unavailable in the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

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

3. **dial-peer voice *dial-peer-number* voip**
4. **clid substitute name**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config-dial-peer)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid substitute name Example: Router(config-dial-peer)# clid substitute name	Substitutes the calling number for the display name when the display name is unavailable in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring Globally the SIP: Pass-Through of the Passing Calling Number as Network-Provided

This field shows whether the Call-ID information was supplied by the network or not. This is for screening purposes.

Formerly the Calling Number from the session initiation protocol to public switched telephone network (SIP-to-PSTN) was always translated to user-provided. This feature introduces a CLI switch to toggle between branding numbers as user-provided or network-provided.

The steps for globally setting set the Screening Indicator to network-provided are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid network-provided**

5. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid network-provided Example: Router(config-voip-serv)# clid network-provided	Enters the network-provided calling number in voice-service-VoIP configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as Network-Provided

The dial-peer specific command to set the Screening Indicator to network-provided is as follows:

SUMMARY STEPS

1. enable
2. configure terminal
3. dial-peer voice *dial-peer-number* voip
4. clid network-provided
5. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid network-provided Example: Router(config-dial-peer)# clid network-provided	Enters the network-provided calling number in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring Globally the SIP: Pass-Through of the Passing Calling Number as User-Provided

The steps for globally setting set the Screening Indicator to user-provided are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **no clid network-provided**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	no clid network-provided Example: Router(config-voip-serv)# no clid network-provided	Enters the network-provided calling number in voice-service-VoIP configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as User-Provided

The dial-peer specific command to set the Screening Indicator to user-provided is as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *dial-peer-number* voip**
4. **no clid network-provided**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice dial-peer-number voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	no clid network-provided Example: Router(config-dial-peer)# no clid network-provided	Enters the user-provided calling number in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring SIP: ISDN Suspend/Resume Support

Suspend and Resume functionality is enabled by default. However, the functionality is also configurable. To configure Suspend and Resume for all dial peers on the VoIP network, perform the steps below on both originating and terminating gateways.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **suspend-resume**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	suspend-resume Example: Router(config-sip-ua)# suspend-resume	Enables support for Suspend and Resume.
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring SIP PSTN Transport Using the Cisco Generic Transparency Descriptor

To forward the GTD payload to the gateway either for all dial peers on the VoIP network or for individual dial peers, perform the following steps.

Prerequisites

- Configure the Cisco PGW2200 to encapsulate SS7 ISUP messages in GTD format before using the **signaling forward** command with the Cisco PGW 2200 signaling controller on the Cisco gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip** or **signaling forward**
4. **signaling forward**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<pre>enable</pre> <p>Example: Router> Enable</p>	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<pre>configure terminal</pre> <p>Example: Router# configure terminal</p>	Enters global configuration mode.
Step 3	<pre>voice service voip</pre> <p>Example: Router(config)# voice service voip</p> <p>OR</p> <pre>dial-peer voice tag {pots voip mmoip vofr voatm}</pre> <p>Example: Router(config)# dial-peer voice 100 voip</p>	<p>Enters one of the following configuration modes:</p> <ul style="list-style-type: none"> Voice-service configuration mode for all dial peers on the VoIP network Dial-peer voice configuration mode for an individual dial peer
Step 4	<pre>signaling forward {none unconditional}</pre> <p>Example: Router(conf-voi-serv)# signaling forward unconditional</p>	<p>Specifies whether or not the OGW forwards the signaling payload to the TGW. Keywords are as follows:</p> <ul style="list-style-type: none"> none—Prevent the gateway from passing the signaling payload to the TGW. unconditional—Forward the signaling payload received in the OGW to the TGW, even if the attached external route server has modified it. <p>Note The conditional keyword is not supported for SIP configuration. If you specify that keyword, the gateway treats it as if you had specified none.</p>
Step 5	<pre>exit</pre> <p>Example: Router(conf-voi-serv)# exit</p>	Exits the current mode.

Verifying SIP ISDN Support Features

To verify configuration of SIP ISDN support features, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. show dial-peer voice

2. **show running-config**
3. **show sip-ua status**

DETAILED STEPS

Step 1 **show running-config**

Use this command to display the configuration and verify that the correct dial peers were changed.

Step 2 **show dial-peer voice**

Use this command, for each dial peer configured, to verify that the dial-peer configuration is correct.

Step 3 **show sip-ua status**

Use this command to display whether Suspend and Resume support is enabled or disabled.

The following sample output shows that Suspend and Resume support is enabled.

```
Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Session name line (s=) required
  Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1
SIP support for ISDN SUSPEND/RESUME: ENABLED
```

Troubleshooting Tips



Note

For general troubleshooting tips and a list of important **debug** commands, see the [“General Troubleshooting Tips”](#) section on page 18.

- Make sure that you can make a voice call.
- Use the **debug ccsip messages** command as shown in the examples below.
- Use the **debug ccsip messages** command to enable traces for SIP messages, such as those that are exchanged between the SIP user-agent client (UAC) and the access server.
- Use the **debug isdn q931** command to display information about call setup and teardown of ISDN network connections (Layer 3) between the local router (user side) and the network.

Following is sample output for some of these commands:

- [Sample Output for the debug ccsip messages Command, page 31](#)

Sample Output for the debug ccsip messages Command

The following is a sample INVITE request with B-channel information added as an extension parameter “x-ds0num” to the Via header. The format of the B-channel billing information is: 0 is the D-channel ID, 0 is the T1 controller, and 1 is the B-channel.

```
Router# debug ccsip messages

INVITE sip:3100802@172.18.193.99:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.100:5060;x-ds0num="ISDN 0:D 0:DS1 1:DS0"
From: <sip:3100801@172.18.193.100>;tag=21AC4-594
To: <sip:3100802@172.18.193.99>
Date: Thu, 28 Dec 2000 16:15:28 GMT
Call-ID: 7876AC6C-DC1311D4-8005DBCA-A25DA994@172.18.193.100
Supported: 100rel
Cisco-Guid: 1981523172-3692237268-2147670986-2724047252
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO
CSeq: 101 INVITE
Max-Forwards: 6
Remote-Party-ID: <sip:3100801@172.18.193.100>;party=calling;screen=no;privacy=off
Timestamp: 978020128
Contact: <sip:3100801@172.18.193.100:5060>
Expires: 300
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 254
^M
v=0
o=CiscoSystemsSIP-GW-UserAgent 45 7604 IN IP4 172.18.193.100
s=SIP Call
c=IN IP4 172.18.193.100
t=0 0
m=audio 19492 RTP/AVP 18 0
c=IN IP4 172.18.193.100
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
```

The following sample INVITE request shows the Via header if the incoming trunk is T3. The format of the B-channel billing information is: 7/0 is the T3 controller, 1 is the T1 controller, and 2 is the B channel.

```
Router# debug ccsip messages

Via: SIP/2.0/UDP 172.18.193.120:5060; x-ds0num="ISDN 7/0:D 1:D1 2:DS0"
```

Configuration Examples for SIP ISDN Support Features

This section provides the following configuration examples:

- [ISDN Calling Name Display: Examples, page 32](#)
- [Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks: Example, page 35](#)
- [SIP Carrier Identification Code: Examples, page 36](#)
- [SIP: CLI for Caller ID When Privacy Exists: Examples, page 38](#)
- [SIP: ISDN Suspend/Resume Support: Example, page 47](#)
- [SIP PSTN Transport Using the Cisco Generic Transparency Descriptor: Examples, page 50](#)

ISDN Calling Name Display: Examples


Note

IP addresses and hostnames in examples are fictitious.

```

Router# show running-config

Building configuration...

Current configuration : 3845 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
!
boot-start-marker
boot-end-marker
!
no logging buffered
!
resource-pool disable
clock timezone GMT 5
clock summer-time GMT recurring
!

no aaa new-model
ip subnet-zero
ip tcp path-mtu-discovery
ip name-server 172.18.192.48
!
isdn switch-type primary-ni
isdn voice-call-failure 0
isdn alert-end-to-end
!
voice call send-alert
!
voice service voip
  signaling forward unconditional
  sip
!
fax interface-type fax-mail
!
controller T1 0
  framing esf
  crc-threshold 0
  clock source line primary
  linecode b8zs
  pri-group timeslots 1-24
  description lucent_pbx
!
controller T1 1
  shutdown
  framing esf
  crc-threshold 0
  linecode ami
  description summa_pbx
!
controller T1 2
  shutdown

```



```

framing esf
crc-threshold 0
linecode ami
!
controller T1 3
framing esf
crc-threshold 0
clock source line secondary 1
linecode b8zs
pri-group timeslots 1-24
!
translation-rule 100
Rule 1 ^1 1 ANY national
Rule 2 2% 2 ANY unknown
Rule 4 4% 4 ANY unknown
Rule 5 5% 5 ANY unknown
Rule 6 6% 6 ANY unknown
Rule 7 7% 7 ANY unknown
Rule 8 8% 8 ANY unknown
Rule 9 9% 9 ANY unknown
!
interface Ethernet0
ip address 172.18.193.100 255.255.255.0
no ip route-cache
no ip mroute-cache
ip rsvp bandwidth 1 1
!
interface Serial0:23
no ip address
isdn switch-type primary-ni
isdn incoming-voice modem
isdn guard-timer 3000
isdn supp-service name calling
isdn disconnect-cause 1
fair-queue 64 256 0
no cdp enable
!
interface Serial3:23
no ip address
isdn switch-type primary-ni
isdn protocol-emulate network
isdn incoming-voice modem
isdn guard-timer 3000
isdn supp-service name calling
isdn T310 30000
isdn disconnect-cause 1
isdn bchan-number-order descending
fair-queue 64 256 0
no cdp enable
!
interface FastEthernet0
ip address 10.1.1.2 255.255.255.0
no ip route-cache
no ip mroute-cache
duplex auto
speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.18.193.1
ip route 0.0.0.0 0.0.0.0 172.18.193.129
ip route 0.0.0.0 0.0.0.0 172.18.207.129
ip route 0.0.0.0 0.0.0.0 172.18.16.129
ip route 0.0.0.0 0.0.0.0 Ethernet0
ip route 0.0.0.0 0.0.0.0 172.18.197.1

```

```
ip route 0.0.0.0 255.255.255.0 Ethernet0
ip route 10.2.0.1 255.255.255.255 172.18.16.135
ip route 172.18.0.0 255.255.0.0 Ethernet0
no ip http server
!
map-class dialer test
  dialer voice-call
dialer-list 1 protocol ip permit
!
control-plane
!
voice-port 0:D
!
dial-peer voice 10 pots
  application session.t.old
  destination-pattern 5550100
  prefix 5550100
!
dial-peer voice 4 voip
  application session
  destination-pattern 5550120
  session protocol sipv2
  session target ipv4:172.18.193.99
  incoming called-number 5550125
!
dial-peer voice 1 pots
  application session
  destination-pattern 5550125
  incoming called-number 5550155
  port 0:D
  prefix 95550125
!
dial-peer voice 18 voip
  application session
  destination-pattern 36601
  session protocol sipv2
  session target ipv4:172.18.193.187
  codec g711ulaw
!
dial-peer voice 25 voip
  destination-pattern 5550155
  session protocol sipv2
  session target ipv4:172.18.192.232
!
dial-peer voice 5678 pots
  destination-pattern 5678
  port 3:D
  prefix 5678
!
dial-peer voice 56781 voip
  incoming called-number 5678
!
sip-ua
!
line con 0
line aux 0
line vty 0 4
  password password1
  login
!
end
```

Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks: Example

```
Router# show running-config

Building configuration...

Current configuration : 3394 bytes
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
  h323
    billing b-channel
  sip
    ds0-num

ip dhcp pool vespa
  network 192.168.0.0 255.255.255.0
  option 150 ip 192.168.0.1
  default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
  codec preference 2 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
  ip address 10.8.17.22 255.255.0.0
  half-duplex
!
interface FastEthernet0/0
  ip address 192.168.0.1 255.255.255.0
  speed auto
  no cdp enable
  h323-gateway voip interface
  h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
  network 10.0.0.0
  network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
```

```

!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
dial-peer voice 2 pots
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.example.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
telephony-service
 max-dn 10
 max-conferences 4
!
ephone-dn 1
 number 4001
!
ephone-dn 2
 number 4002
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
line vty 5 15
 login
!
no scheduler allocate
end

```

SIP Carrier Identification Code: Examples

This section provides the following configuration examples:

- [CIC Parameter in SIP URL, page 37](#)
- [CIC Parameter in TEL URL, page 37](#)
- [CIC Parameter and Visual Separators, page 37](#)
- [Copying the CIC Parameter into the Resulting INVITE Request, page 37](#)

CIC Parameter in SIP URL

This configuration example shows support for the CIC parameter in the user information part of the SIP URL. A SIP URL identifies a user's address and appears similar to an e-mail address, such as *user@host*, where *user* is the telephone number and *host* is either a domain name or a numeric network address. For example, the request line of an outgoing INVITE request might appear as:

```
INVITE sip:+5550100;cic=+16789@example.com;user=phone SIP/2.0
```

Where *+5550100*; *cic=+16789* signifies the user information, *example.com* the domain name, and the *user=phone* parameter distinguishes that the user address is a telephone number rather than a username.

CIC Parameter in TEL URL

This configuration example shows support for the CIC parameter in the telephone-subscriber part of the TEL URL. A TEL URL takes the basic form of *tel:telephone subscriber number*, where *tel* requests the local entity to place a voice call, and *telephone subscriber number* is the number to receive the call. For example:

```
tel:+5550100;cic=+16789
```

The additional CIC parameter can be in any of the following three formats:

```
cic=+16789
cic=+1-6789
cic=6789
```

CIC Parameter and Visual Separators

This configuration example shows support for the CIC parameter in different formats —with and without visual separators. However, the CIC parameter usually has no visual separators. All of the following formats are accepted:

```
+12345
cic=+12345
cic=2345
```

Copying the CIC Parameter into the Resulting INVITE Request

This configuration example shows that the CIC parameter can be copied from the user information part of a 3xx Contact SIP URL into the resulting INVITE request.

For example, if a 302 REDIRECT response from a proxy appears like:

```
Contact: <sip:+5550100;cic=+16789@example.com;user=phone>
```

or like:

```
Contact: <sip:+5550100;cic=6789@example.com;user=phone>
```

The result is an INVITE request that sends the CIC with a +1 prefixed to it.

```
INVITE sip:+5550100;cic=+16789@example.com;user=phone SIP/2.0
```

SIP: CLI for Caller ID When Privacy Exists: Examples

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when enabled globally and disabled on the dial-peer level:

```
Router# show running-config

Building configuration...
Current configuration: 1234 bytes
!
version 12.4
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname pip
!
boot-start-marker
boot system tftp user1/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkIqPXzKq4uKme.LhzGut0
enable password password1
!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name example.sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature enabled globally
clid substitute name
clid strip pi-restrict all
clid network-provided
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
```

```
codec preference 11 gsmefr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
```

```

dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550100
session protocol sipv2
session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
dial-peer voice 5550199 voip <- SIP: CLI for Caller ID When Privacy Exists feature
disabled on dial-peer
destination-pattern 3100801
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2
destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550199 5550100
gateway
timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
line con 0

```



```

exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password password1
login
!
no process cpu extended
no process cpu autoprofile hog
ntp clock-period 17180176
ntp server 192.0.10.150 prefer
!
end

```

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when disabled globally and disabled on the dial-peer level:

Router# **show running-config**

```

Building configuration...
Current configuration: 1234 bytes
!
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname pip
!
boot-start-marker
boot system tftp user1/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkiqPXzKq4uKme.LhzGut0
enable password password1
!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name example.sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature disabled globally
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw

```

```
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
codec preference 11 gsmefr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
```

```
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550100
session protocol sipv2
session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
dial-peer voice 5550199 voip <- SIP: CLI for Caller ID When Privacy Exists feature
disabled on dial-peer
destination-pattern 5550199
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2
destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550101 5550198
gateway
```

```

timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password password1
login
!
no process cpu extended
no process cpu autoprofile hog
ntp clock-period 17180176
ntp server 192.0.10.150 prefer
!
end

```

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when disabled globally and enabled on the dial-peer level:

```
Router# show running-config
```

```

Building configuration...
Current configuration: 1234 bytes
!
version 12.4
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname pip
!
boot-start-marker
boot system tftp judyg/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkIqPXzKq4uKme.LhzGut0
enable password password1
!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name example.sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!

```

```
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature disabled
globally
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
codec preference 11 gsmfr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
```

```

!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550102
session protocol sipv2
session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
dial-peer voice 5550100 voip <- SIP: CLI for Caller ID When Privacy Exists feature enabled
on dial-peer
destination-pattern 5550100
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
clid strip pi-restrict all
clid network-provided
clid substitute name
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2

```

```

destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550101 5550198
gateway
timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password password1
login
!
no process cpu extended
no process cpu autopprofile hog
ntp clock-period 17180176
ntp server 192.0.10.150 prefer
!
end

```

SIP: ISDN Suspend/Resume Support: Example

The following example shows SIP Suspend and Resume disabled on the gateway (SIP Suspend and Resume is enabled by default on the gateway).



Note

IP addresses and hostnames in examples are fictitious.

```

Router# show running-config

Building configuration...

Current configuration : 3845 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
!
boot-start-marker
boot-end-marker
!
no logging buffered
!
resource-pool disable

```

```

clock timezone GMT 5
clock summer-time GMT recurring
!

no aaa new-model
ip subnet-zero
ip tcp path-mtu-discovery
ip name-server 172.18.192.48
!
isdn switch-type primary-ni
isdn voice-call-failure 0
isdn alert-end-to-end
!
voice call send-alert
!
voice service voip
  signaling forward unconditional
  sip
!
fax interface-type fax-mail
!
controller T1 0
  framing esf
  crc-threshold 0
  clock source line primary
  linecode b8zs
  pri-group timeslots 1-24
  description lucent_pbx
!
controller T1 1
  shutdown
  framing esf
  crc-threshold 0
  linecode ami
  description summa_pbx
!
controller T1 2
  shutdown
  framing esf
  crc-threshold 0
  linecode ami
!
controller T1 3
  framing esf
  crc-threshold 0
  clock source line secondary 1
  linecode b8zs
  pri-group timeslots 1-24
!
translation-rule 100
  Rule 1 ^1 1 ANY national
  Rule 2 2% 2 ANY unknown
  Rule 4 4% 4 ANY unknown
  Rule 5 5% 5 ANY unknown
  Rule 6 6% 6 ANY unknown
  Rule 7 7% 7 ANY unknown
  Rule 8 8% 8 ANY unknown
  Rule 9 9% 9 ANY unknown
!
interface Ethernet0
ip address 172.18.193.100 255.255.255.0
no ip route-cache
no ip mroute-cache

```



```

ip rsvp bandwidth 1 1
!
interface Serial0:23
no ip address
isdn switch-type primary-ni
isdn incoming-voice modem
isdn guard-timer 3000
isdn supp-service name calling
isdn disconnect-cause 1
fair-queue 64 256 0
no cdp enable
!
interface Serial13:23
no ip address
isdn switch-type primary-ni
isdn protocol-emulate network
isdn incoming-voice modem
isdn guard-timer 3000
isdn supp-service name calling
isdn T310 30000
isdn disconnect-cause 1
isdn bchan-number-order descending
fair-queue 64 256 0
no cdp enable
!
interface FastEthernet0
ip address 10.1.1.2 255.255.255.0
no ip route-cache
no ip mroute-cache
duplex auto
speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.18.193.1
ip route 0.0.0.0 0.0.0.0 172.18.193.129
ip route 0.0.0.0 0.0.0.0 172.18.207.129
ip route 0.0.0.0 0.0.0.0 172.18.16.129
ip route 0.0.0.0 0.0.0.0 Ethernet0
ip route 0.0.0.0 0.0.0.0 172.18.197.1
ip route 0.0.0.0 255.255.255.0 Ethernet0
ip route 10.2.0.1 255.255.255.255 172.18.16.135
ip route 172.18.0.0 255.255.0.0 Ethernet0
no ip http server
!
map-class dialer test
dialer voice-call
dialer-list 1 protocol ip permit
!
control-plane
!
voice-port 0:D
!
dial-peer voice 10 pots
application session.t.old
destination-pattern 5550100
prefix 5550100
!
dial-peer voice 4 voip
application session
destination-pattern 5550120
session protocol sipv2
session target ipv4:172.18.193.99
incoming called-number 5550125
!

```

```

dial-peer voice 1 pots
  application session
  destination-pattern 5550125
  incoming called-number 5550155
  port 0:D
  prefix 95550125
!
dial-peer voice 18 voip
  application session
  destination-pattern 36601
  session protocol sipv2
  session target ipv4:172.18.193.187
  codec g711ulaw
!
dial-peer voice 25 voip
  destination-pattern 5550155
  session protocol sipv2
  session target ipv4:172.18.192.232
!
dial-peer voice 5678 pots
  destination-pattern 5678
  port 3:D
  prefix 5678
!
dial-peer voice 56781 voip
  incoming called-number 5678
!
sip-ua
  no suspend-resume
  retry invite 1
  retry bye 1

line con 0
line aux 0
line vty 0 4
  password password1
  login
!
end

```

SIP PSTN Transport Using the Cisco Generic Transparency Descriptor: Examples

Configuring GTD Globally

The following examples shows that GTD is configured.

```
Router# show running-config
```

```
Building configuration...
```

```
Current configuration : 4192 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal

```

```
service udp-small-servers
!
hostname router
!
voice service voip
  signaling forward unconditional
  sip
.
```

Configuring GTD for an Individual Dial Peer

The following example shows GTD configured with unconditional forwarding on two dial peers:

```
Router# show running-config

Building configuration...

Current configuration : 4169 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
hostname router
.
.
.
dial-peer voice 36 voip
  incoming called-number 3100802
  destination-pattern 3100801
  signaling forward unconditional
  session protocol sipv2
  session target ipv4:192.0.2.209
!
dial-peer voice 5 voip
  destination-pattern 5555555
  signaling forward unconditional
  session protocol sipv2
  session target ipv4:172.18.192.218
.
.
.
```

Additional References

General SIP References

- [“SIP Features Roadmap” on page 1](#)—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- [“Basic SIP Configuration” on page 1](#)—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter (listed alphabetically)

- *Configuring H.323 Gateways* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgr/vvfax_c/callc_c/h323_c/323confg/3gwconf.htm
- *Redundant Link Manager (RLM)* at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_serv/as5400/sw_conf/ios_121/pull_rm.htm

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Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP Cisco Unified Border Element

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Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP Cisco Unified Border Element was introduced on Cisco IOS SIP gateways in phases. In the first phase, the Transparent Tunneling of QSIG over SIP TDM Gateway feature added the ability to transparently tunnel Q-signaling (QSIG) protocol ISDN messages across the Session Initiation Protocol (SIP) trunk. With this feature, QSIG messages (supplementary services carried within Q.931 FACILITY-based messages) can be passed end to end across a SIP network. However, in Cisco IOS Release 12.4(15)XY, deployment of this feature is limited to QSIG messages over SIP Time-Division Multiplexing (TDM) gateways. In later releases, the ISDN Q.931 Tunneling over SIP TDM Gateway feature adds support for transparent tunneling of all Q.931 messages over SIP and for the Transparent Tunneling of QSIG and Q.931 over a SIP-SIP Cisco Unified Border Element.

Transparent tunneling is accomplished by encapsulating QSIG or Q.931 messages within SIP message bodies. These messages are encapsulated using “application/qsig” or “application/x-q931” Multipurpose Internet Mail Extensions (MIME) to tunnel between SIP endpoints. Using MIME to tunnel through Cisco SIP messaging does not include any additional QSIG/Q.931 services to SIP interworking.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To reach links to specific feature documentation in this module and to see a list of the releases in which each feature is supported, use the [“Feature Information for Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP Cisco Unified Border Element”](#) section on page 13.

Finding Support Information for Platforms and Cisco IOS and Catalyst OS Software Images

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



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- [Information About Transparent Tunneling of QSIG or Q.931 over SIP, page 2](#)
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Prerequisites for Transparent Tunneling of QSIG or Q.931 over SIP

The Transparent Tunneling of QSIG over SIP TDM Gateway feature is intended for TDM PBX toll bypass and call center applications. In its first release (Cisco IOS Release 12.4(15)XY), only tunneling of QSIG messages is supported and only on TDM gateways. From Cisco IOS release 12.4(15)XZ and 12.4(20)T onward, support is added for the ISDN Q.931 Tunneling over SIP TDM Gateway and Transparent Tunneling of QSIG and Q.931 over SIP-SIP Cisco Unified Border Element.

Before configuring transparent tunneling of QSIG and Q.931 over a SIP trunk, verify the SIP configuration within the VoIP network for the appropriate originating and terminating gateways as described in documentation listed in the [“Related Documents” section on page 11](#).

Restrictions for Transparent Tunneling of QSIG or Q.931 over SIP

Transparent tunneling of QSIG or Q.931 does not function unless both the originating gateway (OGW) and the terminating gateway (TGW) are configured using the same ISDN switch type. Additionally, this function is supported only on SIP-to-SIP configurations on Cisco Unified Border Element. Tunneling of QSIG or Q.931 is not supported on SIP-to-H.323 or H.323-to-H.323 configurations on Cisco Unified Border Element.

Information About Transparent Tunneling of QSIG or Q.931 over SIP

To configure transparent tunneling of QSIG or Q.931 over SIP, you should understand the following concepts:

- [Use of the QSIG or Q.931 Protocols, page 2](#)
- [Purpose of Tunneling QSIG or Q.931 over SIP, page 3](#)
- [Encapsulation of QSIG in SIP Messaging, page 4](#)
- [Mapping of QSIG Message Elements to SIP Message Elements, page 5](#)

Use of the QSIG or Q.931 Protocols

Q-series documents, controlled by the International Telecommunication Union (ITU), define the network Layer. The Q.931 document defines the Layer 3 protocol that serves as the connection control protocol for ISDN signaling—it is used primarily to manage the initiation, maintenance, and termination of connections over a digital network.

The Q signaling (QSIG) protocol is based on the Q.931 standard and is used for ISDN communications in a Private Integrated Services Network (PISN). The QSIG protocol makes it possible to pass calls from one circuit switched network, such as a PBX or private integrated services network exchange (PINX), to another. QSIG messages are, essentially, a subset of Q.931 messages that ensure the essential Q.931 FACILITY-based functions successfully traverse the network regardless of the various hardware involved.

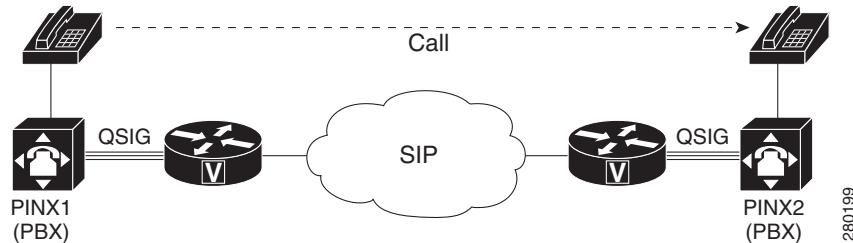
Q.931 tunneling over Cisco IOS SIP gateways was introduced as the ability to transparently tunnel only QSIG messages—the FACILITY-based Q.931 messages. Beginning with Cisco IOS Release 12.4(15)XZ and Cisco IOS Release 12.4(20)T, tunneling of all Q.931 messages (SETUP, ALERTING, CONNECT, and RELEASE COMPLETE messages in addition to FACILITY-based messages) is supported on Cisco IOS SIP gateways. However, for clarity, the descriptions and examples in this document focus primarily on QSIG messages.

Purpose of Tunneling QSIG or Q.931 over SIP

TDM Gateways

Transparently tunneling QSIG or Q.931 messages over SIP through SIP TDM gateways allows calls from one PINX to another to be passed through a SIP-based IP network with the equivalent functionality of passing through an H.323 network—without losing the functionality of the QSIG or Q.931 protocol to establish the call. To do this, QSIG or Q.931 messages are encapsulated within SIP messages (see [Figure 1](#)).

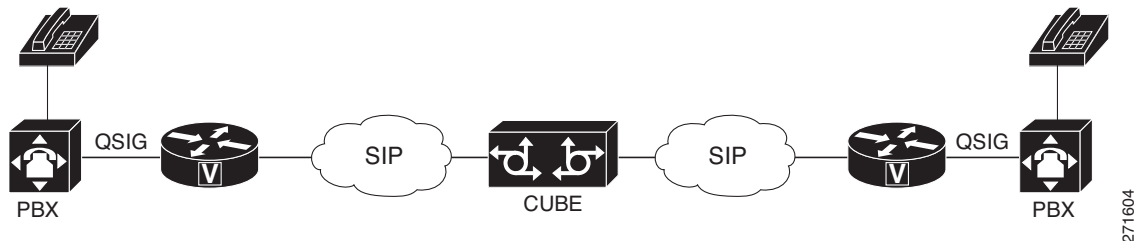
Figure 1 Tunneling QSIG (or Q.931) Messages Across a SIP Trunk



Cisco Unified Border Elements

Transparently tunneling QSIG or Q.931 over SIP through a Cisco Unified Border Element allows calls from one network to be passed through a SIP-to-SIP Cisco Unified Border Element connection to a bordering network (see [Figure 2](#)).

Figure 2 Tunneling QSIG (or Q.931) Messages Through a SIP-SIP Cisco Unified Border Element



Encapsulation of QSIG in SIP Messaging

QSIG messages are tunneled by encapsulating them as a MIME body in a SIP INVITE message on the OGW. Then, the MIME body is extracted from the SIP message by the TGW at the other end of the SIP network. To tunnel QSIG messages to a TGW on another network, configure and use a SIP-to-SIP Cisco Unified Border Element connection between each network over which the SIP INVITE must travel to reach the TGW. This tunneling process helps preserve all QSIG capabilities associated with a call or call-independent signal as it travels to its destination.

The following events make it possible to tunnel QSIG messaging across a SIP network:

- The ingress gateway (OGW) receives a QSIG call (or signal) establishment request (a SETUP message) and generates a corresponding SIP INVITE request.
- A corresponding SIP INVITE message is created and will contain the following:
 - A Request-URI—message part containing a destination derived from the called party number information element (IE) in the QSIG SETUP message. The destination can be the egress (TGW or the Cisco Unified Border Element) for exiting the SIP network or it can be the required destination, leaving SIP proxies to determine which gateway will be used.
 - A From header—message header containing a uniform resource identifier (URI) for either the OGW or calling party itself.
 - A Session Description Protocol (SDP) offer—a message part proposing two media streams, one for each direction.
 - A Multipart-MIME body—message part containing the tunneled QSIG data.
- In addition to normal user agent (UA) handling of a SIP response, the OGW performs a corresponding action when it receives a SIP response, as follows:
 - OGW receives 18x response with tunneled content—identifies the QSIG message (FACILITY, ALERTING, or PROGRESS) and sends a corresponding ISDN message.
 - OGW receives 3xx, 4xx, 5xx, or 6xx final response—attempts alternative action to route the initial QSIG message or clears the call or signal using an appropriate QSIG cause value (DISCONNECT, RELEASE, or RELEASE COMPLETE). When the OGW receives a valid encapsulated QSIG RELEASE COMPLETE message, the OGW should use the cause value included in that QSIG message to determine the cause value.



Note You should expect a SIP 415 final response message (Unsupported Media Type) if the user agent server (UAS) is unable to process tunneled QSIG or Q.931 messages.

- OGW receives a SIP 200 OK response—performs normal SIP processing, which includes sending an ACK message. Additionally, the OGW will encapsulate the QSIG message in the response to the PSTN side and will connect the QSIG user information channel to the appropriate media streams as called out in the SDP reply.



Note A nonzero port number for each media stream must be provided in a SIP 200 OK response to the OGW before the OGW receives the QSIG CONNECT message. Otherwise, the OGW will behave as if the QSIG T301 timer expired.

- The TGW sends and the OGW receives a 200 OK response—the OGW sends an ACK message to the TGW and all successive messages during the session are encapsulated into the body of SIP INFO request messages. There are two exceptions:
 - When a SIP connection requires an extended handshake process, renegotiation, or an update, the gateway may encapsulate a waiting QSIG message into a SIP re-INVITE or SIP UPDATE message during QSIG call establishment.

- When the session is terminated, gateways send a SIP BYE message. If the session is terminated by notice of a QSIG RELEASE COMPLETE message, that message can be encapsulated into the SIP BYE message.

Mapping of QSIG Message Elements to SIP Message Elements

This section lists QSIG message elements and their associated SIP message elements when QSIG messages are tunneled over a SIP trunk.

- | | | |
|-----------------------------|-----|---------------------------------|
| • QSIG FACILITY/NOTIFY/INFO | <—> | SIP INFO |
| • QSIG SETUP | <—> | SIP INVITE |
| • QSIG ALERTING | <—> | SIP 180 RINGING |
| • QSIG PROGRESS | <—> | SIP 183 PROGRESS |
| • QSIG CONNECT | <—> | SIP 200 OK |
| • QSIG DISCONNECT | <—> | SIP BYE/CANCEL/4xx—6xx Response |

How to Transparently Tunnel QSIG over SIP

To create a tunnel for QSIG messages across a SIP trunk, you must configure signaling forward settings on both the OGW and the TGW.

In the IP TDM gateway scenario, a gateway receives QSIG messages from PSTN and the ISDN module passes the raw QSIG message and, additionally, creates and includes a Generic Transparency Descriptor (GTD) that is passed with the raw QSIG message across the IP leg of the call.

In the SIP TDM gateway scenario, there are two options—raw message (rawmsg) and unconditional. The rawmsg option specifies tunneling of only raw message (application/qsig or application/x-q931). The unconditional option specifies tunneling of all additional message bodies, such as GTD and raw message (application/qsig or application/x-q931).

Use the **signaling forward** command at the global configuration level to configure the feature for the entire gateway. You can also enable the QSIG tunneling feature for only a specific interface. If you enable this feature at both the global and dial peer configuration level and the option specified for the interface is different than for the gateway, the interface setting will override the global setting. The processes for specifying either option at both levels are included in the following sections:

- [Configuring Signaling Forward Settings for a Gateway, page 5](#)
- [Configuring Signaling Forward Settings for an Interface, page 7](#)

Configuring Signaling Forward Settings for a Gateway

To create a tunnel for QSIG messages across a SIP trunk using the same signaling forward setting for all interfaces on a gateway, configure the signaling forward settings in voice service voip configuration mode.

Signaling Forward Settings for a Gateway

The two options—raw messages (rawmsg) and unconditional—are mutually exclusive, which means you can specify only one option at the global configuration level. To enable and specify the signaling forward option, use the **signaling forward** command in voice service voip configuration mode.



Note

To override the global setting for a specific interface, use the **signaling forward** command at the dial-peer level (see the [“Configuring Signaling Forward Settings for an Interface”](#) section on page 7).

Prerequisites

To create QSIG tunnels using the signaling forward configuration, configure both gateways. You can configure gateways globally or you can configure one or more interfaces on a gateway. In either case, you must include the recommended configuration for PRACK to avoid message/data loss.



Note

It is not necessary that both gateways are configured with the same signaling forward option but, if they are not, only raw QSIG messages can be tunneled. However, it is recommended that you tunnel QSIG messages with at least one interface configured on both gateways. If only one gateway is configured, QSIG tunneling might work in one direction but may not work properly in both directions.

You must also specify the central office switch type on the ISDN interface for both the OGW and the TGW. Use the **isdn switch-type** command in global or dial peer configuration mode to enable and specify the switch type for QSIG or Q.931 support (see the [“Related Documents”](#) section on page 11).

Furthermore, before the **isdn switch-type** setting can function properly, you must assign network-side functionality for the primary-qsig switch type (either at the global or dial-peer level) using the **isdn protocol-emulate** command (see the [“Related Documents”](#) section on page 11).

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **signaling forward** *option*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<code>voice service voip</code> Example: Router(config)# voice service voip	Enters voice-service configuration mode and specifies a voice-encapsulation type globally.
Step 4	<code>signaling forward message-type</code> Example: Router(conf-voi-serv)# signaling forward rawmsg or Router(conf-voi-serv)# signaling forward unconditional	Enables tunneling of QSIG raw messages (application-qsig) only. or Enables tunneling of all QSIG message bodies unconditionally.

Configuring Signaling Forward Settings for an Interface

To create a tunnel for QSIG messages across a SIP trunk on a specific interface on a gateway, configure the signaling forward settings in dial peer configuration mode.

Signaling Forward Settings for an Interface

The two options—raw messages (rawmsg) and unconditional—are mutually exclusive, which means you can specify only one option per interface at the dial-peer level. To enable and specify the signaling forward option for an interface, use the **signaling forward** command in dial peer configuration mode.



Note

To set the signaling forward option for an entire gateway, use the **signaling forward** command at the global level (see the [“Configuring Signaling Forward Settings for a Gateway”](#) section on page 5).

Prerequisites

To create QSIG tunnels using the signaling forward configuration, configure at least one interface on both gateways. You can also configure all interfaces at once by configuring the gateway globally. In either case, you must include the recommended configuration for PRACK to avoid data loss.



Note

It is not necessary that both gateways are configured with the same signaling forward option but, if they are not, only raw QSIG messages can be tunneled. However, it is recommended that you tunnel QSIG messages with at least one interface configured on both gateways. If only one gateway is configured, QSIG tunneling might work in one direction but may not work properly in both directions.

You must also specify the central office switch type on the ISDN interface for both the OGW and the TGW. Use the **isdn switch-type** command in global or dial peer configuration mode to enable and specify the switch type for QSIG or Q.931 support (see the [“Related Documents”](#) section on page 11).

Furthermore, before the **isdn switch-type** setting can function properly, you must assign network-side functionality for the primary-qsig switch type (either at the global or dial-peer level) using the **isdn protocol-emulate** command (see the [“Related Documents”](#) section on page 11).

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *number* voip**
4. **signaling forward *option***

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> voip Example: Router(config)# dial-peer voice 3 voip	Enters voice-service configuration mode and specifies a voice-encapsulation type for a specific interface.
Step 4	signaling forward <i>message-type</i> Example: Router(config-dial-peer)# signaling forward rawmsg or Router(config-dial-peer)# signaling forward unconditional	Enables tunneling of QSIG raw messages (application-qsig) only. or Enables tunneling of all QSIG message bodies unconditionally.

Configuration Examples for Transparent Tunneling of QSIG over SIP

This section provides the following configuration examples:

- Configuration at the global level:
 - [Tunneling QSIG Raw Messages over SIP on an OGW or TGW: Example, page 9](#)
 - [Tunneling QSIG Messages Unconditionally over SIP on an OGW or TGW: Example, page 9](#)
- Configuration at the dial peer (interface) level:
 - [!Tunneling QSIG Raw Messages over SIP on an OGW and TGW Interface: Example, page 9](#)
 - [!Tunneling QSIG Messages Unconditionally over SIP on an OGW or TGW Interface: Example, page 10](#)

Tunneling QSIG Raw Messages over SIP on an OGW or TGW: Example

The following example shows how to configure transparent tunneling of only QSIG raw messages (application-qsig) through a SIP TDM gateway on a SIP trunk at either the OGW or TGW:

```
!
voice service voip
  signaling forward rawmsg
  sip
    rel1xx require "100rel"
!
```

Tunneling QSIG Messages Unconditionally over SIP on an OGW or TGW: Example

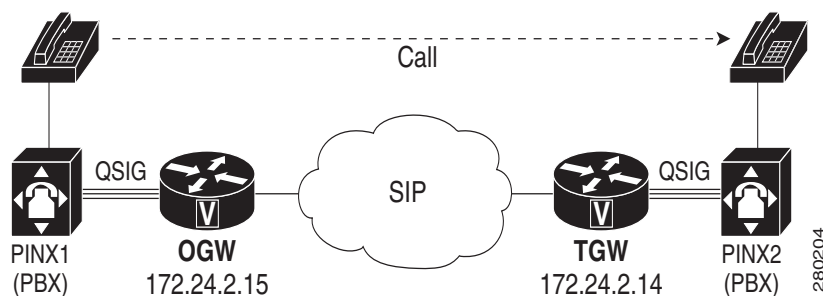
The following example shows how to configure transparent tunneling of QSIG messages unconditionally through a SIP TDM gateway on a SIP trunk at either the OGW or TGW:

```
!
voice service voip
  signaling forward unconditional
  sip
    rel1xx require "100rel"
!
```

Tunneling QSIG Raw Messages over SIP on an OGW and TGW Interface: Example

The following example shows how to configure transparent tunneling of only QSIG raw messages (application-qsig) on a gateway interface in a SIP network (see [Figure 3](#)):

Figure 3 Tunneling of Only QSIG Raw Messages over a SIP Trunk (Interface-Level)



Configuration for OGW (172.24.2.15) Tunneling only QSIG Raw Mmessages

```
!
dial-peer voice 7777 voip
description OGW-OUT-TGW
destination-pattern 222
signaling forward rawmsg
session protocol sipv2
session target ipv4:172.24.2.14
!
```

Configuration for TGW (172.24.2.14) Tunneling only QSIG Raw Mmessages

```

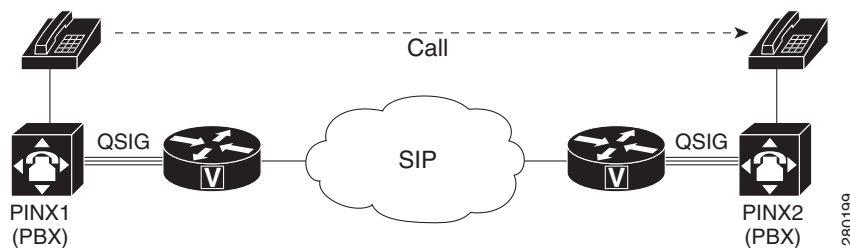
!
dial-peer voice 333 voip
description TGW_RSVP_IN-DP
session protocol sipv2
signaling forward rawmsg
incoming called-number 222
!

```

!Tunneling QSIG Messages Unconditionally over SIP on an OGW or TGW Interface: Example

The following example shows how to configure transparent tunneling of QSIG messages unconditionally over a gateway interface in a SIP network (see [Figure 4](#)):

Figure 4 Tunneling of QSIG Messages Unconditionally over a SIP Trunk (Interface-Level)

**Configuration for OGW (172.24.2.14) Tunneling QSIG Messages Unconditionally**

```

dial-peer voice 7777 voip
description OGW-OUT-TGW
destination-pattern 222
signaling forward unconditional
session protocol sipv2
session target ipv4:172.24.2.14

```

Configuration for TGW (172.24.2.15) Tunneling QSIG Messages Unconditionally

```

dial-peer voice 333 voip
description TGW-RSVP-IN-DP
session protocol sipv2
signaling forward unconditional
incoming called-number 222

```

Additional References

The following sections provide references related to the Transparent Tunneling of QSIG and Q.931 over SIP-TDM Gateway and SIP-SIP Cisco Unified Border Element features.

Related Documents

Related Topic	Document Title
Cisco IOS dial peer overview	Dial Peer Configuration on Voice Gateway Routers Configuration Guide
Cisco IOS dial technologies command information	Cisco IOS Dial Technologies Command Reference
Cisco IOS dial technologies configuration information	Cisco IOS Dial Technologies Configuration Guide
Cisco IOS SIP configuration information	Cisco IOS SIP Configuration Guide
Cisco IOS voice command information	Cisco IOS Voice Command Reference
Cisco IOS voice configuration information	Cisco IOS Voice Configuration Library
Cisco Unified CME command information	Cisco Unified Communications Manager Express Command Reference
Cisco Unified CME configuration information	Cisco Unified CME Support Documentation Home Page

Standards

Standard	Title
Standard ECMA-355	Corporate Telecommunication Networks - Tunnelling of QSIG over SIP

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download existing MIBs for selected platforms, Cisco IOS releases, and feature sets, use the Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 3204	MIME Media Types for ISUP and QSIG Objects
RFC 4497	Interworking Between the Session Initiation Protocol (SIP) and QSIG

Technical Assistance

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

Command Reference

The following commands are introduced or modified in the feature or features documented in this module. For information about these commands, see the *Cisco IOS Voice Command Reference* at http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html. For information about all Cisco IOS commands, use the Command Lookup Tool at <http://tools.cisco.com/Support/CLILookup> or the *Cisco IOS Master Command List, All Releases*, at http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all_book.html.

For information about commands specific to a dial peer associated with Cisco Unified CME, refer to the *Cisco Unified Communications Manager Express Command Reference* at http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_cr.html.

- **isdn global-disconnect**
- **signaling forward**
- **signaling forward (dial peer)**

Feature Information for Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP Cisco Unified Border Element

Table 1 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, refer to documentation listed in the “[Command Reference](#)” section on page 12.

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



Note

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

Table 1 Feature Information for Transparent Tunneling of QSIG over SIP

Feature Name	Releases	Feature Information
Transparent Tunneling of QSIG over SIP-TDM Gateway	12.4(15)XY 12.4(20)T	This feature provides transparent tunneling of ISDN communications that use the QSIG protocol across an IP network. The QSIG tunneling feature does not add any QSIG services to SIP interworking. Additionally, for Cisco IOS Release 12.4(15)XY, the QSIG tunneling feature targets only time-division multiplexing (TDM) SIP gateways. This feature uses no new or modified commands.
ISDN Q.931 Tunneling over SIP TDM Gateway	12.4(15)XZ 12.4(20)T	This feature expands transparent tunneling of QSIG messages to all other Q.931 messages (SETUP, ALERTING, CONNECT, and RELEASE COMPLETE). The QSIG and Q.931 tunneling feature does not add any QSIG or Q.931 services to SIP interworking.
Transparent Tunneling of QSIG and Q.931 over SIP-SIP Cisco Unified Border Element	12.4(15)XZ 12.4(20)T	This feature extends support of QSIG and Q.931 tunneling to the Cisco Unified Border Element.

Glossary

ISDN—Integrated Services Digital Network.

MIME—Multipurpose Internet Mail Extensions.

OGW—originating gateway (ingress gateway).

PBX—Private Branch Exchange.

PINX—private integrated services network exchange.

PISN—private integrated services network.

QSIG—Q Signaling protocol.

SDP—Session Description Protocol.

SIP—Session Initiation Protocol.

TDM—Time-Division Multiplexing.

TGW—terminating gateway (egress gateway).

URI—uniform resource identifier.

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Configuring SIP RSVP Features

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Cisco IOS software combines Session Initiation Protocol (SIP) with Resource Reservation Protocol (RSVP) to enhance RSVP Application ID support (RFC 2872) and RSVP precondition support (RFC 3312 and RFC 4032). The RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express (Cisco Unified CME) feature introduces application-specific reservations that enhance the granularity of local policy match criteria on Cisco IOS SIP devices. Additionally, this feature provides support for SIP audio RSVP preconditions for audio on both SIP time-division multiplexing (TDM) gateways and on SIP trunks for Skinny Client Control Protocol (SCCP) line-side Cisco Unified CME devices.

The RSVP Preconditions for Video Gateway feature expands existing support for SIP video calls on H.324-SIP video gateways to include H.320-SIP video gateways. Additionally, this feature adds support for SIP video RSVP preconditions for SIP video calls on both H.320-SIP and H.324-SIP video gateways.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the [“Feature Information for SIP RSVP Features” section on page 28](#).

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

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- [Restrictions for SIP RSVP Features, page 2](#)
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- [Glossary, page 30](#)



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Prerequisites for SIP RSVP Features

You must configure RSVP on one or more interfaces on at least two neighboring routers that share a link within the network before you can configure RSVP application ID support for quality of service (QoS) management.

SIP time-division multiplexing (TDM) gateways already support RSVP preconditions based on [RFC 3312](#) end-to-end preconditions for basic audio call and midcall bandwidth changes.

Restrictions for SIP RSVP Features

The following restrictions apply to RSVP features on SIP TDM gateways:

- Merging of global and interface-based local policies is not supported—you cannot match on multiple policies.
- RSVP is the only precondition-based QoS mechanism supported.
- Only end-to-end preconditions are supported and segmented preconditions are not supported.
- QoS strength must be the same (mandatory or optional) in both directions (SENDRECV).
- Midcall QoS negotiation does not use provisional responses as depicted in RFC examples.
- New session parameters (codec and remote address) are used immediately after OFFER/ANSWER during the midcall QoS negotiation even when strength is mandatory (see [RFC 3312](#), Section 6).
- RSVP initiation is supported only for single stream.

Additionally, RSVP support is not available for the following features and devices:

- SIP call forking.
- H.323 gateways.
- Video supplementary services.
- Video calls on Cisco Unified Border Elements.
- Video calls on Cisco Unified CME.
- Video calls connected to Cisco Unified Communications Manager or any third-party endpoints.
- H.320, H.323, and H.324 audio-only calls.
- Basic video call escalations and de-escalations.

Information About SIP RSVP Features

The information in this section focuses primarily on RSVP preconditions for SIP. It includes information about support for SIP RSVP audio requests issued from Cisco Unified CME and SIP TDM gateways and about support for SIP video calls on H.320-SIP and H.324-SIP video gateways. The RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express feature is available only on supported devices that are running Cisco IOS Release 12.4(22)T or later releases and the RSVP Preconditions for Video Gateway feature is available only on supported devices running Cisco IOS Release 12.4(24)T or later releases. For more information about the RSVP Applications ID feature, see the “[Configuring RSVP](#)” module in the subsections about RSVP in the “Signalling” part in the *Cisco IOS Quality of Service Solutions Configuration Guide*.

Before configuring SIP RSVP application ID support or SIP RSVP preconditions on a SIP TDM gateway or Cisco Unified CME, you should understand the following concepts:

- [RSVP Bandwidth Limits, page 3](#)
- [RSVP Preconditions, page 3](#)
- [Global and Per-Interface RSVP Policies, page 4](#)
- [RSVP Policy Applications, page 5](#)
- [Preemption and Defending Priorities, page 5](#)
- [Supported SIP RSVP Implementations and Functions, page 6](#)

RSVP Bandwidth Limits

Multiple applications, such as voice and video, need RSVP support. In Cisco IOS software, RSVP can process and accept requests by referring to multiple bandwidth pools that are based on application IDs and configured RSVP local policies. These pools specify which applications are allowed and how much bandwidth can be reserved for each until specified bandwidth limits are reached.

When there is limited or no available bandwidth remaining, RSVP rejects requests that are not configured and prioritized in the bandwidth pools. For example, if video calls are configured in a pool but voice calls are not, just a few video calls could prevent most, maybe all, voice calls from being established—the video calls require such a large amount of bandwidth that there may not be enough bandwidth remaining. Such behavior could prevent deployment of RSVP for multiple applications, not just voice, when video is one of the applications for which RSVP is required.

To prevent one application type from consuming all bandwidth, [RFC 2872, *Application and Sub Application Identity Policy Element for Use with RSVP*](#), allows for creation of separate bandwidth reservation pools. For example, an RSVP reservation pool can be created for voice traffic and another for video traffic so that reservations tagged with these application IDs can then be matched to the interface bandwidth pools using RSVP local policies. To limit bandwidth per application, though, you must configure a bandwidth limit for each application and configure each with a reservation flag that associates the application with the appropriate bandwidth limit.

RSVP Preconditions

Information about RSVP preconditions for SIP calls is provided in the following sections:

- [SIP Audio RSVP Preconditions, page 3](#)
- [SIP Video RSVP Preconditions, page 4](#)

SIP Audio RSVP Preconditions

The RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified CME feature enables configuration of RSVP as a precondition for establishment of SIP sessions initiated on a SIP gateway or a Cisco Unified CME SIP trunk. To enforce RSVP limitations, both endpoints must be configured to accept and support RSVP connections. Once configured, RSVP precondition support allows you to ensure support of RSVP application IDs on a SIP TDM gateway by requiring that the participant reserve network resources before continuing with the session.

RSVP support on Cisco IOS SIP gateways includes support for call hold, call forward, call transfer, and shared line features. This implementation uses a SIP header precondition option tag that enables synchronization of call control with the RSVP layer.

The RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified CME feature enhances RSVP support on Cisco IOS SIP gateways to handle the reINVITE and REFER/302-based supplementary services initiated by Cisco IOS SIP-TDM gateways.

SIP Video RSVP Preconditions

The RSVP Preconditions for Video Gateway feature expands existing support for SIP video calls on H.324-SIP video gateways to include H.320-SIP video gateways. Additionally, this feature adds support for SIP video RSVP preconditions for SIP video calls on both H.320-SIP and H.324-SIP video gateways. However, there are significant differences in the bandwidth reservation attributes for each of these gateways.

Bandwidth Reservations for H.320 Video Gateways

- An H.320 call can occupy up to 16 B channels (determined by the max-bandwidth configuration on the POTS dial peer.)
- During precondition negotiation, 64 KB/s plus 20 percent overhead is reserved for the audio stream and the configured max-bandwidth amount (plus 20 percent overhead) is reserved for the video stream.
- Once the call is established, if the negotiated audio codec requires less than 64 KB/s, excess audio bandwidth is released. Additionally, the required video bandwidth is determined (the configured max-bandwidth setting minus the audio bandwidth) and excess video bandwidth is released.

Bandwidth Reservations for H.324 Video Gateways

- An H.324 call occupies only one B channel.
- The audio bandwidth will be either 12.2 KB/s for the adaptive multirate (AMR) codec or 64 KB/s for the G.711 codec. The H.324 codec is always AMR so if the audio codec on the SIP side is G.711, then the digital signal processor (DSP) will perform transcoding between the two.
- The video bandwidth will always be 50 KB/s (64 KB/s minus 12.2 KB/s).
- As with H.320 calls, during precondition negotiation, 64 KB/s plus 20 percent overhead is reserved for the audio stream. But for H.324 calls, bandwidth reserved for video will always be 64 KB/s (plus 20 percent overhead).
- Once the call is established and the audio codec is negotiated, excess bandwidth is released, which results in 50 KB/s reserved bandwidth for video and 64 KB/s or 12.2 KB/s (depending on the negotiated codec) reserved for audio.

Global and Per-Interface RSVP Policies

You can configure RSVP policies globally and on a per-interface basis. You can also configure multiple global policies and multiple policies per interface.

Global RSVP policies restrict how much RSVP bandwidth a router uses regardless of the number of interfaces. You should configure a global policy if your router has CPU restrictions, one interface, or multiple interfaces that do not require different bandwidth limits.

Per-interface RSVP policies allow you to configure separate bandwidth pools with varying limits so that no one application, such as video, can consume all the RSVP bandwidth on a specified interface at the expense of other applications, such as voice, which would be dropped. You should configure a per-interface policy when you need greater control of the available bandwidth.

RSVP Policy Applications

RSVP searches for policies whenever an RSVP message is processed. The policy tells RSVP if any special handling is required for that message.

If your network configuration has global and per-interface RSVP policies, the per-interface policies are applied first meaning that RSVP looks for policy-match criteria in the order in which the policies were configured. RSVP searches for policy-match criteria in the following order:

- Nondefault interface policies
- Default interface policy
- Nondefault global policies
- Global default policy

If RSVP finds no policy-match criteria, it accepts all incoming messages. To change this decision from accept to reject, issue the **ip rsvp policy default-reject** command.

Preemption and Defending Priorities

Preemption happens when one reservation receives priority over another because there is insufficient bandwidth in an RSVP pool. General information about preemption behavior and configuration is provided in the following sections:

- [Preemption Behavior Overview, page 5](#)
- [Preemption Priority Signaling, page 6](#)
- [Preemption Behavior Configuration, page 6](#)

Preemption Behavior Overview

There are two types of RSVP bandwidth pools: local policy pools and interface pools. Local policies can be global or interface-specific. RSVP performs admission control against these pools when a RESV message arrives.

If an incoming reservation request matches an RSVP local policy with an RSVP bandwidth limit that has already been reached, RSVP tries to preempt lower-priority reservations that were admitted by that policy. If there are not enough lower-priority reservations that can be preempted to make room for the incoming higher priority request, then RSVP rejects it. If there are enough lower-priority reservations that can be preempted to make room for the new call, then RSVP continues the reservation process by next checking the interface bandwidth pool to determine if bandwidth is available on the interface.

If the interface bandwidth pool limit has been reached, then RSVP tries to preempt lower-priority reservations on that interface to accommodate the new reservation request. However, RSVP does not take into account which local policies admitted the reservations—if there is not enough bandwidth on the interface bandwidth pool that can be preempted to make room for the new call, RSVP rejects the new reservation even though the new reservation was able to obtain bandwidth from the local policy pool.

Preemption can also happen when you manually reconfigure an RSVP bandwidth pool of any type to a lower value such that the existing reservations using that pool no longer fit in the pool. Assigning preemption and defending priority values allows reservations to register with those values and preempt or avoid preemption when competing with other reservations for available bandwidth.

Preemption Priority Signaling

If a received RSVP PATH or RESV message does not contain preemption priorities, the **ip rsvp policy preempt** command is enabled globally, and the message matches a local policy that contains an **ip qos preempt-priority** command, then a POLICY_DATA object with a preemption priority element that contains the local policy's priorities is added to the message as part of the policy decision. These priorities are stored with the RSVP state in the router and forwarded to neighbors.

Preemption Behavior Configuration

The **ip rsvp policy preempt** command controls whether or not a router preempts any reservations when required. When you issue this command, a RESV message that subsequently arrives on an interface can preempt the bandwidth of one or more reservations on that interface if the assigned setup priority of the new reservation is higher than the assigned hold priorities of the installed reservations.

Additionally, you can use the **ip qos defending-priority** and **ip qos preempt-priority** commands in dial peer VoIP configuration mode to configure the RSVP defending and preemption priority values, respectively, for more specific configuration of QoS behavior.

Supported SIP RSVP Implementations and Functions

The Cisco IOS Audio RSVP Preconditions feature also adds RSVP support on the SIP trunk of SCCP line-side Cisco Unified CME devices. The following RSVP scenarios are supported on SIP-TDM gateways and on the SIP trunk of SCCP line-side Cisco Unified CME devices for audio calls:

- Basic call with SCCP-Cisco Unified CME and Cisco IOS SIP TDM gateways
- SCCP-Cisco Unified CME and Cisco IOS SIP-TDM gateway initiated call hold, call forward, and call transfer (blind and consult).

The feature also provides support for the functions described in the following sections:

- [Configurable RSVP Application ID, page 6](#)
- [Call Treatment Policies on Reservation Failures, page 6](#)
- [Configurable DSCP Values Based on No RSVP, RSVP Success, and RSVP Failure, page 7](#)

Configurable RSVP Application ID

Prior to Cisco IOS Release 12.4(22)T, Cisco IOS SIP implementations did not pass any RSVP application information to the QoS module. Since then, a command was added in dial peer VoIP configuration mode so that Cisco IOS devices running Cisco IOS release 12.4(22)T and later can be configured to pass RSVP application IDs to the QoS module while requesting RSVP for audio and video streams.

Call Treatment Policies on Reservation Failures

A reservation failure could happen during the initial RSVP establishment attempt, during subsequent RSVP rereservation attempts (such as for a session target change), or during the tear-down of an established RSVP session. Regardless at which of these points the failure occurs, RSVP failure policies are applied. For pre-alert calls, refer to the default policy described in [RFC 3312](#). For post-alert calls, locally configured RSVP failure policies can be applied using the **voice-class sip rsvp-fail-policy** command in dial peer VoIP configuration mode.

The strength of local RSVP failure policies can be configured as either “mandatory” or “optional” with the following options:

- Strength configured as optional—The failure policy specifies only the interval at which the RSVP reservation is retried (the interval setting can be configured as “infinity,” which specifies no retries).
- Strength configured as mandatory—The failure policy determines both the interval at which the RSVP reservation is retried (“infinite,” or no retries is an option) and the number of retries attempted before the call is disconnected. When appropriate, the number of retries for a failure policy with a mandatory strength setting can be configured as “infinity,” which specifies that the call is not to be disconnected upon failure.

Configurable DSCP Values Based on No RSVP, RSVP Success, and RSVP Failure

In releases prior to Cisco IOS Release 12.4(22)T, there existed a command (the **ip qos dscp** command in dial peer VoIP configuration mode) for assigning a different DSCP value for video packets according to three different RSVP scenarios: when RSVP is disabled, when it is successful, and when it fails. But only one DSCP value could be configured for audio packets regardless of the RSVP scenario.

In Cisco IOS Release 12.4(22)T and later releases, the **ip qos dscp** command includes a modification that makes it possible to configure different DSCP values for audio packets that are also specific to the three different RSVP scenarios (disabled, successful, and failed). With this command, a unique DSCP value can be configured and sent to the RTP library for each RSVP status for both video and audio packets according to the RSVP status.

How to Configure SIP RSVP Features

This section contains the following procedures:

- [Configuring SIP RSVP Application ID Support, page 7](#) (required)
- [Configuring SIP RSVP Bandwidth Reservation, page 11](#) (required)
- [Configuring SIP RSVP Preconditions, page 13](#) (required)

Configuring SIP RSVP Application ID Support

Perform the tasks in this section to configure SIP RSVP application IDs, defending priority settings, and, if needed, preemption priority settings:

- [Configuring Application Identities for SIP Audio RSVP Preconditions, page 8](#) (required)
- [Configuring Defending Priority for RSVP, page 9](#) (required)
- [Configuring Preemption Priority for RSVP, page 10](#) (optional)

Configuring Application Identities for SIP Audio RSVP Preconditions

Perform this task to configure application identities for specifying SIP audio RSVP preconditions. (This task does not apply to video calls. The default policy will apply for video.)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **ip qos policy-locator {video | voice} [app app-string] [guid guid-string] [sapp subapp-string] [ver version-string]**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 1 voip	Enters dial peer VoIP configuration mode from global configuration mode.
Step 4	ip qos policy-locator {video voice} [app app-string] [guid guid-string] [sapp subapp-string] [ver version-string] Example: Router(config-dial-peer)# ip qos policy-locator voice app CISCO guid aaa sapp bbb ver ccc	Configures application IDs.

Configuring Defending Priority for RSVP

Perform this task to configure the defending priority of RSVP reservations.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *tag* voip**
4. **ip qos defending-priority *defending-pri-value***

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>tag</i> voip Example: Router(config)# dial-peer voice 1 voip	Enters dial peer VoIP configuration mode from global configuration mode.
Step 4	ip qos defending-priority <i>defending-pri-value</i> Example: Router(config-dial-peer)# ip qos defending-priority 65	Configures the RSVP defending priority.

Configuring Preemption Priority for RSVP

Perform this task to configure the preemption priority of RSVP reservations.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *tag* voip**
4. **ip qos **preemption-priority** *preemption-pri-value***

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>tag</i> voip Example: Router(config)# dial-peer voice 1 voip	Enters dial peer VoIP configuration mode from global configuration mode.
Step 4	ip qos preemption-priority <i>preemption-pri-value</i> Example: Router(config-dial-peer)# ip qos preemption-priority 45	Configures the RSVP preemption policy and defending priority.

Configuring SIP RSVP Bandwidth Reservation

Perform the tasks in this section to configure RSVP bandwidth reservation settings for use with the SIP RSVP Preconditions feature:

- [Configuring RSVP Bandwidth Reservations on an Interface, page 11](#) (required)
- [Setting Bearer Capability for an H.320 Dial Peer, page 12](#) (optional)

Configuring RSVP Bandwidth Reservations on an Interface

Perform this task to configure RSVP bandwidth reservations on an interface.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** *type number*
4. **ip rsvp bandwidth** [*interface-kbps*] [*single-flow-kbps*] [**rdm** *kbps* {**subpool** *kbps* | **bc1** *subpool*} | **mam max-reservable-bw** *kbps* **bc0** *kbps* **bc1** *kbps*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	interface <i>type number</i> Example: Router(config)# interface FastEthernet 0/1	Specifies an interface and enters interface configuration mode.
Step 4	ip rsvp bandwidth [<i>interface-kbps</i>] [<i>single-flow-kbps</i>] [rdm <i>kbps</i> { subpool <i>kbps</i> bc1 <i>subpool</i> } mam max-reservable-bw <i>kbps</i> bc0 <i>kbps</i> bc1 <i>kbps</i>] Example: Router(config-if)# ip rsvp bandwidth 1158 100	Configures bandwidth reservations for RSVP on an interface.

Setting Bearer Capability for an H.320 Dial Peer

Perform this task to configure the bearer capability setting, which enables support of unrestricted digital media on an H.320 dial peer.


Note

This task is required for H.320-SIP calls. It is not required for H.324-SIP calls or for RSVP specifically but it must be configured before using RSVP preconditions on an H.320 dial peer.

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 1 voip	Enters dial peer VoIP configuration mode.
Step 4	voice-class sip calltype-video Example: Router(config-dial-peer)# voice-class sip calltype-video	Configures the bearer capability setting on an H.320 dial peer to support unrestricted digital media.

Configuring SIP RSVP Preconditions

Perform the tasks in this section to configure the SIP RSVP Preconditions feature on a SIP TDM gateway or Cisco Unified CME:

- [Configuring RSVP Failure Policies for SIP Audio RSVP Preconditions, page 13](#) (required for audio only)
- [Configuring DSCP Identity, page 15](#) (required)

Configuring RSVP Failure Policies for SIP Audio RSVP Preconditions

Perform this task to configure RSVP failure policies for SIP audio RSVP preconditions. (This task does not apply to video calls. Due to the bandwidth calculation algorithm, there is no RSVP failures post alert for video.)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** *type* [*number*]
4. **ip rsvp bandwidth** [*interface-kbps*] [*single-flow-kbps*] [**rdm** *kbps* {**subpool** *kbps* | **bc1** *subpool*} | **mam max-reservable-bw** *kbps* **bc0** *kbps* **bc1** *kbps*]
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **req-qos** {**best-effort** | **controlled-load** | **guaranteed-delay**} [{**audio** *bandwidth* | **video** *bandwidth*} **default** | **max** *bandwidth-value*]
8. **acc-qos** {**best-effort** | **controlled-load** | **guaranteed-delay**} [**audio** | **video**]
9. **voice-class sip rsvp-fail-policy** {**video** | **voice**} **post-alert** {**optional keep-alive** | **mandatory** {**keep-alive** | **disconnect** **retry** *retry-attempts*}} **interval** *retry-interval*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	interface <i>interface type</i> [<i>number</i>] Example: Router(config)# interface FastEthernet 0	Specifies an interface and enters interface configuration mode.

Step 4	<pre>ip rsvp bandwidth [interface-kbps] [<i>single-flow-kbps</i>] [rdm <i>kbps</i> {subpool <i>kbps</i> bc1 <i>subpool</i>} mam max-reservable-bw <i>kbps</i> bc0 <i>kbps</i> bc1 <i>kbps</i>]</pre> <p>Example: Router(config-if)# ip rsvp bandwidth 85 85</p>	Enables RSVP for IP on the specified interface.
Step 5	<pre>exit</pre> <p>Example: Router(config-if)# exit</p>	Exits interface configuration mode and returns to global configuration mode.
Step 6	<pre>dial-peer voice <i>tag</i> voip</pre> <p>Example: Router(config)# dial-peer voice 1 voip</p>	Enters dial peer VoIP configuration mode from global configuration mode.
Step 7	<pre>req-qos {best-effort controlled-load guaranteed-delay} [{audio <i>bandwidth</i> video <i>bandwidth</i>} default max <i>bandwidth-value</i>]</pre> <p>Example: Router(config)# req-qos controlled-load</p>	Specifies the desired quality of service to be used in reaching a specified VoIP dial peer.
Step 8	<pre>acc-qos {best-effort controlled-load guaranteed-delay} [audio video]</pre> <p>Example: Router(config)# acc-qos controlled-load</p>	Defines the acceptable QoS for any inbound and outbound call on a VoIP dial peer.
Step 9	<pre>voice-class sip rsvp-fail-policy {video voice} post-alert {optional keep-alive mandatory {keep-alive disconnect retry <i>retry-attempts</i>}} interval <i>retry-interval</i></pre> <p>Example: Router(config-dial-peer)# voice-class sip rsvp-fail-policy voice post-alert optional keep-alive interval 60</p>	Configures RSVP failure policies.

Configuring DSCP Identity

Perform this task to configure the DSCP identity.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **ip qos dscp {dscp-value | set-af | set-cs | default | ef} {signaling | media [rsvp-pass | rsvp-fail] | video [rsvp-none | rsvp-pass | rsvp-fail]}**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 1 voip	Enters dial peer VoIP configuration mode from global configuration mode.
Step 4	ip qos dscp {dscp-value set-af set-cs default ef} {signaling media [rsvp-pass rsvp-fail] video [rsvp-none rsvp-pass rsvp-fail]} Example: Router(config-dial-peer)# ip qos dscp 12 media rsvp-pass	Configures the DSCP identity.

Configuration Examples for SIP RSVP

This section provides the following configuration example:

- [SIP Audio RSVP Preconditions on a SIP Gateway: Example, page 16](#)
- [SIP Video RSVP Preconditions on an H.320-SIP Gateway: Example, page 18](#)
- [SIP Video RSVP Preconditions on an H.324-SIP Gateway: Example, page 19](#)
- [RSVP Preconditions Behavior Verification in SIP Calls: Example, page 22](#)

SIP Audio RSVP Preconditions on a SIP Gateway: Example

The following example shows how to configure SIP audio RSVP preconditions for audio calls on a SIP-TDM gateway:

```

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3845-RSVP-1
!
boot-start-marker
boot-end-marker
!
no aaa new-model
no network-clock-participate slot 2
!
ip cef
!
no ip domain lookup
multilink bundle-name authenticated
!
voice-card 0
no dspfarm
!
voice-card 2
no dspfarm
!
voice service voip
sip
!
voice class codec 1
codec preference 1 g711alaw
codec preference 2 g729r8
codec preference 3 g729br8
!
archive
log config
hidekeys
!
controller T1 2/1/0
framing esf
linecode b8zs
!
controller T1 2/1/1
framing esf
linecode b8zs
!
interface GigabitEthernet0/0
no ip address

```

```
shutdown
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1
ip address 172.25.19.72 255.255.255.0
duplex auto
speed auto
media-type rj45
ip rsvp bandwidth 85 85
!
ip route 0.0.0.0 0.0.0.0 172.25.19.1
!
ip http server
ip rsvp policy preempt
!
control-plane
!
voice-port 2/0/0
!
voice-port 2/0/1
!
dial-peer voice 1 voip
description TO RSVP GW-2
destination-pattern 2001
voice-class codec 1
voice-class sip rsvp-fail-policy audio post-alert optional keep-alive interval 60
voice-class sip rsvp-fail-policy audio post-alert mandatory disconnect retry 2 interval 30
interval 60
session protocol sipv2
session target ipv4:172.25.19.71
incoming called-number 1001
req-qos controlled-load
ip qos defending-priority 65534
ip qos preemption-priority 45
!
dial-peer voice 2 pots
destination-pattern 1001
port 2/0/0
!
dial-peer voice 3 voip
description TO CME-1
destination-pattern 6001
session protocol sipv2
session target ipv4:172.25.19.73
req-qos controlled-load
acc-qos controlled-load
ip qos defending-priority 65
!
dial-peer voice 4 voip
description TO CME-2
destination-pattern 7001
session protocol sipv2
session target ipv4:172.25.19.74
ip qos dscp af21 media rsvp-fail
ip qos dscp af21 media rsvp-pass
!
dial-peer voice 100 voip
description TO THE CUCM
destination-pattern 5000
session protocol sipv2
session target ipv4:172.25.19.3
!
```

```

sip-ua
handle-replaces
!
alias exec sdp show running-config | sec dial-peer
!
line con 0
exec-timeout 0 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
!
scheduler allocate 20000 1000
!
End

```

SIP Video RSVP Preconditions on an H.320-SIP Gateway: Example

The following example shows how to configure SIP video RSVP preconditions for video calls on an H.320-SIP video gateway:

```

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname H320_GW2
!
voice-card 0
no dspfarm
!
voice-card 1
dspfarm
!
ip cef
!
isdn switch-type primary-ni
!
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
h323
call start slow
!
voice class called number pool 100
index 1 5551005 - 5551015
!
voice class called number pool 7002
index 1 7003 - 7018
!
controller T1 3/0
framing esf
linecode b8zs
pri-group timeslots 1-24
!
interface GigabitEthernet0/1
ip address 172.25.19.72 255.255.255.0
duplex auto
speed auto
media-type rj45
ip rsvp bandwidth 400 400

```

```

!
interface Serial3/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn protocol-emulate network
  isdn integrate calltype all
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1
!
ip http server
!
voice-port 3/0:23
  voice-class called-number-pool 100
!
dial-peer voice 1 pots
  description INCOMING DP FOR 8000
  information-type video
  incoming called-number 800
  video calltype h320
  bandwidth maximum 384
  direct-inward-dial
  forward-digits all
!
dial-peer voice 3 voip
  description OUTGOING DP FOR 8000
  destination-pattern 8001
  voice-class sip rel1xx require "100rel"
  session protocol sipv2
  session target ipv4:172.25.19.77
  req-qos controlled-load audio
  req-qos controlled-load video
  acc-qos controlled-load audio
  acc-qos controlled-load video
!
line con 0
  exec-timeout 0 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
!
scheduler allocate 20000 1000
!
End

```

SIP Video RSVP Preconditions on an H.324-SIP Gateway: Example

The following example shows how to configure SIP video RSVP preconditions for video calls on an H.324-SIP video gateway:

```

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname LB-5400-uut7
!
boot-start-marker
no boot startup-test
boot-end-marker
!

```

```

logging message-counter syslog
logging buffered 500000
no logging console
!
resource-pool disable
no aaa new-model
voice-card 1
!
voice-card 7
!
ip source-route
!
ip cef
!
no ipv6 cef
multilink bundle-name authenticated
isdn switch-type primary-ni
!
voice service voip
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  fax protocol none
  h323
  call start slow
  sip
  no update-callerid
!
voice class codec 200
  codec preference 1 g711ulaw
  video codec h261
  video codec h263
!
voice class codec 301
  codec preference 1 g711ulaw
  video codec h263
  video codec mpeg4
!
license feature gsmamrnb-codec-pack
!
archive
  log config
  hidekeys
!
controller T1 7/0
  framing ESF
  linecode b8zs
  pri-group timeslots 1-24
!
controller T1 7/1
!
interface GigabitEthernet0/0
  ip address 172.25.19.37 255.255.0.0
  no ip proxy-arp
  duplex auto
  speed auto
  negotiation auto
  ip rsvp bandwidth 1000 1000
!
interface GigabitEthernet0/1
  no ip address
  shutdown
  duplex auto

```

```
speed auto
negotiation auto
!
interface Serial0/0
no ip address
shutdown
clock rate 2000000
no fair-queue
!
interface Serial0/1
no ip address
shutdown
clock rate 2000000
!
interface Serial7/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn timer T310 400000
isdn protocol-emulate network
no cdp enable
!
ip default-gateway 172.25.1.1
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.25.1.1
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
no ip http server
!
control-plane
!
voice-port 7/0:D
!
mgcp fax t38 ecm
!
dial-peer voice 301 voip
voice-class sip rel1xx require "100rel"
voice-class sip calltype-video
session protocol sipv2
session target ipv4:172.25.19.5
incoming called-number 8000
req-qos controlled-load audio
req-qos controlled-load video
acc-qos controlled-load audio
acc-qos controlled-load video
codec g711ulaw
!
dial-peer voice 300 pots
destination-pattern 8000
progress_ind alert strip
information-type video
direct-inward-dial
port 7/0:D
forward-digits all
!
ss7 mtp2-variant Company 0
ss7 mtp2-variant Company 1
ss7 mtp2-variant Company 2
ss7 mtp2-variant Company 3
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
```

```

login
!
exception data-corruption buffer truncate
end

```

RSVP Preconditions Behavior Verification in SIP Calls: Example

The following is sample output from the **show sip-ua calls** command to verify RSVP preconditions settings and behavior when mandatory QoS is configured at both endpoints and RSVP has succeeded:

```

Router# show sip-ua calls

SIP UAC CALL INFO

    Number of SIP User Agent Client(UAC) calls: 0

SIP UAS CALL INFO

Call 1
SIP Call ID           : F31FEA20-CFF411DC-8068DDB4-22C622B8@172.18.19.73
State of the call     : STATE_ACTIVE (7)
Substate of the call  : SUBSTATE_NONE (0)
Calling Number        : 6001
Called Number         : 1001
Bit Flags              : 0x8C4401E 0x100 0x4
CC Call ID            : 30
Source IP Address (Sig) : 172.18.19.72
Destn SIP Req Addr:Port : 172.18.19.73:5060
Destn SIP Resp Addr:Port : 172.18.19.73:64440
Destination Name      : 172.18.19.73
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object       : 0x0
Media Mode             : flow-through
Media Stream 1
State of the stream   : STREAM_ACTIVE
Stream Call ID        : 30
Stream Type           : voice-only (0)
Negotiated Codec      : g711ulaw (160 bytes)
Codec Payload Type    : 0
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
Media Source IP Addr:Port : 172.18.19.72:18542
Media Dest IP Addr:Port  : 172.18.19.73:16912
Orig Media Dest IP Addr:Port : 0.0.0.0:0
QoS ID                : -2
Local QoS Strength    : Mandatory
Negotiated QoS Strength : Mandatory
Negotiated QoS Direction : SendRecv
Local QoS Status       : Success

Options-Ping    ENABLED:NO    ACTIVE:NO
Number of SIP User Agent Server(UAS) calls: 1

```

The following is sample output from the **show sip-ua calls** command to verify RSVP preconditions settings and behavior when optional QoS is configured at both endpoints and RSVP has succeeded:

```

Router# show sip-ua calls

SIP UAC CALL INFO

```



```

Number of SIP User Agent Client(UAC) calls: 0

SIP UAS CALL INFO

Call 1
SIP Call ID           : 867EA226-D01311DC-8041CA97-F9A5F4F1@172.18.19.73
State of the call     : STATE_ACTIVE (7)
Substate of the call  : SUBSTATE_NONE (0)
Calling Number        : 6001
Called Number         : 1001
Bit Flags             : 0x8C4401E 0x100 0x4
CC Call ID           : 30
Source IP Address (Sig) : 172.18.19.72
Destn SIP Req Addr:Port : 172.18.19.73:5060
Destn SIP Resp Addr:Port: 172.18.19.73:25055
Destination Name      : 172.18.19.73
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object       : 0x0
Media Mode            : flow-through
Media Stream 1
State of the stream   : STREAM_ACTIVE
Stream Call ID        : 30
Stream Type           : voice-only (0)
Negotiated Codec      : g711ulaw (160 bytes)
Codec Payload Type    : 0
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
Media Source IP Addr:Port: 172.18.19.72:17556
Media Dest IP Addr:Port : 172.18.19.73:17966
Orig Media Dest IP Addr:Port : 0.0.0.0:0
QoS ID                : -2
Local QoS Strength    : Optional
Negotiated QoS Strength : Optional
Negotiated QoS Direction : SendRecv
Local QoS Status      : Success

Options-Ping    ENABLED:NO    ACTIVE:NO
Number of SIP User Agent Server(UAS) calls: 1

```

The following is sample output from the **show sip-ua calls** command to verify RSVP preconditions settings and behavior when optional QoS is configured at both endpoints and RSVP has failed:

```

Router# show sip-ua calls

SIP UAC CALL INFO

Number of SIP User Agent Client(UAC) calls: 0

SIP UAS CALL INFO

Call 1
SIP Call ID           : 867EA226-D01311DC-8041CA97-F9A5F4F1@172.18.19.73
State of the call     : STATE_ACTIVE (7)
Substate of the call  : SUBSTATE_NONE (0)
Calling Number        : 6001
Called Number         : 1001
Bit Flags             : 0x8C4401E 0x100 0x4
CC Call ID           : 30
Source IP Address (Sig) : 172.18.19.72
Destn SIP Req Addr:Port : 172.18.19.73:5060
Destn SIP Resp Addr:Port: 172.18.19.73:25055
Destination Name      : 172.18.19.73
Number of Media Streams : 1

```

```

Number of Active Streams: 1
RTP Fork Object       : 0x0
Media Mode            : flow-through
Media Stream 1
  State of the stream  : STREAM_ACTIVE
  Stream Call ID       : 30
  Stream Type          : voice-only (0)
  Negotiated Codec     : g711ulaw (160 bytes)
  Codec Payload Type   : 0
  Negotiated Dtmf-relay : inband-voice
  Dtmf-relay Payload Type : 0
  Media Source IP Addr:Port: 172.18.19.72:17556
  Media Dest IP Addr:Port : 172.18.19.73:17966
  Orig Media Dest IP Addr:Port : 0.0.0.0:0
  QoS ID               : -2
  Local QoS Strength   : Optional
  Negotiated QoS Strength : Optional
  Negotiated QoS Direction : SendRecv
  Local QoS Status     : Fail

Options-Ping    ENABLED:NO    ACTIVE:NO
  Number of SIP User Agent Server(UAS) calls: 1

```

The following is sample output from the **show sip-ua calls** command on the originating gateway (OGW) to verify RSVP preconditions settings and behavior when optional QoS is configured on the OGW, mandatory QoS is configured on the terminating gateway (TGW), and RSVP has succeeded:

```

Router# show sip-ua calls

SIP UAC CALL INFO

  Number of SIP User Agent Client(UAC) calls: 0

SIP UAS CALL INFO

Call 1
SIP Call ID           : 867EA226-D01311DC-8041CA97-F9A5F4F1@172.18.19.73
  State of the call    : STATE_ACTIVE (7)
  Substate of the call : SUBSTATE_NONE (0)
  Calling Number       : 6001
  Called Number        : 1001
  Bit Flags            : 0x8C4401E 0x100 0x4
  CC Call ID          : 30
  Source IP Address (Sig) : 172.18.19.72
  Destn SIP Req Addr:Port : 172.18.19.73:5060
  Destn SIP Resp Addr:Port: 172.18.19.73:25055
  Destination Name     : 172.18.19.73
  Number of Media Streams : 1
  Number of Active Streams: 1
  RTP Fork Object      : 0x0
  Media Mode           : flow-through
Media Stream 1
  State of the stream  : STREAM_ACTIVE
  Stream Call ID       : 30
  Stream Type          : voice-only (0)
  Negotiated Codec     : g711ulaw (160 bytes)
  Codec Payload Type   : 0
  Negotiated Dtmf-relay : inband-voice
  Dtmf-relay Payload Type : 0
  Media Source IP Addr:Port: 172.18.19.72:17556
  Media Dest IP Addr:Port : 172.18.19.73:17966
  Orig Media Dest IP Addr:Port : 0.0.0.0:0
  QoS ID               : -2

```

```
Local QoS Strength      : Optional
Negotiated QoS Strength : Mandatory
Negotiated QoS Direction : SendRecv
Local QoS Status       : Success
```

```
Options-Ping    ENABLED:NO    ACTIVE:NO
Number of SIP User Agent Server(UAS) calls: 1
```

Additional References

The following sections provide references related to SIP RSVP preconditions on SIP TDM gateways and Cisco Unified CME.

Related Documents

Related Topic	Document Title
Channel aggregation in video conferencing using ISDN	H.320 Video - ISO/IEC-13871 Bonding
Configuring ISDN PRI interfaces to support integration of data and voice calls on multiservice access routers	Integrating Data and Voice Services for ISDN PRI Interfaces on Multiservice Access Routers
Configuring midcall video escalation/deescalation capability for an H.324 SIP call	Cisco Integrated 3G-324M Gateway
Configuring RSVP	“Configuring RSVP” module in the subsections about RSVP in the “Signalling” part in the <i>Cisco IOS Quality of Service Solutions Configuration Guide</i>
Quality of service solutions	Cisco IOS Quality of Service Solutions Configuration Guide
Roadmap for SIP features	SIP Features Roadmap
SIP overview	Overview of SIP

Standards

Standard	Title
None	—

MIBs

MIB	MIBs Link
None	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 2205	Resource ReSerVation Protocol (RSVP)
RFC 2872	Application and Sub Application Identity Policy Element for Use with RSVP
RFC 3181	Signaled Preemption Priority Policy Element
RFC 3182	Identity Representation for RSVP

RFC	Title
RFC 3312	<i>Integration of Resource Management and Session Initiation Protocol (SIP)</i>
RFC 3725	<i>Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)</i>
RFC 4032	<i>Update to the Session Initiation Protocol (SIP) Preconditions Framework</i>

Technical Assistance

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

Feature Information for SIP RSVP Features

Table 1 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

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



Note

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

Table 1 Feature Information for SIP RSVP Features

Feature Name	Releases	Feature Information
RSVP Application ID Support	12.2(33)SRB 12.4(6)T 12.4(22)T	<p>The RSVP Application ID Support feature was introduced in Cisco IOS Release 12.2(33)SRB (and later integrated into Cisco IOS Release 12.4(6)T) to support application-specific reservations, which enhance the granularity for local policy-match criteria so that you can manage QoS on the basis of application type.</p> <p>To accommodate the RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express feature, the RSVP Application ID Support feature was modified in Cisco IOS Release 12.4(22)T. Detailed information for this feature is provided in the following section:</p> <ul style="list-style-type: none"> • Configuring SIP RSVP Application ID Support, page 7 <p>For more information, see the “Configuring RSVP” module in the subsections about RSVP in the “Signalling” part in the <i>Cisco IOS Quality of Service Solutions Configuration Guide</i>.</p> <p>The following commands were introduced or modified: ip qos defending-priority, ip qos policy-locator, ip qos preemption-priority.</p>
RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express	12.4(22)T	<p>The RSVP Preconditions for Audio on SIP-TDM Gateway and Cisco Unified Communications Manager Express feature enables SIP RSVP as a precondition for establishment of SIP calls on SIP-TDM gateways and Cisco Unified CME, enabling interoperability with both Cisco Unified CM and Cisco Unified CVP. Detailed information is provided in the following section:</p> <ul style="list-style-type: none"> • Configuring SIP RSVP Preconditions, page 13 <p>The following commands were introduced or modified: handle-replaces, ip qos dscp, show sip-ua calls, voice-class sip rsvp-fail-policy.</p>

Table 1 **Feature Information for SIP RSVP Features**

Feature Name	Releases	Feature Information
RSVP Preconditions for Video Gateway	12.4(24)T	<p>The RSVP Preconditions for Video Gateway feature expands existing support for SIP video calls on H.324-SIP video gateways to include H.320-SIP video gateways. Additionally, this feature adds support for SIP video RSVP preconditions for SIP video calls on both H.320-SIP and H.324-SIP video gateways. Detailed information is provided in the following section:</p> <ul style="list-style-type: none"><li data-bbox="799 520 1354 552">• Configuring SIP RSVP Preconditions, page 13 <p>The following command was introduced: voice-class sip calltype-video.</p>

Glossary

CAC—Call Admission Control.

CME—Communications Manager Express.

CVP—Customer Voice Portal.

GW—gateway.

mline—The media-level section of an SDP session begins and ends with an “m” line that confines the information about the media stream.

MOH—Music on Hold.

QoS—quality of service.

RSVP—Resource Reservation Protocol.

SDP—Session Description Protocol.

SIP—Session Initiation Protocol.

TDM—time-division multiplexing.

UA—user agent.

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Configuring SIP QoS Features

This chapter discusses the following features that affect quality of service (QoS) in SIP networks:

- Enhanced Codec Support for SIP Using Dynamic Payloads
- Measurement-Based Call Admission Control for SIP
- SIP Gateway Support of RSVP
- SIP Gateway Support of ‘tel’ URL
- SIP: Hold Timer Support
- SIP Media Inactivity Timer
- SIP Stack Portability



Note This feature is described in the [“Configuring SIP Message, Timer, and Response Features” on page 1.](#)

Feature History for Enhanced Codec Support for SIP Using Dynamic Payloads

Release	Modification
12.2(11)T	This feature was introduced.

Feature History for Measurement-Based Call Admission Control for SIP

Release	Modification
12.2(15)T	This feature was introduced.

Feature History for SIP Gateway Support of RSVP

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB1	This feature was implemented on an additional platform.
12.2(8)T	This feature was integrated into this release.
12.2(11)T	This feature was implemented on additional platforms.



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Feature History for SIP Gateway Support of 'tel' URL

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(2)XB1	This feature was implemented on an additional platform.
12.2(8)T	This feature was integrated into this release.
12.2(11)T	This feature was implemented on additional platforms.

Feature History for SIP: Hold Timer Support

Release	Modification
12.3(13)	This feature was introduced.

Feature History for SIP Media Inactivity Timer

Release	Modification
12.2(2)XB	This feature was introduced.
12.2(8)T	This feature was integrated into this release.
12.2(11)T	This feature was implemented on additional platforms.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Prerequisites for SIP QoS, page 2](#)
- [Restrictions for SIP QoS, page 4](#)
- [Information About SIP QoS, page 4](#)
- [How to Configure SIP QoS Features, page 18](#)
- [Configuration Examples for SIP QoS Features, page 51](#)
- [Additional References, page 61](#)

Prerequisites for SIP QoS

Measurement-Based Call Admission Control for SIP Feature

- By default, gateways support reliable provisional responses. That is, no additional configuration tasks are necessary to enable reliable provisional responses.

**Note**

For information on configuring reliable provisional responses including enabling the feature again if it was disabled, see *SIP Gateway Support of RSVP and TEL URL*.

- Configure a basic VoIP network.
- Enable Service Assurance Agent (SAA) Responder on the originating and terminating gateway.



Note For information on configuring Service Assurance Agent, see [Network Monitoring Using Cisco Service Assurance Agent](#).



Note For information about configuring VoIP, see [Cisco IOS Voice, Video, and Fax Configuration Guide](#), Release 12.2.

For information about configuring reliable provisional responses including reenabling the feature if it was disabled, see [SIP Gateway Support of RSVP and TEL URL](#).

For information about configuring Service Assurance Agent, see [Network Monitoring Using Cisco Service Assurance Agent](#).

SIP Gateway Support of RSVP and SIP Gateway Support of 'tel' URL Features

- Enable RSVP on the appropriate gateway interfaces by using the **ip rsvp bandwidth** command.



Note For details on the command, see the [Cisco IOS Quality of Service Solutions Command Reference](#), Release 12.3.

- Enable weighted fair queuing (WFQ) on these interfaces by using the **fair-queue** command. This ensures that the voice packets get priority over the interface.



Note For details on the command, see the [Cisco IOS Quality of Service Solutions Command Reference](#), Release 12.3. For an example, see the “SIP Gateway Support of RSVP and TEL URL: Example” section on page 51.

- Set the desired and acceptable quality of service (QoS) levels in your dial peers by using the **req-qos** and **acc-qos** dial-peer configuration commands.

Bandwidth reservation is not attempted unless the desired QoS for the associated dial peer is set to **controlled-load** or **guaranteed-delay**. If the desired QoS level is set to the default of **best-effort**, bandwidth reservation is *not* attempted. With the **req-qos** command, synchronized RSVP is attempted for a SIP call as long as the desired (requested) QoS for the associated dial peer is set to **controlled-load** or **guaranteed-delay**.



Note For details on the commands, see the [Cisco IOS Voice Command Reference](#), Release 12.3. For an example, see the “SIP Gateway Support of RSVP and TEL URL: Example” section on page 51.

Restrictions for SIP QoS

Enhanced Codec Support for SIP Using Dynamic Payloads Feature

- Dynamic payload values can be configured using the **rtp payload-type** command only for the payload types listed in [Table 57 on page 8](#).
- Dynamic payloads cannot be configured for the codecs shown in [Table 58 on page 9](#).

Measurement-Based Call Admission Control for SIP Feature

- When detecting network congestion, the PSTN fallback feature does not affect an existing call; it affects only subsequent calls.
- Only a single calculated planning impairment factor (ICPIF) delay or loss value is allowed per system.
- A small additional call setup delay can be expected for the first call to a new IP destination.
- The Service Assurance Agent Responder feature, a network congestion analysis mechanism, cannot be configured for non-Cisco devices.

SIP Gateway Support of RSVP and SIP Gateway Support of 'tel' URL Features

- Bandwidth reservation (QoS) is not supported for Session Description Protocol (SDP) changes between 183 Session Progress/180 Alerting and 200 OK responses.
- Bandwidth reservation (QoS) is not attempted if the desired QoS level is set to the default of **best-effort**. The desired QoS for the associated dial peer must be set to **controlled-load** or **guaranteed-delay**.
- Distributed Call Signaling (DCS) headers and extensions are not supported.
- SIP gateways do not support codecs other than those listed in the SIP codec table listed in [Table 56 on page 5](#). When an unsupported codec is selected during configuration of the dial peers, the action taken depends on the selected gateway:
 - If on the originating gateway, an appropriate SIP debug trace is presented, indicating the failure to originate the SIP call leg.
 - If on the terminating gateway, an appropriate SIP response (4xx) with a warning indicating incompatible media types is sent.

Information About SIP QoS

To configure SIP QoS features, you should understand the following concepts:

- [Enhanced Codec Support for SIP Using Dynamic Payloads, page 5](#)
- [Measurement-Based Call Admission Control for SIP, page 9](#)
- [SIP Gateway Support of RSVP and TEL URL, page 13](#)
- [SIP: Hold Timer Support, page 15](#)
- [SIP Media Inactivity Timer, page 17](#)

Enhanced Codec Support for SIP Using Dynamic Payloads

The Enhanced Codec Support for SIP Using Dynamic Payloads feature enhances codec selection and payload negotiation between originating and terminating SIP gateways.

This feature offers the following benefits:

- Expanded dynamic payload support on Cisco IOS gateways, resulting in enhanced bandwidth control
- Expanded ability to advertise and negotiate all codecs available on a given platform
- Expanded interoperability and interconnectivity between gateways, applications, and services in the network

The feature provides the SIP enhancements described in the following sections:

- [Additional Codec Support, page 5](#)
- [Payload Type Selection, page 6](#)
- [Advertising Codec Capabilities, page 7](#)

Additional Codec Support

Codecs are a digital signal processor (DSP) software algorithm used to compress or decompress speech or audio signals. Previous implementations of the SIP stack on Cisco IOS gateways supported only a subset of the available codecs for each platform.

Support for codecs varies on different platforms. See [Table 56](#) for a listing of SIP codec support by platform. Use the `codec ?` command to determine the codecs available on a specific platform.

Table 56 *SIP Codec Support by Platform and Cisco IOS Release*

Codec	Cisco 2600 Series, Cisco 3620, Cisco 3640, Cisco 3660	Cisco 7200 Series	Cisco AS5300	Cisco AS5350, Cisco AS5400, Cisco AS5850
Clear-channel	Yes	No	Yes	Yes
G711alaw	Yes	Yes	Yes	Yes
G711ulaw	Yes	Yes	Yes	Yes
G723r53	Yes	Yes	Yes	Yes
G723r63	Yes	Yes	Yes	Yes
G723r53	Yes	Yes	Yes	Yes
G723r63	Yes	Yes	Yes	Yes
G726r16	Yes	Yes	Yes	Yes
G726r24	Yes	Yes	Yes	Yes
G726r32	Yes	Yes	Yes	Yes
G728	Yes	Yes	Yes	No
G729br8	Yes	Yes	Yes	Yes
G729r8	Yes	Yes	Yes	Yes

Table 56 SIP Codec Support by Platform and Cisco IOS Release (continued)

Codec	Cisco 2600 Series, Cisco 3620, Cisco 3640, Cisco 3660	Cisco 7200 Series	Cisco AS5300	Cisco AS5350, Cisco AS5400, Cisco AS5850
GSM-EFR	Yes	No	Yes	No
GSM-FR	Yes	No/No	Yes	Yes

Payload Type Selection

Payload types define the content and format of Real-Time Transport Protocol (RTP) packets and the resulting stream of data generated by the RTP flow. The payload type defines the codec in use and is identified in the payload type field of the header of each RTP packet. There are two mechanisms for specifying payload type, static and dynamic.

Static payload types are assigned to specific RTP formats by RFC 1890 and these mappings are registered with the Internet Assigned Numbers Authority (IANA). Although not required, static payload types can also be mapped to RTP encodings using the `rtptime` attribute. The following SIP-supported codecs have static payload values defined by the IANA:

- G711ulaw
- G711alaw
- G723r63
- G726r32
- G728
- G729r8
- GSM-FR

Dynamic payload values are used for codecs that do not have static payload values defined. Dynamic payload types do not have fixed mappings, and must be mapped to RTP encodings within the Session Description Protocol (SDP) itself using the `a=rtptime` line. The feature allows dynamic payload values to be used for the following codecs with no static payload values defined:

- Clear-channel
- G726r16
- G726r24
- GSM-EFR

Of the four codecs listed that allow dynamic payload values to be assigned, only the payload type for the clear-channel codec can be configured using the command-line interface (CLI). The remaining G.726r16, G.726r24 and GSM-EFR codecs are selected on a per-call basis by the SIP subsystem. The dynamic payload range is assigned by the IANA, with values from 96 to 127. The SIP subsystem looks for and uses the first value in the range that is both available and not reserved for Cisco IOS applications. Once a dynamic payload value is picked for a particular payload type, it cannot be used for other payload types. Of the 32 available IANA values, those reserved for special Cisco IOS applications are listed in [Table 57](#). To configure dynamic payload values for the payload types listed in [Table 57](#), use the `rtptime` payload-type command; otherwise the default values for the payload types are used.

Table 57 **Default Dynamic Payload Values**

Dynamic Payload Type	Default Dynamic Payload Value	Supported by SIP
Cisco-rtp-dtmf-relay	121	Yes
Named Signal Event	100	Yes
Named Telephony Event	101	Yes
Cisco-cas-payload	123	No
Cisco-clear-channel	125	No
Cisco-codec-fax-ack	97	No
Cisco-codec-fax-ind	96	No
Cisco-fax-relay	122	No
Cisco-pcm-switch-over-alaw	127	No
Cisco-pcm-switch-over-ulaw	126	No

**Note**

After a dynamic payload value has been assigned from the reserved range, it cannot be used for any other payload types.

Advertising Codec Capabilities

The dynamic payload value selected by the SIP subsystem is advertised in the outgoing SIP INVITE request. The Enhanced Codec Support for SIP Using Dynamic Payloads feature supports dynamic payloads by expanding the SIP subsystem ability to advertise and negotiate available codecs. SIP uses the connection, media, and attribute fields of the SDP message during connection negotiation.

The feature supports the following Internet Engineering Task Force (IETF) drafts:

- [draft-ietf-avt-rtp-mime-06.txt](#), *MIME Type Registration of RTP Payload Formats* (further developed and later published as RFC 3555).
- [draft-ietf-avt-profile-new-12.txt](#), *RTP Profile for Audio and Video Conferences with Minimal Control* (further developed and later published as RFC 3551).

The following sample SIP INVITE message shows the payload value and codec selection resulting from the payload negotiation process. The media m= field includes the added payload value. The attribute a= field includes the selected codec. In this outgoing INVITE message, the first available dynamic payload value of 115 is selected by the SIP subsystem for a GSM-EFR codec.

```
INVITE sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601" <sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 00:05:14 GMT
Call-ID: 4326879A-14EF11CC-80069792-19DC655A@172.18.193.98
Cisco-Guid: 1092278192-351211980-2147784594-433874266
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730944314
Contact: <sip:36601@172.18.193.98:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 228
```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 6973 8772 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 17928 RTP/AVP 18 115
a=rtpmap:18 G729/8000
a=rtpmap:115 GSM-EFR/8000

```

G723 Codec Versions

In addition to the previously supported G.723r63 version of the G.723 codec, the feature supports the following versions:

- G723r53, where the number 53 indicates the bit rate of 5.3 kbps
- G723ar53, where the letter a indicates support for Annex A, which specifies voice activity detection (VAD)
- G723ar63, where the number 63 indicates a bit rate of 6.3 kbps

A static payload value of 4 is used for all versions of the G.723 codec.

Expanded codec support allows the originating and terminating gateways to advertise and negotiate additional codec capabilities. Cisco implements support for multiple G.723 codec versions by using a=fmtp and a=rtpmap attributes in the SDP body of outgoing INVITE requests to define the G.723 codec version. For the G.723 codec, the value of a=fmtp is 4 (the IANA assigned static value), and the annexa value is either yes or no. The default for annexa is yes.

[Table 58](#) lists the possible codec configurations, that, taken together with Annex A support at the remote end, result in selecting the negotiated codec.

Table 58 G723 Codecs

Configured Codec(s)	Remote End Supports Annex A	Negotiated Codec
G723r63	annexa = no or no fmtp line	G723r63
G723r53	annexa = no or no fmtp line	G723r53
G723r53 and G723r63	annexa = no or no fmtp line	G723r63
G723ar63	annexa=yes or no fmtp line	G723ar63
G723ar53	annexa=yes or no fmtp line	G723ar53
G723ar53 and G723ar63	annexa=yes or no fmtp line	G723ar63
G723ar53 and G723r53	annexa=yes or no fmtp line	G723ar53
G723ar63 and G723r63	annexa=yes or no fmtp line	G723ar63
G723ar63 and G723r53	annexa=yes or no fmtp line	G723ar63
G723ar53 and G723r63	annexa=yes or no fmtp line	G723ar63
G723ar53, G723r53, G723ar63, and G723r63	annexa = no or no fmtp line	G723ar63

The following partial SDP body shows the media m= field and attribute a= field for a gateway with G.723 codecs and Annex A specified.

```

m=audio 62986 RTP/AVP 4
a=rtpmap:4 G723/8000
a=fmtp:4 annexa=yes

```


G729 Codec Versions

The feature supports the following versions of G.729 codecs:

- G729r8, where r8 indicates the bit rate of 8 kbps
- G729br8, where b indicates support for Annex B, which specifies VAD, Discontinuity Transmission (DTX), and Comfort Noise generation (CNG).

A static payload value of 18 is used for all versions of the G.729 codec.

Cisco implements support for multiple G.729 codec versions by using a=fmtp and a=rtpmap attributes in the SDP body of outgoing INVITE requests. For the G.729 codec, the value of a=fmtp is 18 (the IANA assigned static value), and the annexb value is either yes or no. The default for annexb is yes.

Table 59 lists the possible codec configuration that, taken together with Annex B support at the remote end, result in selecting the negotiated codec.

Table 59 **G729 Codecs**

Configured Codec(s)	Remote End Supports Annex B	Negotiated Codec
G729r8	annexb= no or no fmtp line	G729r8
G729br8	annexb = yes or no fmtp line	G729br8
G729r8 and G729br8	annexb= yes or no fmtp line	G729br8
G729r8 and G729br8	no fmtp line	G729br8
G729r8 and G729br8	annexb=no or no fmtp line	G729r8
G729r8 and G729br8	annexb=yes	G729br8

The following partial SDP body shows the media m= field and attribute a= field for a gateway with G.729 codecs and Annex B specified:

```
m=audio 17928 RTP/AVP 18
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
```

Measurement-Based Call Admission Control for SIP

The Measurement-Based Call Admission Control for SIP feature implements support within SIP to monitor IP network capacity and reject or redirect calls based on congestion detection.

Feature benefits include the following:

- PSTN fallback
 - Automatically routes a call to an alternate destination when the data network is congested at the time of call setup, thereby enabling higher call completion rates.
 - Enables the service provider to give a reasonable guarantee about the quality of the conversation to VoIP users at the time of call admission.
 - PSTN fallback provides network congestion measurement, including delay, jitter, and packet loss information for the configured IP addresses.
 - A new call need not wait for probe results before being admitted, thereby minimizing delays.

- Call admission control
 - Configurable call treatment allows the Internet service provider (ISP) the flexibility to configure how the call will be treated when local resources to process the call are not available.
 - Resource unavailable signaling allows you to automatically busy out channels when local resources are not available to handle the call.
 - User-selected thresholds allow you the flexibility to configure thresholds to determine resource availability.

The Measurement-Based Call Admission Control for SIP feature does the following:

- Verifies that adequate resources are available to carry a successful VoIP session.
- Implements a mechanism to prevent calls arriving from the IP network from entering the gateway when required resources are not available to process the call.
- Supports measurement-based call admission control (CAC) processes.

Before the CAC feature was developed, gateways did not have a mechanism to check for IP network congestion and resource unavailability. Although quality of service (QoS) mechanisms provide a level of low latency and guaranteed delivery that is required for voice traffic, CAC mechanisms are intended to extend the capabilities of QoS to protect voice traffic from being negatively affected by other voice traffic. CAC is used to gracefully deny network access under congestion conditions and provide alternative call rerouting to prevent dropped or delayed calls. There are a variety of CAC mechanisms, including the following:

- Measurement-based CAC, which uses probes to look ahead into the packet network to gauge the state of the network to determine whether to allow a new call.
- Resource-based CAC, which calculates resources needed for the call, determines their availability, and reserves those resources.

The Cisco IOS VoiceXML feature provides an alternative to Resource Reservation Protocol (RSVP) for VoIP service providers that do not deploy RSVP.

The new feature implements measurement-based CAC using the mechanisms described in the following sections:

- [Service Assurance Agents, page 10](#)
- [Calculated Planning Impairment Factor, page 11](#)
- [PSTN Fallback, page 12](#)
- [Call Admission Thresholds, page 13](#)
- [Call Treatment Options, page 13](#)
- [Resource Unavailable Signaling, page 13](#)

Service Assurance Agents

Service Assurance Agents (SAA) is a generic network management feature that provides a mechanism for network congestion analysis. SAA determine latency, delay, and jitter and provides real-time ICPIF calculations before establishing a call across an IP infrastructure. The SAA Responder feature uses SAA probes to traverse the network to a given IP destination and measure the loss and delay characteristics of the network along the path traveled. These values are returned to the outgoing gateway to use in making a decision on the condition of the network and its ability to carry a call. Threshold values for rejecting a call are configured at the outgoing gateway (see the [“PSTN Fallback” section on page 12](#)).

Each probe consists of multiple packets, a configurable parameter of this feature. SAA packets emulate voice packets and receive the same priority as voice throughout the entire network. The delay, loss, and ICPIF values entered into the cache for the IP destination are averaged from all the responses. If the call uses G.729 and G.711 codecs, the probe packet sizes mimic those of a voice packet for that codec. Other codecs use G.711-like probes. In Cisco IOS software releases later than Release 12.1(3)T, other codec choices may also be supported with their own specific probes.

The IP precedence of the probe packets can also be configured to simulate the priority of a voice packet more closely. This parameter should be set equal to the IP precedence used for other voice media packets in the network.

SAA probes used for CAC go out randomly on ports selected from within the top end of the audio User Datagram Protocol (UDP) defined port range (16384 to 32767). Probes use a packet size based on the codec the call will use. IP precedence can be set if desired, and a full Realtime Transport Protocol (RTP), UDP, or IP header is used just as a real voice packet would carry. The SAA Responder feature was called Response Time Reporter (RTR) in earlier releases of Cisco IOS software.

The SAA Responder feature can not be configured for non-Cisco devices. For a complete description of SAA configuration, see the [Cisco IOS Configuration Fundamentals and Network Management Configuration Guide, Release 12.3](#).

Calculated Planning Impairment Factor

The Cisco IOS VoiceXML feature supports the determination of ICPIF, as specified by International Telecommunications Union (ITU) standard G.113. The SIP subsystem calculates an impairment factor for network conditions to a particular IP address. ICPIF checks for end-to-end resource availability by calculating a Total Impairment Value, which is a function of codecs used and loss or delay of packets. You can configure router resources to make call admission decisions, using either the ICPIF threshold, or by setting delay and loss thresholds.

Configurable ICPIF values that represent the ITU specification for quality of voice as described in G.113 are the following:

- 5—Very good
- 10—Good
- 20—Adequate
- 30—Limiting case
- 45—Exceptional limiting case
- 55—Customers likely to react strongly

The default value is 20. SAA probe delay and loss information is used in calculating an ICPIF value, which is then used as a threshold for CAC decisions. You can base such decisions on either the ITU interpretation described or on the requirements of an individual customer network.

PSTN Fallback

The Cisco IOS VoiceXML feature supports PSTN Fallback, which monitors congestion in the IP network and either redirects calls to the public switched telephone network (PSTN) or rejects calls based on network congestion. Calls can be rerouted to an alternate IP destination or to the PSTN if the IP network is found unsuitable for voice traffic at that time. You can define congestion thresholds based on the configured network. This functionality allows the service provider to give a reasonable guarantee about the quality of the conversation to VoIP users at the time of call admission.

**Note**

PSTN Fallback does not provide assurances that a VoIP call that proceeds over the IP network is protected from the effects of congestion. This is the function of the other QoS mechanisms, such as IP Real-Time Transport Protocol (RTP) priority or low latency queuing (LLQ).

PSTN Fallback includes the following capabilities:

- Provides the ability to define the congestion thresholds based on the network.
 - Defines a threshold based on ICPIF, which is derived as part of ITU G.113 (see the [“Service Assurance Agents”](#) section on page 10).
 - Defines a threshold based solely on packet delay and loss measurements.
- Uses SAA probes to provide packet delay, jitter, and loss information for the relevant IP addresses. Based on the packet loss, delay, and jitter encountered by these probes, an ICPIF or delay or loss value is calculated.
- Supports calls of any codec. Only G.729 and G.711 have accurately simulated probes. Calls of all other codecs are emulated by a G.711 probe.

The call fallback subsystem has a network traffic cache that maintains the ICPIF or delay or loss values for various destinations. This capability helps performance, because each new call to a well-known destination need not wait for a probe to be admitted because the value is usually cached from a previous call.

Once the ICPIF or delay or loss value is calculated, they are stored in a fallback cache where they remain until the cache ages out or overflows. Until an entry ages out, probes are sent periodically for that particular destination. This time interval is configurable.

Media Information for Fallback Services

SIP reliable provisional responses ensure that media information is exchanged and that resource and network checks can take place prior to connecting the call. The following SIP methods have been implemented to support fallback services:

- INVITE with Session Description Protocol (SDP) body. The PSTN Fallback feature provides support for a new attribute line, `a=rtr`, in the SDP message body. The `rtr` attribute enables support for invoking fallback services. The INVITE message with SDP body provides media connection information, including IP address and negotiated codec.
- Provisional Acknowledgment (PRACK). PRACK allows reliable exchanges of SIP provisional responses between SIP endpoints. When the INVITE message has no SDP body, that is, no delayed media, the terminating gateway sends media information in the 183 Session Progress message and expects the SDP from the originating gateway in the PRACK message.
- Conditions Met (COMET), which indicates if the preconditions for a given call have been met.

Call Admission Thresholds

User-selected thresholds allow you to configure call admission thresholds for local resources and end-to-end memory and CPU resources. You can configure two thresholds, high and low, for each global or interface-related resource. The specified call treatment is triggered when the current value of a resource goes beyond the configured high, and remains in effect until the current resource value falls below the configured low.

Call Treatment Options

You can select how the call should be treated when local resources are not available to handle the call. For example, when the current resource value for any one of the configured triggers for call threshold exceeds the configured threshold, you have the following the call treatment choices:

- TDM hairpinning—Hairpins the calls through the POTS dial peer.
- Reject—Disconnects the call.
- Play message or tone—Plays a configured message or tone to the user.

Resource Unavailable Signaling

The Resource Unavailable Signaling feature supports autobusyout capability, which busies out channels when local resources are not available to handle the call. Autobusyout is supported on both channel-associated signaling (CAS) and primary rate interface (PRI) channels:

- CAS—Uses busyout to signal local resources are unavailable.
- PRI—Uses either service messages or disconnect with correct cause-code to signal resources are unavailable.

SIP Gateway Support of RSVP and TEL URL

The SIP Gateway Support of RSVP and TEL URL feature provides the following SIP enhancements:

- [RSVP, page 14](#)
- [Synchronization with Cisco IOS QoS, page 14](#)
- [TEL URL Format in SIP Messages, page 15](#)
- [SIP and TEL URL Examples, page 15](#)
- [Reliability of SIP Provisional Responses, page 15](#)

This section describes the SIP Gateway Support of RSVP and the SIP Gateway Support of 'tel' URL features. SIP gateways can enable resource reservation using Resource Reservation Protocol (RSVP). Resource reservation on SIP gateways synchronizes RSVP procedures with SIP call establishment procedures, ensuring that the required quality of service (QoS) for a call is maintained across the IP network.

Feature benefits include the following:

- SIP Gateway Support of RSVP and TEL URL enables Quality of Service (QoS), ensuring certain bandwidth reservations for specific calls. The bandwidth reservation can be **best-effort**, in which case the call is completed even if the reservation is not supported by both sides or cannot be established. Or the bandwidth reservation can be *required*, and the call is not set up if the bandwidth reservation is not performed successfully.

- With the reliable provisional response features, you can ensure that media information is exchanged and resource reservation takes place before connecting a call.
- Gateways now accept TEL calls sent through the Internet, which provides interoperability with other equipment that uses TEL URL. The TEL URL feature also gives service providers a way to differentiate services based on the type of call, allowing for deployment of specific services.

RSVP

Before this feature was implemented, SIP applications over IP networks functioned as best-effort services—their media packets were delivered with no performance guarantees. However, SIP gateway support of RSVP and TEL URL ensures quality of service (QoS) by coordinating SIP call signaling and RSVP resource management. This feature reserves sufficient network-layer resources to guarantee bandwidth and bounds on packet loss, delay, and jitter; thus ensuring that the called party’s phone rings only after bandwidth required for the call has been successfully reserved.

Additionally, appropriate changes to the resources reserved for the SIP call are made when mid-call INVITE messages, requiring media change (such as a change of codec) are requested.

Synchronization with Cisco IOS QoS

A QoS module is provided that acts as a broker between the VoIP service-provider interfaces (SPIs) and the Cisco IOS RSVP subsystem. The QoS module enables the VoIP SPIs to initiate resource reservation, modify parameters of an existing reservation, and clean up the reserved resources. The QoS module then communicates the results of the operation to the RSVP subsystem.

The conditions for SIP calls using QoS are as shown in [Table 60](#).

Table 60 Conditions for SIP Calls Using QoS

SIP Call Setup	Result
Bandwidth reservation (QoS) is attempted when:	The desired (requested) QoS for the associated dial peer is set to controlled-load or guaranteed-delay .
Bandwidth reservation (QoS) is not attempted when:	The desired QoS level is set to the default of best-effort .
If bandwidth reservation (QoS) is attempted but fails, the acceptable QoS for the dial peer determines the outcome of the call:	The call proceeds without any bandwidth reservation in place if the acceptable QoS is configured with best-effort .
—	The call is released if the acceptable QoS on either gateway is configured with controlled-load or guaranteed-delay .

The desired QoS and acceptable QoS are configured through Cisco IOS software by using the **req-qos** and **acc-qos** dial-peer configuration commands, respectively.

TEL URL Format in SIP Messages

The SIP Gateway Support of RSVP and TEL URL feature also supports Telephone Uniform Resource Locators or TEL URL. Currently SIP gateways support URLs in the SIP format. SIP URLs are used in SIP messages to indicate the originator, recipient, and destination of the SIP request. However, SIP gateways may also encounter URLs in other formats, such as TEL URLs. TEL URLs describe voice call connections. They also enable the gateway to accept TEL calls sent through the Internet, and to generate TEL URLs in the request line of outgoing INVITEs requests.

SIP and TEL URL Examples

SIP URL

A SIP URL identifies a user's address and appears similar to an email address: `user@host` where *user* is the telephone number and *host* is either a domain name or a numeric network address. For example, the request line of an outgoing INVITE request might appear as:

```
INVITE sip:5550100@example.com; user=phone.
```

The *user=phone* parameter distinguishes that the user address is a telephone number rather than a username.

TEL URL

A TEL URL takes the basic form of *tel:telephone subscriber number*, where *tel* requests the local entity to place a voice call, and *telephone subscriber number* is the number to receive the call. For example:

```
tel:+555-0100
```

For more detailed information on the structure of TEL URL, see RFC 2806, *URLs for Telephone Calls*.

Reliability of SIP Provisional Responses

SIP reliable provisional responses ensure that media information is exchanged and resource reservation can take place prior to connecting the call. Provisional acknowledgement (PRACK) and conditions met (COMET) are two methods that have been implemented.

PRACK allows reliable exchanges of SIP provisional responses between SIP endpoints. COMET indicates if the preconditions for a given call or session have been met.

SIP: Hold Timer Support

The SIP: Hold Timer Support feature provides the ability to terminate a call that has been placed on hold in excess of a configurable time period, and to thereby free up trunk resources.

Feature benefits include the following:

- Improved trunk resource utilization
- Improved network monitoring and management capability

The SIP: Hold Timer Support feature provides a new configurable hold timer that allows you to specify a maximum hold time of up to 2880 minutes. Prior to this feature, there was no mechanism to automatically disconnect a call that had been on hold for a set period of time. When the SIP call hold process occurs in response to ISDN Suspend and Resume messages, a media inactivity timer allows a

gateway to monitor and disconnect a VoIP call if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period. This timer is deactivated when a call is placed on hold and no media packets are sent. As a result, a call is potentially allowed to stay on hold indefinitely.



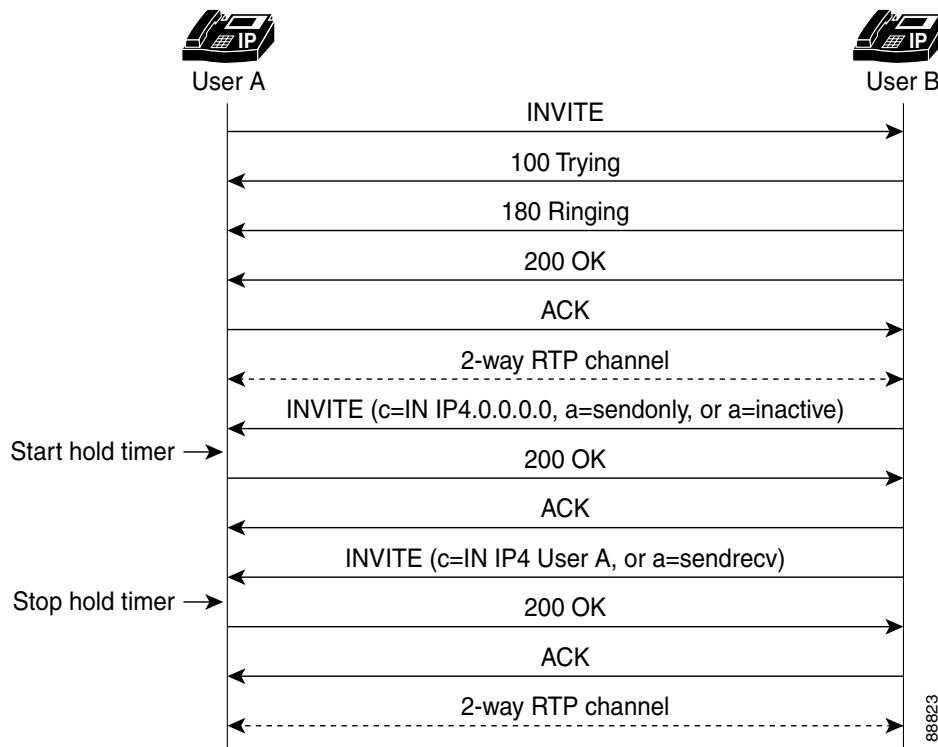
Note

For information on the media inactivity timer, see *SIP Media Inactivity Timer* and *SIP: ISDN Suspend/Resume Support*.

The SIP: Hold Timer Support feature resolves this problem by allowing you to configure a gateway to disconnect a held call when the hold timer is exceeded. The hold timer is activated when a gateway receives a call hold request from the other endpoint, for example, a SIP phone. SIP gateways receive notice of a call hold when the originating gateway sends a re-INVITE to the terminating gateway containing one of the following Session Description Protocol (SDP) lines: a connection IP address set to 0.0.0.0 (c=0.0.0.0), or the attribute field set to send only (a=sendonly) or to inactive (a=inactive). When the SIP phone or user-agent client cancels the hold, the originating gateway takes the call off hold by sending a re-INVITE with the attribute field (a=) set to sendrec or with the connection field (c=) set back to the actual IP address of the remote SIP entity, in place of 0.0.0.0.

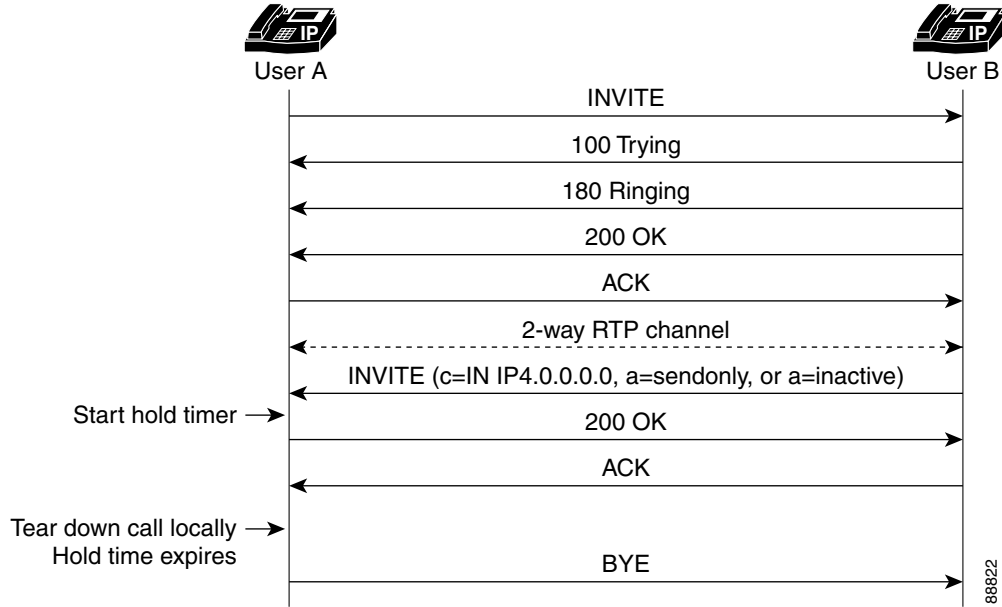
The following call flows show gateway behavior upon receiving a call hold request from a SIP endpoint. In **Figure 93**, the originating gateway sends an INVITE with an indication to place a call on hold (c=IN IP4.0.0.0, a=sendonly, or a=inactive in the SDP), which starts the hold timer. When the gateway on hold receives a re-INVITE with the indication to resume the call (c=IN IP4 User A or a=sendrecv), it stops the hold timer, sends a 200 OK, and resumes the call.

Figure 93 Start and Stop Hold Time



In [Figure 94](#), the hold timer expires, the gateway on hold tears down the call and sends a BYE request to the other end.

Figure 94 Hold Timer Expiration



SIP Media Inactivity Timer

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

When RTCP reports are not received by a Cisco gateway, the SIP Media Inactivity Timer feature releases the hung session and its network resources in an orderly manner. These network resources include the gateway digital signal processor (DSP) and time-division multiplexing (TDM) channel resources that are utilized by the hung sessions. Because call signaling is sent to tear down the call, any stateful SIP proxies involved in the call are also notified to clear the state that they have associated with the hung session. The call is also cleared back through the TDM port so that any attached TDM switching equipment also clears its resources.

Feature benefits include the following:

- Provides a mechanism for detecting and freeing hung network resources when no RTCP packets are received by the gateway.

How to Configure SIP QoS Features

This section contains the following procedures:

- [Configuring Enhanced Codec Support for SIP Using Dynamic Payloads, page 18](#)
- [Configuring Measurement-Based Call Admission Control for SIP, page 20](#)
- [Configuring SIP Gateway Support of RSVP and TEL URL, page 27](#)
- [Reenabling SIP Hold Timer Support, page 39](#)
- [Configuring the SIP Media Inactivity Timer, page 40](#)
- [Verifying SIP QoS Features, page 42](#)
- [Troubleshooting Tips, page 48](#)

**Note**

-
- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP QoS” section on page 2](#)
 - [“Restrictions for SIP QoS” section on page 4](#)
 - For help with a procedure, see the verification and troubleshooting sections listed above.
-

Configuring Enhanced Codec Support for SIP Using Dynamic Payloads

To configure enhanced codec support for SIP using dynamic payloads, perform the following steps.

**Note**

This procedure is optional and selects a dynamic payload value from the IANA defined range of 96 to 127.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voip**
4. **rtp payload-type**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p>enable</p> <p>Example: Router> Enable</p>	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<p>configure terminal</p> <p>Example: Router# configure terminal</p>	Enters global configuration mode.
Step 3	<p>dial-peer voip <i>number</i></p> <p>Example: Router(config)# dial-peer voip 110</p>	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	<p>rtp payload-type <i>type number</i></p> <p>Example: Router(config-dial-peer)# rtp payload-type nte 125</p>	<p>Identifies the payload type of a RTP packet. Arguments are as follows:</p> <ul style="list-style-type: none"> • <i>type number</i>—Payload type. Valid values are the following: <ul style="list-style-type: none"> – cisco-cas-payload—Cisco CAS RTP payload – cisco-clear-channel—Cisco clear-channel RTP payload – cisco-codec-fax-ack—Cisco codec fax acknowledge – cisco-codec-fax-ind—Cisco codec fax indication – cisco-fax relay—Cisco fax relay – cisco-pcm-switch-over-alaw—Cisco RTP PCM codec switch over indication (a-law) – cisco-pcm-switch-over-ulaw—Cisco RTP PCM codec switch over indication (u-law) – cisco-rtp-dtmf-relay—Cisco RTP DTMF relay – nte—Named telephone event – nse—Named signaling event • <i>number</i>—Payload identity. Range: 96 to 127. Default: 101.
Step 5	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	Exits the current mode.

Configuring Measurement-Based Call Admission Control for SIP

This section includes the following procedures:

- [Configure SAA Responder, page 20](#) (required)
- [Configure PSTN Fallback, page 21](#) (required)
- [Configure Resource-Availability Check, page 23](#) (required)
- [Configure SIP Reliable Provisional Response, page 26](#) (required)

Configure SAA Responder

To configure the SAA responder, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **rtr responder**
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 1	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 1	rtr responder Example: Router(config)# rtr responder	Enables SAA Responder functionality on a device.
Step 2	exit Example: Router(config)# exit	Exits the current mode.

Configure PSTN Fallback

To configure PSTN fallback, perform the following steps.



Note

- PSTN fallback configuration applies to both inbound and outbound gateways. In most networks, gateways generate calls to each other, so that every gateway is both an outgoing gateway and a terminating gateway.
- Configure the destination node, which is often but not necessarily the terminating gateway, with the SAA Responder feature.
- PSTN fallback configuration is done at the global level and therefore applies to all calls attempted by the gateway. You cannot selectively apply PSTN fallback only to calls initiated by specific PSTN or PBX interfaces.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback active**
4. **call fallback cache size**
5. **call fallback instantaneous-value-weight**
6. **call fallback jitter-probe num-packets**
7. **call fallback jitter-probe precedence**
8. **call fallback jitter-probe priority-queue**
9. **call fallback threshold delay**
10. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	call fallback active Example: Router(config) # call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.

	Command or Action	Purpose
Step 4	<p>call fallback cache size <i>number</i></p> <p>Example: Router(config) # call fallback cache size 128</p>	<p>Specifies the call-fallback cache size for network traffic probe entries. The argument is as follows:</p> <ul style="list-style-type: none"> <i>number</i>—Cache size, in number of entries. Range: 1 to 256. Default: 128.
Step 5	<p>call fallback instantaneous-value-weight <i>weight</i></p> <p>Example: Router(config) # call fallback instantaneous-value-weight 50</p>	<p>Configures the call-fallback subsystem to determine an average value based on the last two probes registered in the cache for call requests. This command allows the call-fallback subsystem to recover gradually from network congestion conditions. The argument is as follows:</p> <ul style="list-style-type: none"> <i>weight</i>—By percent, when a new probe is received, how much to rely upon the new probe as opposed to the previous cache entry. The configured weight applies to the new probe first. Range: 0 to 100. Default: 66.
Step 6	<p>call fallback jitter-probe num-packets <i>number-of-packets</i></p> <p>Example: Router(config) # call fallback jitter-probe num-packets 15</p>	<p>Specifies the number of packets in a jitter probe used to determine network conditions. The argument is as follows:</p> <ul style="list-style-type: none"> <i>number-of-packets</i>—Number of packets. Range: 2 to 50. Default: 15.
Step 7	<p>call fallback jitter-probe precedence <i>precedence-value</i></p> <p>Example: Router(config) # call fallback jitter-probe precedence 2</p>	<p>Specifies the priority of the jitter-probe transmission by setting the IP precedence of IP packets. The argument is as follows:</p> <ul style="list-style-type: none"> <i>precedence-value</i>—Jitter-probe precedence. Range: 0 to 6. Default: 2.
Step 8	<p>call fallback jitter-probe priority-queue</p> <p>Example: Router(config) # call fallback jitter-probe priority-queue</p>	<p>Assigns a priority queue for jitter-probe transmissions. You must set IP priority queueing for UDP voice ports 16384 to 32767.</p>
Step 9	<p>call fallback threshold delay <i>delay-value</i> loss <i>loss-value</i></p> <p>Example: Router(config) # call fallback threshold delay 36000 loss 50</p>	<p>Configures the call-fallback threshold to use only specified packet delay and loss values. Arguments are as follows:</p> <ul style="list-style-type: none"> <i>delay-value</i>—Delay value, in ms. Range: 1 to 2147483647. No default. <i>loss-value</i>—Loss value, as a percentage. Range: 0 to 100. No default.
Step 10	<p>exit</p> <p>Example: Router(config) # exit</p>	<p>Exits the current mode.</p>

Configure Resource-Availability Check

To enable resource-availability checking, perform one of the following tasks:

- [Configuring Global Resources, page 23](#)
- [Configuring Interface Resources, page 25](#)

Configuring Global Resources

To configure resource availability checking for global resources, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call threshold global**
4. **call treatment**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	call threshold global <i>trigger-name</i> low <i>value</i> high <i>value</i> [busyout treatment] Example: Router(config)# call threshold global total-calls low 5 high 1000 busyout	Enables a trigger and define associated parameters to allow or disallow new calls on the router. Action is enabled when the trigger value exceeds the value specified by the high keyword and is disabled when it drops below the value specified by the low keyword. Keywords and arguments are as follows: <ul style="list-style-type: none"> • <i>trigger-name</i>—Global resources on the gateway to be used as call admission or utilization triggers. Valid values are the following: <ul style="list-style-type: none"> – cpu-5sec—CPU utilization in the last 5 seconds – cpu-avg—Average CPU utilization – io-mem—IO memory utilization – proc-mem—Processor memory utilization – total-calls—Total number of calls – total-mem—Total memory utilization • low value—Low threshold. Range: 1 to 100 percent for utilization triggers and 1 to 10000 for total-calls. • high value—High threshold: Range: 1 to 100 percent for utilization triggers and 1 to 10000 for total-calls. • busyout—Busy out the T1 or E1 channels if the resource is not available • treatment—Apply call treatment from the session application if the resource is not available

Command or Action	Purpose
<p>Step 4</p> <pre>call treatment {on action action [value] cause-code cause-code / isdn-reject value}</pre> <p>Example: Router(config)# call treatment action cause-code 17</p>	<p>Specifies how calls should be processed when local resources are unavailable. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • on—Enable call treatment from the default session application • action—Action to be taken when call treatment is triggered. Valid values are as follows: <ul style="list-style-type: none"> – <i>hairpin</i>—Hairpinning action – <i>playmsg</i>—The gateway plays the selected message. The optional <i>value</i> argument specifies the audio file to play in URL format. – <i>reject</i>—The call should be disconnected and the ISDN cause code passed. • cause-code—Reason for disconnection to the caller. Valid values are as follows: <ul style="list-style-type: none"> – <i>busy</i>—Gateway is busy. – <i>no-QoS</i>—Gateway cannot provide quality of service (QOS). – <i>no-resource</i>—Gateway has no resources available. • isdn-reject value—For ISDN interfaces only, the ISDN reject cause code. Range: 34 to 47 (ISDN cause code for rejection).
<p>Step 5</p> <pre>exit</pre> <p>Example: Router(config)# exit</p>	<p>Exits the current mode.</p>

Configuring Interface Resources

To configure resource availability checking for interface resources, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call threshold interface**
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	call threshold interface <i>interface-name</i> <i>interface-number</i> int-calls low <i>value</i> high <i>value</i> Example: Router(config)# call threshold interface ethernet 0 int-calls low 5 high 2500	Specifies threshold values for total numbers of voice calls placed through a particular interface. Use it also to allow or disallow admission for new calls on the router. Keywords and arguments are as follows: <ul style="list-style-type: none"> • <i>interface-name</i>—Interface used in making call admission decisions. Types of interfaces and their numbers depend upon the configured interfaces. • <i>interface-number</i>—Number of calls through the interface that triggers a call admission decision. • int-calls—Use the number of calls through the interface as a threshold. • low value—Value of low threshold, in percent. Range: 1 to 100 for the utilization triggers and 1 to 10000 calls for int-calls. • high value—Value of high threshold, in percent. Range: 1 to 100 for the utilization triggers and 1 to 10000 calls for int-calls.
Step 4	exit Example: Router(config)# exit	Exits the current mode.

Configure SIP Reliable Provisional Response

To configure SIP reliable provisional response, perform the following steps.



Note

By default, gateways support reliable provisional responses. That is, no additional configuration tasks are necessary to enable reliable provisional responses. This task enables reliable provisional response if it was disabled using the **no rel1xx** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **rel1xx**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service configuration mode.
Step 4	sip Example: Router (config-voi-serv)# sip	Enters SIP configuration mode.

	Command or Action	Purpose
Step 5	<pre>relixx {supported value require value disable}</pre> <p>Example:</p> <pre>Router(config-srv-sip)# relixx supported 100rel</pre> <pre>Router(config-srv-sip)# relixx require 100rel</pre> <pre>Router(config-srv-sip)# relixx disable</pre>	<p>Enables all SIP provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint.</p> <ul style="list-style-type: none"> • supported value—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it the same. Default value is supported with the 100rel value. • require value—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the UAC and UAS configure it the same. • disable—Disables the use of reliable provisional responses.
Step 6	<pre>exit</pre> <p>Example:</p> <pre>Router(config-srv-sip)# exit</pre>	<p>Exits the current mode.</p>

Configuring SIP Gateway Support of RSVP and TEL URL

This section contains the following procedures (you must perform them in the order listed):

- [Configure SIP Gateway Support of RSVP, page 27](#) (required)
- [Configure SIP Gateway Support of TEL URL, page 30](#) (required)
- [Configure Reliability of SIP Provisional Responses, page 33](#) (optional)

Configure SIP Gateway Support of RSVP

This section contains the following procedures:

- [Configuring Fair Queuing and RSVP, page 27](#) (required)
- [Configuring QoS Levels, page 29](#) (required)

Configuring Fair Queuing and RSVP

To configure fair queuing and RSVP, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface fastethernet**
4. **ip rsvp bandwidth**
5. **fair-queue**
6. **exit**

DETAILED STEPS

**Note**

For details on these commands, see the [Cisco IOS Quality of Service Solutions Command Reference](#), Release 12.3. For an example, see the “SIP Gateway Support of RSVP and TEL URL: Example” section on page 51.

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	interface fastethernet <i>number</i> Example: Router(conf)# interface fastethernet 1	Selects a particular fast ethernet interface for configuration. The argument is as follows: <ul style="list-style-type: none"> <i>number</i>—Port, connector, or interface card number. On a Cisco 4500 or Cisco 4700 series, the network-interface module or network-processor-module number. Numbers are assigned at the factory at the time of installation or when added to a system.
Step 4	ip rsvp bandwidth [<i>interface-kbps</i>] [<i>single-flow-kbps</i>] Example: Router(conf-if)# ip rsvp bandwidth 100 100	Enables resource reservation protocol for IP on an interface. Arguments are as follows: <ul style="list-style-type: none"> <i>interface-kbps</i>—Maximum amount of bandwidth, in kbps, that may be allotted by RSVP flows. Range: 1 to 10000000. <i>single-flow-kbps</i>—Maximum amount of bandwidth, in kbps, that may be allocated in a single flow. Range: 1 to 10000000.

Command or Action	Purpose
<p>Step 5</p> <p>fair-queue [<i>congestive-discard-threshold</i>] [<i>dynamic-queues</i> [<i>reservable-queues</i>]]</p> <p>Example: Router(config-if)# fair-queue 32 16 100</p>	<p>Enables weighted fair queuing for an interface. Arguments are as follows:</p> <ul style="list-style-type: none"> • <i>congestive-discard-threshold</i>—Number of messages allowed in each queue. When a conversation reaches this threshold, new message packets are discarded. Valid values: powers of 2 in the range from 16 to 4096. Default: 64. • <i>dynamic-queues</i>—Number of dynamic queues used for best-effort conversations (that is, a normal conversation not requiring any special network services). Valid values: 16, 32, 64, 128, 256, 512, 1024, 2048, and 4096. See tables in the fair-queue (class-default) command for the default number of dynamic queues. • <i>reservable-queues</i>—Number of reservable queues used for reserved conversations. Reservable queues are used for interfaces configured for features such as RSVP. Range: 0 to 1000. Default: 0.
<p>Step 6</p> <p>exit</p> <p>Example: Router(config-if)# exit</p>	<p>Exits the current mode.</p>

Configuring QoS Levels

To configure desired and acceptable QoS levels, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **acc-qos**
5. **req-qos**
6. **exit**

DETAILED STEPS



Note

For details on these commands, see the [Cisco IOS Voice Command Reference](#), Release 12.3. For an example, see “[SIP Gateway Support of RSVP and TEL URL: Example](#)” section on page 51.

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 10 voip	Enter VoIP dial-peer configuration modes for the specified VoIP dial peer.
Step 4	acc-qos {best-effort controlled-load guaranteed-delay} Example: Router(config dial-peer)# acc-qos best-effort	Defines the acceptable QoS for any inbound and outbound call on a VoIP dial peer. Keywords are as follows: <ul style="list-style-type: none"> • best-effort—RSVP makes no bandwidth reservation. This is the default. • controlled-load—RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to ensure that preferential service is received even when the bandwidth is overloaded. • guaranteed-delay—RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.
Step 5	req-qos {best-effort controlled-load guaranteed-delay} Example: Router(config dial-peer)# req-qos best-effort	Specifies the desired QoS to be used in reaching a specific dial peer. Keywords are as above.
Step 6	exit Example: Router(config dial-peer)# exit	Exits the current mode.

Configure SIP Gateway Support of TEL URL

This section contains the following procedures:

- [Configuring TEL URLs for All VoIP SIP Calls, page 31](#) (optional)
- [Configuring TEL URLs for All Dial-Peer SIP Calls, page 32](#) (optional)

Configuring TEL URLs for All VoIP SIP Calls

To configure TEL URL format for all VoIP SIP calls, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **url**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Specifies the voice encapsulation type.
Step 4	sip Example: Router(config-voi-srv)# sip	Enters SIP configuration mode.
Step 5	url {sip tel} Example: Router(conf-serv-sip)# url sip	Configures URLs to either the SIP or TEL format for your VoIP SIP calls. Keywords are as follows: <ul style="list-style-type: none"> • sip—Generate URLs in SIP format for VoIP calls. This is the default. • tel—Generate URLs in TEL format for VoIP calls.
Step 6	exit Example: Router(conf-serv-sip)# exit	Exits the current mode.

Configuring TEL URLs for All Dial-Peer SIP Calls

To configure TEL URLs for all dial-peer SIP calls, perform the following steps.



Note

The **voice-class sip url** command in dial-peer configuration mode takes precedence over the **url** command in **global configuration**. However, if the **voice-class sip url** command contains the configuration of **system**, the gateway uses what was globally configured under the **url** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **voice-class sip url**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 4	voice-class sip url {sip sips system tel} Example: Router(config-dial-peer)# voice-class sip url sip	Configures URLs to either the SIP or TEL format for your dial-peer SIP call. Keywords are as follows: <ul style="list-style-type: none"> • sip—Generate URLs in the SIP-format for calls on a dial-peer basis. • sips—Generate URLs in the SIPS-format for calls on a dial-peer basis. • tel—Generate URLs in the TEL format for calls on a dial-peer basis. • system—Use the system value. This is the default.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configure Reliability of SIP Provisional Responses

The following are tasks for configuring reliability of SIP provisional responses:

- [Configuring Specific Reliable Provisional Responses, page 34](#) (optional)
- [Configuring Global Reliable Provisional Responses, page 35](#) (required when using RSVP)
- [Configuring PRACK Timers and Retries, page 36](#) (optional)
- [Configuring COMET Timers and Retries, page 37](#) (optional)
- [Configuring Reliable-Provisional-Response Timers and Retries, page 38](#) (optional)

By default, gateways support reliable provisional responses. That is, no additional configuration tasks are necessary to enable reliable provisional responses.

However, there are instances when you may want control over the use of reliable provisional responses. For example, you may want to:

- Always require the use of reliable provisional responses (use the Required header)
- Never use reliable provisional responses

In these cases, there are two ways to configure reliable provisional responses:

- Dial-peer mode. In this mode you can configure reliable provisional responses for the specific dial peer only. Configure with the **voice-class sip rel1xx** command.
- SIP mode. In this mode you can configure reliable provisional responses globally. Configure with the **rel1xx** command.

When the **voice-class sip rel1xx** command under dial-peer configuration is configured, it takes precedence over **the global configuration of the rel1xx** command. However, if the **voice-class sip rel1xx** command contains the configuration of **system**, the gateway uses what was globally configured under the **rel1xx** command.

[Table 61](#) shows the possible configurations achieved with the **voice-class sip rel1xx** and the **rel1xx** commands. It outlines the possible configurations on both the originating gateway and the terminating gateway, and the results of the various configurations.



Note

When configured with the **supported** option, the SIP gateway uses the Supported header in outgoing INVITE messages. When configured with the **require** option, the SIP gateway uses the Required header in outgoing INVITE messages.

Table 61 Configuration Results Based on Originating and Terminating Gateway Configurations

Originating Gateway	Terminating Gateway	Result
supported 100rel	supported 100rel	Reliable provisional responses
supported 100rel	require 100rel	Reliable provisional responses
supported 100rel	disable	No reliable provisional responses, call proceeds
require 100rel	supported 100rel	Reliable provisional responses
require 100rel	require 100rel	Reliable provisional responses
require 100rel	disable	Call fails. TG sends 420 with “Unsupported: 100rel” header
disable	supported 100rel	No reliable provisional responses

Table 61 Configuration Results Based on Originating and Terminating Gateway Configurations (continued)

Originating Gateway	Terminating Gateway	Result
disable	require 100rel	No reliable provisional responses
disable	disable	No reliable provisional responses

Configuring Specific Reliable Provisional Responses

To configure specific reliable provisional responses, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice voip**
4. **voice-class sip rel1xx**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Router(config)# dial-peer voice 29 voip	Enter dial-peer configuration mode for the specified VoIP dial peer.

	Command or Action	Purpose
Step 4	<p>voice-class sip rel1xx {supported <i>value</i> require <i>value</i> system disable}</p> <p>Example: Router(config-dial-peer)# voice-class sip rel1xx supported 100rel</p>	<p>Enables all SIP provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • supported <i>value</i>—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it as the same. • required <i>value</i>—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the UAC and UAS configure it the same. • system—Use the value configured in voice service mode. Default is the system value. • disable—Disable the use of rel1xx provisional responses.
Step 5	<p>exit</p> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits the current mode.</p>

Configuring Global Reliable Provisional Responses

To configure global reliable provisional responses, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **rel1x**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.
Step 3	<code>voice service voip</code> Example: Router(config)# voice service voip	Enters voice-service configuration mode for VoIP.
Step 4	<code>sip</code> Example: Router(config-voi-srv)# sip	Enters SIP configuration mode.
Step 5	<code>rellxx {supported value require value disable}</code> Example: Router(config-srv-sip)# rellxx supported 100rel	Enables all SIP provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint. <ul style="list-style-type: none"> • supported value—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it the same. The default value is supported with the 100rel value. • require value—Use provisional responses and you set the value; for instance, 100rel. The <i>value</i> argument may have any value, as long as both the UAC and UAS configure it the same. • disable—Disables the use of reliable provisional responses.
Step 6	<code>exit</code> Example: Router(config-srv-sip)# exit	Exits the current mode.

Configuring PRACK Timers and Retries

To configure PRACK timers and retries, perform the following steps.

SUMMARY STEPS

1. `enable`

2. **configure terminal**
3. **sip-ua**
4. **timers prack**
5. **retry prack**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers prack <i>number</i> Example: Router(config-sip-ua)# timers prack 500	Sets the amount of time that the user agent waits before retransmitting PRACK requests. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Time (in ms) to wait before retransmitting. Range: 100 to 1000. Default: 500.
Step 5	retry prack <i>number</i> Example: Router(config-sip-ua)# retry prack 9	Sets the number of times that the PRACK request is retransmitted to the other user agent. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Number of retries. Range: 1 to 10. Default: 10.
Step 6	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring COMET Timers and Retries

To configure COMET timers and retries, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**

4. **timers comet**
5. **retry comet**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	timers comet <i>number</i> Example: Router(config-sip-ua)# timers comet 100	Sets the amount of time that the user agent waits before retransmitting COMET requests. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Time (in ms) to wait before retransmitting. Range: 100 to 1000. Default: 500.
Step 5	retry comet <i>number</i> Example: Router(config-sip-ua)# retry comet 10	Sets the number of times that a COMET request is retransmitted to the other user agent. The argument is as follows: <ul style="list-style-type: none"> • <i>number</i>—Number of retries. Range: 1 to 10. Default: 10.
Step 6	exit Example: Router(config-sip-ua)# exit	Exits the current mode.

Configuring Reliable-Provisional-Response Timers and Retries

To configure reliable-provisional-response timers and retries, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **timers rel1xx**
5. **retry rel1xx**

6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: Router# configure terminal	Enters global configuration mode.
Step 3	<code>sip-ua</code> Example: Router(config)# sip-ua	Enters SIP user-agent mode.
Step 4	<code>timers rellxx number</code> Example: Router(config-sip-ua)# timers rellxx 500	Sets the amount of time that the user agent waits before retransmitting the reliable 1xx responses. The argument is as follows: <ul style="list-style-type: none"> <i>number</i>—Time (in ms) to wait before retransmitting. Range: 100 to 1000. Default: 500.
Step 5	<code>retry rellxx number</code> Example: Router(config-sip-ua)# retries rellxx 10	Sets the number of times the reliable 1xx response is retransmitted to the other user agent. The argument is as follows: <ul style="list-style-type: none"> <i>number</i>—Number of retries. Range: 1 to 10. Default: 6.
Step 6	<code>exit</code> Example: Router(config-sip-ua)# exit	Exits the current mode.

Reenabling SIP Hold Timer Support

To configure SIP hold timer support, perform the following steps.



Note

The SIP: Hold Timer Support feature is enabled by default; no configuration tasks are required to enable this feature. This task enables the feature again if it was disabled by using the **no timers hold** command.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `timers hold`

4. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	timers hold time Example: Router(config)# timers hold 120	Enables the SIP hold timer and sets the timer interval. The argument is as follows: <ul style="list-style-type: none"> <i>time</i>—Time, in minutes, before the gateway disconnects held calls. Range: 15 to 2880. Default: 2880.
Step 4	exit Example: Router(config)# exit	Exits the current mode.

Configuring the SIP Media Inactivity Timer

The SIP Media Inactivity Timer feature requires configuration of the **ip rtcp report interval** command and the **timer receive-rtcp** command to enable detection of RTCP packets by the gateway. When these commands are configured, the gateway uses RTCP report detection, rather than Real-Time Protocol (RTP) packet detection, to determine whether calls on the gateway are still active or should be disconnected. This method is more reliable because there are periods during voice calls when one or both parties are not sending RTP packets.

One common example of a voice session in which no RTP is sent is when a caller dials into a conference call and mutes his endpoint. If voice activity detection (VAD, also known as silence suppression) is enabled, no RTP packets are sent while the endpoint is muted. However, the muted endpoint continues to send RTCP reports at the interval specified by the **ip rtcp report interval** command.

The **timer receive-rtcp value** argument (or Mfactor) is multiplied with the interval that is set using the **ip rtcp report interval** command. If no RTCP packets are received in the resulting time period, the call is disconnected. The gateway signals the disconnect to the SIP network and the TDM network so that upstream and downstream devices can clear their resources. The gateway sends a SIP BYE to disconnect the call and sends a Q.931 DISCONNECT back to the TDM network to clear the call upon the expiration of the timer. The Q.931 DISCONNECT is sent with a Cause code value of 3 (no route). There is no Q.931 Progress Indicator (PI) value included in the DISCONNECT.

SUMMARY STEPS

1. enable

2. **configure terminal**
3. **gateway**
4. **timer receive-rtcp**
5. **exit**
6. **ip rtcp report interval**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> Enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	gateway Example: Router(config)# gateway	Enables the H.323 VoIP gateway.
Step 4	timer receive-rtcp timer Example: Router(config-gateway)# timer receive-rtcp 100	Enables the Real-Time Control Protocol (RTCP) timer and to configure a multiplication factor for the RTCP timer interval for the SIP. The argument is as follows: <ul style="list-style-type: none"> • <i>timer</i>—Multiples of the RTCP report transmission interval. Range: 2 to 1000. Default: 5.
Step 5	exit Example: Router(config-gateway)# exit	Exits the current mode.

	Command or Action	Purpose
Step 6	<pre>ip rtcp report interval value</pre> <p>Example: Router(config)# ip rtcp report interval 500</p>	<p>Sets the average reporting interval between subsequent RTCP report transmissions. The argument is as follows:</p> <ul style="list-style-type: none"> <i>value</i>—Average interval (in ms) for RTCP report transmissions. Range: 1 to 65535. Default: 5000. <p>Note RFC 1889, <i>RTP: A Transport Protocol for Real-Time Applications</i>, recommends a minimum 5-second average reporting interval between successive RTCP reports. It also recommends that this interval be varied randomly. The randomization function is performed automatically and cannot be disabled. Therefore, the reporting interval does not remain constant throughout a given voice session, but its average is the specified reporting interval.</p>
Step 7	<pre>exit</pre> <p>Example: Router(config)# exit</p>	Exits the current mode.

Verifying SIP QoS Features

To verify configuration of SIP QoS features, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. `show call fallback cache`
2. `show call fallback config`
3. `show call fallback stats`
4. `show call rsvp-sync conf`
5. `show call rsvp-sync stats`
6. `show dial-peer voice`
7. `show ip rsvp reservation`
8. `show running-conf`
9. `show sip-ua retry`
10. `show sip-ua statistics`
11. `show sip-ua status`
12. `show sip-ua timers`
13. `test call fallback probe`

DETAILED STEPS

Step 1 `show call fallback cache`

Use this command to display the current ICPIF estimates for all IP addresses in the cache.

Step 2 **show call fallback config**

Use this command to display the call fallback configuration.

Step 3 **show call fallback stats**

Use this command to display call fallback statistics.

Step 4 **show call rsvp-sync conf**

Use this command to display the configuration settings for RSVP synchronization.

Step 5 **show call rsvp-sync stats**

Use this command to display the configuration settings for RSVP synchronization.

```
Router# show call rsvp-sync

conf Show RSVP/Voice Synchronization Config. information
state Show RSVP/Voice Statistics
```

The following sample output also shows configuration settings for RSVP synchronization. Of particular note in the example are the following:

- Overture Synchronization is ON—Indicates that RSVP synchronization is enabled.
- Reservation Timer is set to 10 seconds—Number of seconds for which the RSVP reservation timer is configured.

```
Router# show call rsvp-sync conf

VoIP QoS:RSVP/Voice Signaling Synchronization config:
Overture Synchronization is ON
Reservation Timer is set to 10 seconds
```

The following sample output shows configuration settings for RSVP synchronization. Of particular note in the example are the following:

- Number of calls for which QoS was initiated—Number of calls for which RSVP setup was attempted.
- Number of calls for which QoS was torn down—Number of calls for which an established RSVP reservation was released.
- Number of calls for which Reservation Success was notified—Number of calls for which an RSVP reservation was successfully established.
- Total Number of PATH Errors encountered—Number of path errors that occurred.
- Total Number of RESV Errors encountered—Number of reservation errors that occurred.
- Total Number of Reservation Timeouts encountered—Number of calls in which the reservation setup was not complete before the reservation timer expires.

```
Router# show call rsvp-sync stats

VoIP QoS:Statistics Information:
Number of calls for which QoS was initiated : 0
Number of calls for which QoS was torn down : 0
Number of calls for which Reservation Success was notified : 0
Total Number of PATH Errors encountered : 0
Total Number of RESV Errors encountered : 0
Total Number of Reservation Timeouts encountered : 0
```

Step 6 **show dial-peer voice**

Use this command to display detailed information for a specific voice dial peer.

```
Router# show dial-peer voice 5
```

```
VoiceOverIpPeer5
  information type = voice,
  tag = 5, destination-pattern = `5550100',
  answer-address = `', preference=0,
  numbering Type = `unknown'
  group = 5, Admin state is up, Operation state is up,
  incoming called-number = `', connections/maximum = 0/unlimited,
  DTMF Relay = disabled,
  modem passthrough = system,
  huntstop = disabled,
  in bound application associated:session
  out bound application associated
  dnis-map =
  permission :both
  incoming COR list:maximum capability
  outgoing COR list:minimum requirement
  type = voip, session-target = `ipv4:172.18.192.218',
  technology prefix:
  settle-call = disabled
  ip media DSCP = default, ip signaling DSCP = default, UDP checksum = disabled,
  session-protocol = sipv2, session-transport = system, req-qos = best-effort,
  acc-qos = best-effort,
  fax rate = voice,   payload size = 20 bytes
  fax protocol = system
  fax NSF = 0xAD0051 (default)
  codec = g711ulaw,   payload size = 160 bytes,
  Expect factor = 0, Icpif = 20,
  Playout Mode is set to default,
  Initial 60 ms, Max 300 ms
  Playout-delay Minimum mode is set to default, value 40 ms
  Expect factor = 0,
Max Redirects = 1, Icpif = 20,signaling-type = cas,
CLID Restrict = disabled
VAD = enabled, Poor QOV Trap = disabled,
voice class sip url = system,
voice class sip rellxx = system,
voice class perm tag = ``
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
```

Step 7 **show ip rsvp reservation**

Use this command to display RSVP-related receiver information currently in the database.

The following sample output shows, in the “To” field, the IP address of the receiver.

```
Router # show ip rsvp reservation
```

```
To          From          Pro DPort Sport Next Hop      I/F  Fi Serv BPS Bytes
172.18.193.101 172.18.193.102 UDP 20532 20600                FF LOAD 24K 120
172.18.193.102 172.18.193.101 UDP 20600 20532 172.18.193.102 Et0/0 FF LOAD 24K 120
```

Step 8 **show running-conf**

Use this command to display the contents of the currently running configuration file, the configuration for a specific interface, or map class information. Use it to display SIP user-agent statistics, including reliable provisional response information. Use it also to display configuration for the SIP Extensions for Caller Identity and Privacy feature.

The following sample output shows SIP user-agent statistics, including reliable provisional response information. In the following partial output, a dynamic payload value of 115 is configured, freeing up the reserved value of 101.

```
Router# show running-config

Building configuration...
Current configuration: 2024 bytes
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
hostname r4
ip subnet-zero
ip tcp synwait-time 5
no ip domain-lookup
ipx routing 0000.0000.0004
no voice hpi capture buffer
no voice hpi capture destination
fax interface-type fax-mail
mta receive maximum-recipients 0
interface Loopback0
 ip address 10.0.0.0 255.255.255.0
interface FastEthernet0/0
 ip address 10.0.0.1 255.255.255.0
 speed 100
 full-duplex
interface Serial0/0
 ip address 10.0.0.4 255.255.255.0
 encapsulation frame-relay
.
.
.
call rsvp-sync
voice-port 3/0/0
voice-port 3/0/1
mgcp ip qos dscp cs5 media
mgcp ip qos dscp cs3 signaling
no mgcp timer receive-rtcp
mgcp profile default
dial-peer cor custom
dial-peer voice 1234 voip
 rtp payload-type nte 115
alias exec co config t
alias exec br show ip int brief
alias exec i show ip route
alias exec sr show run
alias exec sri sh run interface
alias exec sio show ip ospf
alias exec sioi show ip ospf int
alias exec sion show ip ospf nei
alias exec cir clear ip route *
alias exec ix show ipx route
alias exec b show ip bgp
alias exec sis show isdn status
alias exec fm show frame map
alias exec dm show dialer map
line con 0
```

```

exec-timeout 0 0
privilege level 15
password password1
logging synchronous
line aux 0
line vty 0 4
exec-timeout 0 0
privilege level 15
password password1
logging synchronous
no login
end

```

The following sample output shows configuration for the SIP Extensions for Caller Identity and Privacy feature. If the SIP hold timer is enabled, which is the default setting, and the timer is set to the default value of 2880 minutes, command output does not display the **timers hold 2880** command. In the following partial output, the hold timer is set to a nondefault value of 18 minutes.

```

Router# show running-config

Building configuration...
Current configuration :2791 bytes
.
.
.
sip-ua
  max-forwards 10
  retry invite 1
  retry response 4
  retry bye 1
  retry cancel 1
  timers expires 300000
  timers hold 18
.
.
.
end!

```

Step 9 show sip-ua retry

Use this command to display SIP retry statistics.

```

Router# show sip-ua retry

SIP UA Retry Values
invite retry count = 10  response retry count = 1
bye retry count     = 1   cancel retry count   = 8
prack retry count  = 10  comet retry count   = 10
reliable lxx count = 6

```

Step 10 show sip-ua statistics

Use this command to display response, traffic, and retry SIP statistics.



Note When 0/0 is included in a field, the first number is an inbound count and the last number is an outbound count.

```

Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
  Trying 15/9, Ringing 9/0,

```

```

    Forwarded 0/0, Queued 0/0,
    SessionProgress 36/9
Success:
    OkInvite 6/4, OkBye 5/5,
    OkCancel 5/5, OkOptions 0/0,
    OkPrack 29/8, OkPreconditionMet 11/0
Redirection (Inbound only):
    MultipleChoice 0, MovedPermanently 0,
    MovedTemporarily 0, SeeOther 0,
    UseProxy 0, AlternateService 0
Client Error:
    BadRequest 0/0, Unauthorized 0/0,
    PaymentRequired 0/0, Forbidden 0/0,
    NotFound 0/0, MethodNotAllowed 0/0,
    NotAcceptable 0/0, ProxyAuthReqd 0/0,
    ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
    LengthRequired 0/0, ReqEntityTooLarge 0/0,
    ReqURITooLarge 0/0, UnsupportedMediaType 0/0,
    BadExtension 0/0, TempNotAvailable 0/0,
    CallLegNonExistent 0/0, LoopDetected 0/0,
    TooManyHops 0/0, AddrIncomplete 0/0,
    Ambiguous 0/0, BusyHere 0/0,
    RequestCancel 0/0, NotAcceptableMedia 0/0
Server Error:
    InternalError 0/0, NotImplemented 0/0,
    BadGateway 0/0, ServiceUnavail 0/0,
    GatewayTimeout 0/0, BadSipVer 0/0,
    PreCondFailure 0/0
Global Failure:
    BusyEverywhere 0/0, Decline 0/0,
    NotExistAnywhere 0/0, NotAcceptable 0/0

SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 9/16, Ack 4/6, Bye 5/5,
    Cancel 5/9, Options 0/0,
    Prack 8/43, Comet 0/11

Retry Statistics
    Invite 5, Bye 0, Cancel 4, Response 0,
    Prack 13, Comet 0, Reliable1xx 0

```

Step 11 show sip-ua status

Use this command to display SIP user-agent status.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)

```

Step 12 show sip-ua timers

Use this command to display SIP user-agent timer settings.

```

Router# show sip-ua timers

SIP UA Timer Values (millisecs unless noted)
trying 500, expires 150000, connect 500, disconnect 500
comet 500, prack 500, rellxx 500, notify 500, hold 2880 minutes

```

Step 13 test call fallback probe *ip-address codec*

Use this command to test a probe to a specific IP address and display ICPIF RTR values. Keywords and arguments are as follows:

- *ip-address*—Target IP address.
- *codec*—Codec type to test. Valid values are 711 (G.711 codec) and 729 (G.729 codec).

Troubleshooting Tips

**Note**

For general troubleshooting tips and a list of important **debug** commands, see the “[General Troubleshooting Tips](#)” section on page 18.

- Make sure that you can make a voice call.
- Make sure VoIP is working before call fallback is configured.
- Use the **debug ccsip events** command, which includes output specific to the SIP Media Inactivity Timer feature.
- Use the **debug ccsip events** command to show all SIP SPI events tracing.
- Use the **debug call fallback detail** command to display details of the VoIP call fallback.
- Use the **debug ccsip messages** command to enable CCSIP SPI messages debugging trace.
- Use the **debug ccsip error** command to enable SIP error debugging trace.
- Use the **debug ccsip all** command to enable all SIP debugging traces.
- Use the **debug rtr trace** command to trace the execution of an SAA operation.
- Use the **debug call fallback probe** command to verify that probes are being sent correctly.
- Use the **debug ccsip all** command to enable all SIP debugging capabilities or use one of the following SIP debug commands:
 - **debug ccsip calls**
 - **debug ccsip error**
 - **debug ccsip events**
 - **debug ccsip messages**
 - **debug ccsip states**
- When terminating long distance or international calls over ISDN, the terminating switch receives information from the gateway. Generally, the information received consists of the numbering plan and the ISDN number type. As a default, the gateway tags both the numbering plan and number type as *Unknown*. However, this *Unknown* tag may cause interworking issues with some switches.

You can override the default ISDN numbering plan and number type with custom values, using the **isdn map** command. This command sets values on a per-number basis or on numbers that match set patterns. The following example shows an override of any plan or type with a called or calling number that begins with the numeral 1. Thus, the ISDN setup sent to the switch is used only for long distance numbers, the numbering plan is **ISDN**, and the type of number is **National**:

```
isdn map address 1.* plan isdn type* national
```


For more details on the **isdn map** command, see the [Cisco IOS Dial Technologies Command Reference, Release 12.3](#).

- Verify that SIP supported codecs are used. Support for codecs varies on different platforms. For a listing of SIP codec support by platform, see [Table 56 on page 5](#). Use the **codec ?** command to determine the codecs available on a specific platform.

Following is sample output for some of these commands:

- [Sample Output for the debug ccsip events Command, page 49](#)
- [Sample Output for the debug rtr trace Command, page 49](#)
- [Sample Output for the debug call fallback probe Command, page 50](#)

Sample Output for the debug ccsip events Command

The following example trace shows a timer being set:

```
Router# debug ccsip events
```

```
00:04:29: sipSPICreateAndStartRtpTimer: Valid RTP/RTCP session found and CLI enabled to create and start the inactivity timer
00:04:29: sipSPICreateAndStartRtpTimer:Media Inactivity timer created for call.
Mfactor(from CLI): 5 RTCP bandwidth: 500
RTCP Interval(in ms): 5000
Normalized RTCP interval (in ms):25000
```

The following example trace shows a timer expiring:

```
Router# debug ccsip events
```

```
02:41:03: %LINEPROTO-5-UPDOWN: Line protocol on Interface Ethernet0, changed state to down
*Jan 1 02:41:34.107: sipSPIRtpDiscTimerExpired:RTP/RTCP receive timer expired. Disconnect the call.
*Jan 1 02:41:34.107: Queued event From SIP SPI to CCAPI/DNS : SIPSPI_EV_CC_CALL_DISCONNECT
*Jan 1 02:41:34.107: CCSIP-SPI-CONTROL: act_active_disconnect
```



Note The **timer receive-rtcp** command configures a media activity timer that is common to both H.323 and SIP. If set, it affects both H.323 and SIP calls.

Sample Output for the debug rtr trace Command

```
Router# debug rtr trace
```

```
Router#
*Mar 1 00:11:42.439: RTR 1: Starting An Echo Operation - IP RTR Probe 1
*Mar 1 00:11:42.439: rtt hash insert : 10.1.1.63 32117
*Mar 1 00:11:42.439: source=10.1.1.63(32117) dest-ip=10.1.1.67(32057) vrf tableid = 0
*Mar 1 00:11:42.439: sending control enable:
*Mar 1 00:11:42.439: cmd: command: , ip: 10.1.1.67, port: 32057, duration: 1200
*Mar 1 00:11:42.439: sending control msg:
*Mar 1 00:11:42.439: Ver: 1 ID: 20 Len: 52
*Mar 1 00:11:42.443: receiving reply
*Mar 1 00:11:42.443: Ver: 1 ID: 20 Len: 8
*Mar 1 00:11:42.459: sdTime: -1989906017 dsTime: 2076306018
*Mar 1 00:11:42.459: responseTime (1): 1
*Mar 1 00:11:42.479: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.479: jitterOut: 0
*Mar 1 00:11:42.479: jitterIn: -1
*Mar 1 00:11:42.479: responseTime (2): 1
*Mar 1 00:11:42.499: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.499: jitterOut: 0
```

```

*Mar 1 00:11:42.499: jitterIn: 0
*Mar 1 00:11:42.499: responseTime (3): 1
*Mar 1 00:11:42.519: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.519: jitterOut: 0
*Mar 1 00:11:42.519: jitterIn: 0
*Mar 1 00:11:42.519: responseTime (4): 1
*Mar 1 00:11:42.539: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.539: jitterOut: 0
*Mar 1 00:11:42.539: jitterIn: 0
*Mar 1 00:11:42.539: responseTime (5): 1
*Mar 1 00:11:42.559: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.559: jitterOut: 0
*Mar 1 00:11:42.559: jitterIn: 0
*Mar 1 00:11:42.559: responseTime (6): 1
*Mar 1 00:11:42.579: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.579: jitterOut: 0
*Mar 1 00:11:42.579: jitterIn: 0
*Mar 1 00:11:42.579: responseTime (7): 1
*Mar 1 00:11:42.599: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.599: jitterOut: 0
*Mar 1 00:11:42.599: jitterIn: 0
*Mar 1 00:11:42.599: responseTime (8): 1
*Mar 1 00:11:42.619: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.619: jitterOut: 0
*Mar 1 00:11:42.619: jitterIn: 0
*Mar 1 00:11:42.619: responseTime (9): 1
*Mar 1 00:11:42.639: sdTime: -1989906017 dsTime: 2076306017
*Mar 1 00:11:42.639: jitterOut: 0
*Mar 1 00:11:42.639: jitterIn: 0
*Mar 1 00:11:42.639: responseTime (10): 1
*Mar 1 00:11:42.639: rtt hash remove: 10.1.1.63 32117

```

Router# **debug rtr trace**

```

Router#
*Mar 1 00:14:12.439: RTR 1: Starting An Echo Operation - IP RTR Probe 1
*Mar 1 00:14:12.439: rtt hash insert : 10.1.1.63 32117
*Mar 1 00:14:12.439: source=10.1.1.63(32117) dest-ip=10.1.1.67(32057) vrf tableid = 0
*Mar 1 00:14:12.439: sending control enable:
*Mar 1 00:14:12.439: cmd: command: , ip: 10.1.1.67, port: 32057, duration: 1200
*Mar 1 00:14:12.439: sending control msg:
*Mar 1 00:14:12.439: Ver: 1 ID: 27 Len: 52
*Mar 1 00:14:13.439: control message timeout
*Mar 1 00:14:13.439: sending control msg:
*Mar 1 00:14:13.439: Ver: 1 ID: 28 Len: 52
*Mar 1 00:14:14.439: control message timeout
*Mar 1 00:14:14.439: control message failure: 1
*Mar 1 00:14:14.439: rtt hash remove: 10.1.1.63 32117
*Mar 1 00:14:42.439: RTR 1: Starting An Echo Operation - IP RTR Probe 1
*Mar 1 00:14:42.439: rtt hash insert : 10.1.1.63 32117
*Mar 1 00:14:42.439: source=10.1.1.63(32117) dest-ip=10.1.1.67(32057) vrf tableid = 0

```

Sample Output for the debug call fallback probe Command

Router# **debug call fallback probe**

```

Router#
*Mar 1 00:10:12.439: fb_main: Probe timer expired, 10.1.1.67, codec:g711ulaw
*Mar 1 00:10:12.639: fb_main:NumOfRTT=10, RTTSum=10, loss=0, jitter in=0, jitter out=0->
10.1.1.67, codec:g711ulaw, delay = 28
*Mar 1 00:10:12.639: g113_calc_icpif: loss=0, expect_factor=10, delay (w/codec
delay)=28, Icpif=0
*Mar 1 00:10:12.639: fb_main: New smoothed values: inst_weight=100, ICPIF=0, Delay=28,
Loss=0 -> 10.1.1.67, codec:g711ulaw

```

```

3640SDP#
r 1 00:13:12.439: fb_main: Probe timer expired, 10.1.1.67, codec:g711ulaw
*Mar 1 00:13:14.439: %FALLBACK-3-PROBE_FAILURE: A probe error to 10.1.1.67 occured -
control message failure
*Mar 1 00:13:14.439: fb_main:NumOfRRT=0, RTTSum=0, loss=100, jitter in=0, jitter out=0->
10.1.1.67, codec:g711ulaw, delay is N/A (since loss is 100 percent)
*Mar 1 00:13:14.439: g113_calc_icpif: loss=100, expect_factor=10, delay is N/A (since
loss is 100 percent), Icpif=64
*Mar 1 00:13:14.439: fb_main: New unsmoothed values: inst_weight=100, ICPIF=64,
Delay=N/A, Loss=100 -> 10.1.1.67, codec:g711ulaw
3/0:23(1) is in busyout state

*Mar 1 00:13:22.435: %LINK-3-UPDOWN: Interface ISDN-VOICE 3/0:23(1), changed state to
Administrative Shutdown
*Mar 1 00:13:22.439: %ISDN-6-LAYER2DOWN: Layer 2 for Interface Se3/0:23, TEI 0 changed to
down

```

Configuration Examples for SIP QoS Features

This section provides the following configuration examples:

- [SIP Gateway Support of RSVP and TEL URL: Example, page 51](#)
- [SIP Media Inactivity Timer: Example, page 52](#)

SIP Gateway Support of RSVP and TEL URL: Example

This configuration example shows RSVP for SIP calls on gateways being enabled. Gateway A is the originating gateway and Gateway B is the terminating gateway:

```

GATEWAY A
-----

Router# show running-config
.
.
.
interface Ethernet0/0
 ip address 172.18.193.101 255.255.255.0
 fair-queue 64 256 235
 ip rsvp bandwidth 7500 7500
!
voice-port 1/0/0
!
dial-peer voice 1 pots
 destination-pattern 111
 port 1/0/0
!
dial-peer voice 2 voip
 incoming called-number 111
 destination-pattern 222
 session protocol sipv2
 session target ipv4:172.18.193.102
 req-qos controlled-load
!
GATEWAY B
-----
!
interface Ethernet0/0

```

```

ip address 172.18.193.102 255.255.255.0
fair-queue 64 256 235
ip rsvp bandwidth 7500 7500
!
voice-port 1/0/1
!
dial-peer voice 1 pots
destination-pattern 222
port 1/0/1
!
dial-peer voice 2 voip
incoming called-number 222
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.101
req-qos controlled-load
!

```

SIP Media Inactivity Timer: Example

```

Router# show running-config

!
version 12.3
no parser cache
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
!
hostname madison
boot system flash
no logging buffered
aaa new-model
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection password stop-only group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
!
resource-pool disable
clock timezone EST -5
!
ip subnet-zero
ip tcp path-mtu-discovery
ip name-server 172.18.192.48
ip dhcp smart-relay
!
isdn switch-type primary-ni
!
voice service voip
h323
!
voice class codec 1
codec preference 1 g723ar53
codec preference 2 g723r53
codec preference 3 g729br8
codec preference 4 gsmfr
codec preference 5 g726r24
codec preference 6 g726r32
voice class codec 2
codec preference 1 g729br8

```

```
    codec preference 2 g729r8
    codec preference 3 g723ar53
    codec preference 4 g723ar63
    codec preference 5 g723r53
    codec preference 6 g723r63
    codec preference 7 gsmfr
    codec preference 8 gsmefr
!
voice class codec 3
    codec preference 1 g726r24
    codec preference 2 gsmefr
    codec preference 3 g726r16
!
fax interface-type modem
    mta receive maximum-recipients 0
controller T1 0
    framing esf
    clock source line secondary 1
    linecode ami
    pri-group timeslots 1-24
    description summa_pbx
!
controller T1 1
    framing esf
    linecode ami
    pri-group timeslots 1-24
    description summa_pbx
!
controller T1 2
    framing sf
    linecode ami
!
controller T1 3
    framing esf
    clock source line primary
    linecode b8zs
    ds0-group 0 timeslots 1-24 type e&m-fgb dtmf dnis
    cas-custom 0
!
gw-accounting h323 vsa
gw-accounting voip
interface Ethernet0
    ip address 172.18.193.99 255.255.255.0
    no ip route-cache
    no ip mroute-cache
    ip rsvp bandwidth 7500 7500
!
interface Serial0:23
    no ip address
    isdn switch-type primary-ni
    isdn incoming-voice modem
    isdn guard-timer 3000
    isdn T203 10000
    isdn T306 30000
    isdn T310 4000
    isdn disconnect-cause 1
    fair-queue 64 256 0
    no cdp enable
interface Serial11:23
    no ip address
    isdn switch-type primary-ni
    isdn incoming-voice modem
    isdn guard-timer 3000
    isdn T203 10000
```

```

    isdn disconnect-cause 1
    fair-queue 64 256 0
    no cdp enable
    !
interface FastEthernet0
    ip address 10.1.1.1 255.255.255.0
    no ip route-cache
    no ip mroute-cache
    duplex auto
    speed auto
    ip rsvp bandwidth 7 7
    !
ip classless
ip route 10.0.0.0 255.0.0.0 172.18.193.1
ip route 172.18.0.0 255.255.0.0 172.18.193.1
no ip http server
ip pim bidir-enable
!
ip radius source-interface Ethernet0
!
map-class dialer test
dialer voice-call
dialer-list 1 protocol ip permit
!
radius-server host 172.18.192.108 auth-port 1645 acct-port 1646
radius-server retransmit 1
radius-server key lab
radius-server vsa send accounting
radius-server vsa send authentication
call rsvp-sync
call application voice voice_billing tftp://172.18.207.16/app_passport_silent.2.0.0.0.tcl
!
voice-port 0:D
voice-port 1:D
voice-port 3:0
!
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer voice 10 pots
    destination-pattern 2021010119
    port 3:0
    prefix 2021010119
!
dial-peer voice 11 pots
    incoming called-number 3111100
    destination-pattern 3100802
    progress_ind progress enable 8
    port 0:D
    prefix 93100802
!
dial-peer voice 36 voip
    application session
    incoming called-number 3100802
    destination-pattern 3100801
    session protocol sipv2
    session target ipv4:172.18.193.100
    codec g726r16
!
dial-peer voice 5 voip
    destination-pattern 5550155
    session protocol sipv2
    session target ipv4:172.18.192.218

```

```

!
dial-peer voice 12 pots
 destination-pattern 3111100
 prefix 93111100
!
dial-peer voice 19 pots
 destination-pattern 2017030200
 port 1:D
 prefix 2017030200
!
dial-peer voice 30 voip
 destination-pattern 36602
 voice-class codec 2
 session protocol sipv2
 session target ipv4:172.18.193.120
dial-peer voice 47 pots
 destination-pattern 2021030100
 port 3:0
!
dial-peer voice 3111200 pots
 destination-pattern 311200
 prefix 93100802
!
dial-peer voice 31 voip
 destination-pattern 36601
 session protocol sipv2
 session target ipv4:172.18.193.98
!
dial-peer voice 1234 voip
 incoming called-number 1234
 destination-pattern 1234
 session target loopback:rtp
!
gateway
 timer receive-rtcp 5
!
sip-ua
 aaa username proxy-auth
 retry invite 1
 retry bye 1
!
line con 0
 exec-timeout 0 0
 line aux 0
 line vty 0 4
 password password1
!
end

```

Perform this task to troubleshoot the SIP: Hold Timer Support feature.

-
- Step 1** Make sure VoIP is working before hold timer support is configured.
- Step 2** Use the **debug ccsip all** command to enable SIP debugging traces. To minimize the possibility of performance impact, use this command during periods of minimal traffic.

```

Router# debug ccsip all

Feb 28 21:34:09.479:Received:
INVITE sip:36601@172.18.193.98:5060 SIP/2.0
Via:SIP/2.0/UDP
172.18.193.187:5060;branch=f104ef32-21751ddb-ce8428fe-cffdbf5-1

```

```

Record-Route:
sip:5550155.f104ef32-21751ddb-ce8428fe-cffdbf5@172.18.197.182:5060;maddr=172.18.193.187>
Via:SIP/2.0/UDP 172.18.197.182:5060;received=172.18.197.182
From:"5550155"
sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17
To:<sip:36601@172.18.193.187>;tag=8CDE00-1506
Call-ID:003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182
CSeq:102 INVITE
User-Agent:CSCO/4
Contact:<sip:5550155@172.18.197.182:5060>
Content-Type:application/sdp
Content-Length:243

v=0
o=Cisco-SIPUA 2802 21073 IN IP4 172.18.197.182
s=SIP Call
c=IN IP4 0.0.0.0
t=0 0
m=audio 28478 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

*Feb 28 21:34:09.479:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:
172.18.193.187:37775
*Feb 28 21:34:09.479:*****CCB found in UAS Request table. ccb=0x63C031B0
*Feb 28 21:34:09.479:CCSIP-SPI-CONTROL: act_active_new_message
*Feb 28 21:34:09.479:CCSIP-SPI-CONTROL: sact_active_new_message_request
*Feb 28 21:34:09.479:CCSIP-SPI-CONTROL: Converting TimeZone EST to SIP
default timezone = GMT
*Feb 28 21:34:09.479:sip_stats_method
*Feb 28 21:34:09.479:sact_active_new_message_request:Case of Mid-Call INVITE

*Feb 28 21:34:09.479:CCSIP-SPI-CONTROL: sipSPIHandleMidCallInvite
*Feb 28 21:34:09.479:CCSIP-SPI-CONTROL: sipSPIUASessionTimer
*Feb 28 21:34:09.479:sipSPIDoMediaNegotiation:number of m lines is 1
*Feb 28 21:34:09.479: Codec (No Codec ) is not in preferred list
*Feb 28 21:34:09.479:sipSPIDoAudioNegotiation:An exact codec match not
configured, using interoperable codec g729r8
*Feb 28 21:34:09.479:sipSPIDoAudioNegotiation:Codec (g729r8) Negotiation
Successful on Static Payload for m-line 1

*Feb 28 21:34:09.479:sipSPIDoPtimeNegotiation:No ptime present or
multiple ptime attributes that can't be handled

*Feb 28 21:34:09.479:sipSPIDoDTMFRelayNegotiation:m-line index 1
*Feb 28 21:34:09.479:sipSPIDoDTMFRelayNegotiation:Requested DTMF-RELAY
option(s) not found in Preferred DTMF-RELAY option list!
*Feb 28 21:34:09.479: sipSPIStreamTypeAndDtmfRelay:DTMF Relay mode :Inband Voice

*Feb 28 21:34:09.479:sip_sdp_get_modem_relay_cap_params:
*Feb 28 21:34:09.479:sip_sdp_get_modem_relay_cap_params:NSE payload from
X-cap = 0
*Feb 28 21:34:09.479:sip_select_modem_relay_params:X-tmr not present in SDP.
Disable modem relay
*Feb 28 21:34:09.479:sipSPIGetSDPDirectionAttribute:No direction attribute
present or multiple direction attributes that can't be handled

*Feb 28 21:34:09.479:sipSPIDoAudioNegotiation:Codec negotiation
successful for media line 1 payload_type=18, codec_bytes=20, codec=g729r8,
dtmf_relay=inband-voice stream_type=voice-only (0), dest_ip_address=0.0.0.0,
dest_port=28478

```



```

*Feb 28 21:34:09.479:sipSPICompareSDP
*Feb 28 21:34:09.483:sipSPICompareStreams:stream 1 dest_port:old=28478
new=28478
*Feb 28 21:34:09.483:sipSPICompareConnectionAddress
*Feb 28 21:34:09.483:sipSPICompareConnectionAddress:Call hold activated for
stream 1

*Feb 28 21:34:09.483:sipSPICompareStreams:Flags set for stream 1:
RTP_CHANGE=No
CAPS_CHANGE=No
*Feb 28 21:34:09.483:sipSPICompareSDP:Flags set for call:NEW_MEDIA=No
DSPDNLD_REQD=No
*Feb 28 21:34:09.483:sipSPIGetGtdBody:No valid GTD body found.
*Feb 28 21:34:09.483:sipSPIReplaceSDP
*Feb 28 21:34:09.483:sipSPICopySdpInfo
*Feb 28 21:34:09.483:sipSPISetHoldTimer:Starting hold timer at 15 minutes
!!Timer started
*Feb 28 21:34:09.483:sipSPIUpdCallWithSdpInfo:
  Preferred Codec      :g729r8, bytes :20
  Preferred DTMF relay :inband-voice
  Preferred NTE payload :101
  Early Media         :Yes
  Delayed Media       :No
  Bridge Done         :Yes
  New Media           :No
  DSP DNLD Reqd      :No

*Feb 28 21:34:09.483:sipSPISetMediaSrcAddr: media src addr for stream 1 =
172.18.193.98
*Feb 28 21:34:09.483:sipSPIUpdCallWithSdpInfo:Stream Type:0
  M-line Index      :1
  State              :STREAM_ACTIVE (5)
  Callid            :1
  Negotiated Codec   :g729r8, bytes :20
  Negotiated DTMF relay :inband-voice
  Negotiated NTE payload :0
  Media Srce Addr/Port :172.18.193.98:18764
  Media Dest Addr/Port :0.0.0.0:28478

*Feb 28 21:34:09.483:sipSPIProcessMediaChanges
*Feb 28 21:34:09.483:CCSIP-SPI-CONTROL: sipSPIIncomingCallSDP
*Feb 28 21:34:09.483: SDP already there use old sdp and updatemedia if needed

*Feb 28 21:34:09.483:sipSPIUpdateSrcSdpVariablePart
*Feb 28 21:34:09.483:sipSPIUpdateSrcSdpVariablePart:setting stream 1
portnum to 18764
*Feb 28 21:34:09.483:CCSIP-SPI-CONTROL: sipSPISendInviteResponse
*Feb 28 21:34:09.483:sipSPIAddLocalContact
*Feb 28 21:34:09.483:sip_generate_sdp_xcaps_list:Modem Relay and T38
disabled.
X-cap not needed
*Feb 28 21:34:09.483: Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE
*Feb 28 21:34:09.483:sip_stats_status_code
*Feb 28 21:34:09.483:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP
172.18.193.187:5060;branch=f104ef32-21751ddb-ce8428fe-cffdbf5-1,SIP/2.0/UDP
172.18.197.182:5060;received=172.18.197.182
From:"5550155"
sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17

To:<sip:36601@172.18.193.187>;tag=8CDE00-1506
Date:Mon, 01 Mar 1993 02:34:09 GMT
Call-ID:003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182

```

```

Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
NOTIFY,
INFO
Allow-Events: telephone-event
Contact: <sip:36601@172.18.193.98:5060>
Record-Route:
sip:5550155.f104ef32-21751ddb-ce8428fe-cffdbf5@172.18.197.182:5060;maddr=172.18.193.18
Content-Type: application/sdp
Content-Length: 229

v=0
o=CiscoSystemsSIP-GW-UserAgent 6264 8268 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 18764 RTP/AVP 18 19
c=IN IP4 172.18.193.98
a=rtpmap:18 G729/8000
a=rtpmap:19 CN/8000
a=fmtp:18 annexb=no

*Feb 28 21:34:09.635:Received:
ACK sip:36601@172.18.193.98:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=f104ef32-21751ddb-ce8428fe-cffdbf5
Record-Route:
<sip:36601.f104ef32-21751ddb-ce8428fe-cffdbf5@172.18.193.187:5060;maddr=172.18.193.187

Via: SIP/2.0/UDP 172.18.197.182:5060
From: "5550155"
sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17

To: <sip:36601@172.18.193.187>;tag=8CDE00-1506
Call-ID: 003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182
CSeq: 102 ACK
User-Agent: CSCO/4
Content-Length: 0

*Feb 28 21:34:09.635:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:
172.18.193.187:37779
*Feb 28 21:34:09.635:****CCB found in UAS Request table. ccb=0x63C031B0
*Feb 28 21:34:09.635:CCSIP-SPI-CONTROL: act_active_new_message
*Feb 28 21:34:09.635:CCSIP-SPI-CONTROL: sact_active_new_message_request
*Feb 28 21:34:09.635:CCSIP-SPI-CONTROL: Converting TimeZone EST to SIP
default timezone = GMT
*Feb 28 21:34:09.635:sip_stats_method

Router#
*Feb 28 21:49:09.483:act_onhold_timeout:Hold Timer Expired, tearing down
call
!!Timer expires after 15 minutes and gateway sends out BYE to the other endpoint.

*Feb 28 21:49:09.483:ccsip_set_release_source_for_peer:ownCallId[1], src[6]

*Feb 28 21:49:09.483: Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION
*Feb 28 21:49:09.483:0x63C031B0 :State change from (STATE_ACTIVE,
SUBSTATE_NONE) to (STATE_ACTIVE, SUBSTATE_CONNECTING)
*Feb 28 21:49:09.483:0x63C031B0 :State change from (STATE_ACTIVE,
SUBSTATE_CONNECTING) to (STATE_ACTIVE, SUBSTATE_CONNECTING)
*Feb 28 21:49:09.483: Queued event from SIP SPI :
SIPSPI_EV_CC_CALL_DISCONNECT
*Feb 28 21:49:09.483:CCSIP-SPI-CONTROL: sipSPICheckSocketConnection:

```

```

Connid(1)
created to 172.18.193.187:5060, local_port 51433
*Feb 28 21:49:09.483:0x63C031B0 :State change from (STATE_ACTIVE,
SUBSTATE_CONNECTING) to (STATE_ACTIVE, SUBSTATE_NONE)
*Feb 28 21:49:09.483:sipSPIStopHoldTimer:Stopping hold timer
*Feb 28 21:49:09.483:CCSIP-SPI-CONTROL: sipSPIAddRouteHeaders status = TRUE
Route <sip:5550155@172.18.197.182:5060>
*Feb 28 21:49:09.483: Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE
*Feb 28 21:49:09.483:sip_stats_method
*Feb 28 21:49:09.483:0x63C031B0 :State change from (STATE_ACTIVE,
SUBSTATE_NONE) to (STATE_DISCONNECTING, SUBSTATE_NONE)
*Feb 28 21:49:09.483:CCSIP-SPI-CONTROL: act_disconnecting_disconnect
*Feb 28 21:49:09.483:Sent:
BYE

sip:5550155.d3c5aelf-b5cf873d-17053c1f-e0126b18@172.18.197.182:5060;maddr=172.18.193.187
SIP/2.0
Via:SIP/2.0/UDP 172.18.193.98:5060
From:<sip:36601@172.18.193.187>;tag=8CDE00-1506
To:"5550155"
<sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17
Date:Mon, 01 Mar 1993 02:34:09 GMT
Call-ID:003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:10
Route:<sip:5550155@172.18.197.182:5060>
Timestamp:730954149
CSeq:101 BYE
Content-Length:0

*Feb 28 21:49:09.487:Received:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP 172.18.193.98:5060;received=172.18.193.98
Call-ID:003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182
From:<sip:36601@172.18.193.187>;tag=8CDE00-1506
To:"5550155"
<sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17
CSeq:101 BYE
Content-Length:0

*Feb 28 21:49:09.487:HandleUdpSocketReads :Msg enqueued for SPI with IPAddr:
172.18.193.187:37781
*Feb 28 21:49:09.487:****CCB found in UAS Response table. ccb=0x63C031B0
*Feb 28 21:49:09.487:CCSIP-SPI-CONTROL: act_disconnecting_new_message
*Feb 28 21:49:09.487:
CCSIP-SPI-CONTROL: sact_disconnecting_new_message_response
*Feb 28 21:49:09.487:CCSIP-SPI-CONTROL: sipSPICheckResponse
*Feb 28 21:49:09.487:sip_stats_status_code
*Feb 28 21:49:09.487: Roundtrip delay 4 milliseconds for method BYE

*Feb 28 21:49:09.539:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 172.18.193.98:5060;received=172.18.193.98
From:<sip:36601@172.18.193.187>;tag=8CDE00-1506
To:"5550155"
<sip:5550155@172.18.193.187>;tag=003094c2e56a00aa13cdcefd-61096c17
Call-ID:003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.182
CSeq:101 BYE
Server:CSCO/4
Content-Length:0

```

```

*Feb 28 21:49:09.539:HandleUdpSocketReads :Msg enqueued for SPI with IPAddr:
172.18.193.187:37784
*Feb 28 21:49:09.539:****CCB found in UAS Response table. ccb=0x63C031B0
*Feb 28 21:49:09.539:CCSIP-SPI-CONTROL: act_disconnecting_new_message
*Feb 28 21:49:09.539:
CCSIP-SPI-CONTROL: sact_disconnecting_new_message_response
*Feb 28 21:49:09.539:CCSIP-SPI-CONTROL: sipSPICheckResponse
*Feb 28 21:49:09.539:sip_stats_status_code
*Feb 28 21:49:09.539: Roundtrip delay 56 milliseconds for method BYE

*Feb 28 21:49:09.539:CCSIP-SPI-CONTROL: sipSPICallCleanup
*Feb 28 21:49:09.539:sipSPIIcpifUpdate :CallState:3 Playout:16840
DiscTime:1014954 ConnTime 924101

*Feb 28 21:49:09.539:0x63C031B0 :State change from (STATE_DISCONNECTING,
SUBSTATE_NONE) to (STATE_DEAD, SUBSTATE_NONE)
*Feb 28 21:49:09.539:The Call Setup Information is :
    Call Control Block (CCB) :0x63C031B0
    State of The Call      :STATE_DEAD
    TCP Sockets Used      :NO
    Calling Number        :5550155
    Called Number         :36601
    Number of Media Streams :1

*Feb 28 21:49:09.539:Media Stream 1
    Negotiated Codec      :g729r8
    Negotiated Codec Bytes :20
    Negotiated Dtmf-relay :0
    Dtmf-relay Payload    :0
    Source IP Address (Media):172.18.193.98
    Source IP Port (Media):18764
    Destn IP Address (Media):0.0.0.0
    Destn IP Port (Media):28478

*Feb 28 21:49:09.539:Orig Destn IP Address:Port (Media):0.0.0.0:0

*Feb 28 21:49:09.539:
Source IP Address (Sig ) :172.18.193.98
Destn SIP Req Addr:Port :172.18.193.187:5060
Destn SIP Resp Addr:Port :172.18.193.187:5060
Destination Name       :172.18.193.187

*Feb 28 21:49:09.539:
Disconnect Cause (CC)  :102
Disconnect Cause (SIP) :200

*Feb 28 21:49:09.539:****Deleting from UAS Request table. ccb=0x63C031B0
key=003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.18236601
*Feb 28 21:49:09.539:****Deleting from UAS Response table. ccb=0x63C031B0
key=003094c2-e56a02b9-670be98d-1cf394e0@172.18.197.1828CDE00-1506
*Feb 28 21:49:09.539:Removing call id 1

*Feb 28 21:49:09.543:RequestCloseConnection:Closing connid 1 Local Port 51433
*Feb 28 21:49:09.543: Queued event from SIP SPI :SIPSPI_EV_CLOSE_CONNECTION
*Feb 28 21:49:09.543:sipSPIFlushEventBufferQueue:There are 0 events on the
internal queue that are going to be free'd
*Feb 28 21:49:09.543: freeing ccb 63C031B0

*Feb 28 21:49:09.543:udpsock_close_connect:Socket fd:1 closed for connid 1 with remote
port:5060

```

Additional References

General SIP References

- “SIP Features Roadmap” on page 1—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- “Overview of SIP” on page 1—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter

- *Cisco IOS Quality of Service Solutions Command Reference*, Release 12.3 at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgr/qos_r/index.htm

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Configuring SIP MWI Features

This chapter discusses message-waiting indication (MWI) in a SIP-enabled network.

Feature History for the SIP MWI NOTIFY - QSIG MWI Translation

Release	Modification
12.4(11)T	This feature was introduced.

Feature History for SIP Audible Message-Waiting Indicator for FXS Phones

Release	Modification
12.3(8)T	This feature was introduced.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

- [Prerequisites for SIP MWI, page 2](#)
- [Restrictions for SIP MWI, page 2](#)
- [Information About SIP MWI, page 2](#)
- [How to Configure SIP MWI, page 5](#)
- [Configuration Examples for SIP MWI, page 17](#)
- [Additional References, page 21](#)



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Prerequisites for SIP MWI

SIP MWI NOTIFY - QSIG MWI Translation Feature

- Ensure that you have a working SIP network with the following:
 - A voice-messaging system that provides a SIP MWI Notify message to the phone—including Cisco Unified Communications Manager (formerly known as Cisco CallManager), Release 5.0 or later or Cisco Unified Communications Manager Express (Cisco Unified CME, formerly known as Cisco CallManager Express) Release 4.0 or later.
 - Voice messaging on Cisco Unity 4.0.1 or later releases (colocated or integrated with the Cisco Unified Communications Manager) or an ISDN Q-signaling (QSIG) PBX.
- Connect gateway and Cisco routers directly to a PBX.
- Ensure that phones connected to PBXs support MWI notification.

SIP Audible Message-Waiting Indicator for FXS Phones Feature

- The MWI tone is generated by the voice-mail server. Be sure that you understand how to configure MWI service on a voice-mail server (such as Cisco Unity).

Restrictions for SIP MWI

SIP MWI NOTIFY - QSIG MWI Translation Feature

- Visual MWI for phones is a functionality of the phone itself and is not addressed in this document.
- The feature supports only SIP unsolicited notify and does not support SIP subscribe notify.
- This feature is not supported in trunk groups in ISDN circuits. In this scenario, trunk groups disable the SIP MWI feature.

SIP Audible Message-Waiting Indicator for FXS Phones Feature

- The SIP Audible Message-Waiting Indicator for FXS Phones feature does not provide the following functionality:
 - Security or authentication services
 - Call redirection to the voice-mail server when the line is busy or there is no answer
 - Instructions on accessing the voice-mail server or retrieving voice messages

Information About SIP MWI

The SIP Audible Message-Waiting Indicator for FXS Phones feature enables an FXS port on a voice gateway to receive audible MWI in a SIP-enabled network. The FXS port on a voice gateway is an RJ-11 connector that allows connections to basic telephone service equipment.

This feature provides the following benefits:

- Message waiting is now indicated to FXS phone users through an audible tone, replicating the functionality users have with traditional telephone systems.
- By means of the Cisco IOS command-line interface, you can enable or disable MWI under the voice port and configure one voice-mail server per user agent (UA) or voice gateway.

To configure SIP MWI support, you should understand the following concepts:

- [SUBSCRIBE/NOTIFY MWI, page 3](#)
- [Unsolicited MWI, page 4](#)

SUBSCRIBE/NOTIFY MWI

MWI is a common feature of telephone networks and uses an audible indication (such as a special dial tone) that a message is waiting. The IETF draft *A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)* draft-ietf-sipping-mwi-03.txt defines MWI as “a SIP event package carrying message waiting status and message summaries from a messaging system to an interested user agent.”

In Cisco SIP networks, the event notification mechanisms used to carry message waiting status are the SUBSCRIBE and NOTIFY methods. The SUBSCRIBE method requests notification of an event. The NOTIFY method provides notification that an event requested by an earlier SUBSCRIBE method has occurred.



Note

For information on the SUBSCRIBE and NOTIFY methods, see the “[Configuring Additional SIP Application Support](#)” chapter of the *Cisco IOS SIP Configuration Guide*.

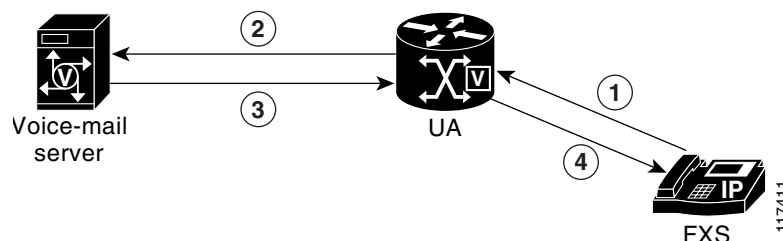
In this feature, a UA (on behalf of the analog FXS phone) subscribes to a voice-mail server to request notification of mailbox status. When the mailbox status changes, the voice-mail server notifies the UA. The UA then indicates that there is a change in mailbox status by providing an MWI tone when the user takes the phone off-hook.

The frequency and cadence of the MWI tone may vary from country to country. For North America, it is defined in GR-506. After you configure the **cp tone** command under your voice port, Cisco IOS software chooses the correct MWI tone accordingly.

Each voice port has its own subscription and notification process. If there are multiple dial peers associated with an FXS voice port, multiple subscriptions are sent to the voice-mail server. If the voice port does not have MWI enabled, the voice gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server.

[Figure 95](#) shows the basic MWI subscription and notification flow.

Figure 95 **MWI Notification Flow**



1. The user enables the MWI service for the FXS phone by configuring the voice gateway.
2. The UA sends a subscription request to the server on the user’s behalf.
3. The voice-mail server notifies the UA when there is a change in voice-mail status.
4. The UA notifies the phone user with an audible tone.

Unsolicited MWI

In addition to the MWI status forwarded by using the SUBSCRIBE and NOTIFY methods, unsolicited MWI notify is also supported. With unsolicited MWI, MWI service is initially configured on the voice-mail server. The UA does not need to subscribe to the voice-mail server to receive MWI service. If configured for unsolicited MWI, the voice-mail server automatically sends a SIP notification message to the UA if the mailbox status changes.

SIP MWI NOTIFY - QSIG MWI Translation

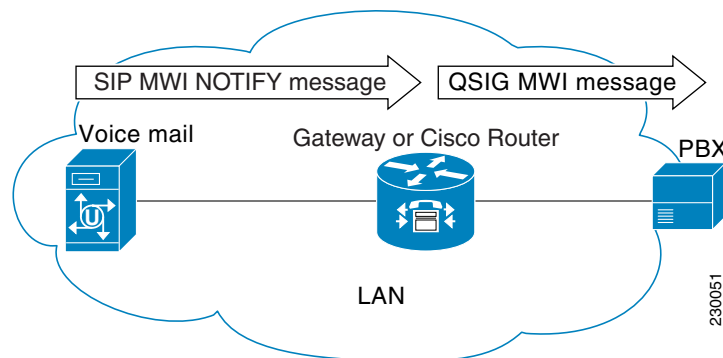
In Cisco IOS Release 12.4(11)T, the SIP MWI NOTIFY - QSIG Translation feature enhances MWI functionality to include SIP-MWI-NOTIFY-to-QSIG-MWI translation between Cisco gateways or routers over a LAN or WAN and extends message waiting indicator (MWI) functionality for SIP MWI and QSIG MWI interoperation to enable sending MWI over QSIG from a Cisco IOS SIP gateway to a PBX.

When the SIP Unsolicited NOTIFY is received from voice mail, the Cisco router translates this event to activate QSIG MWI to the PBX via PSTN. The PBX will switch the MWI lamp either on or off on the corresponding IP phone as appropriate.

This feature supports only Unsolicited NOTIFY. Subscribe NOTIFY is not supported by this feature.

In [Figure 96](#), the Cisco router receives the SIP Unsolicited NOTIFY, performs the protocol translation, and initiates the QSIG MWI call to the PBX, where it is routed to the appropriate phone.

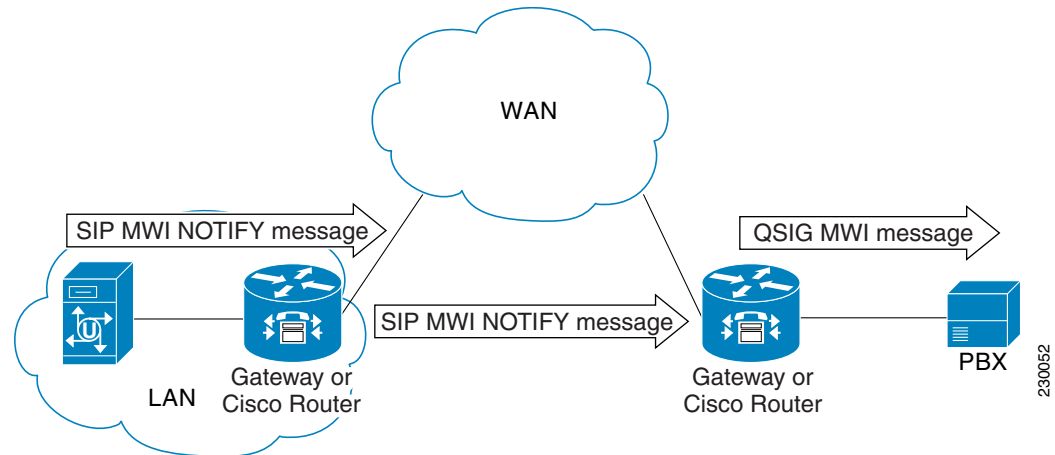
Figure 96 SIP MWI to ISDN QSIG When Voice Mail and Cisco Router are On the Same LAN



Whether the SIP Unsolicited NOTIFY is received via LAN or WAN does not matter as long as the PBX is connected to the gateway or Cisco router, and not to the remote voice mail server.

In [Figure 97](#), a voice mail system, such as Cisco Unity, and Unified CME are connected to the same LAN and a remote Unified CME is connected across the WAN. In this scenario, the protocol translation is performed at the remote Unified CME router and the QSIG MWI message is sent to the PBX.

Figure 97 SIP MWI to ISDN QSIG When PBX is Connected to a Remote Cisco Router



How to Configure SIP MWI

This section contains the following procedures for configuring the SIP Audible MWI for FXS Phones feature:

- [Configuring SIP MWI NOTIFY - QSIG MWI Translation, page 5](#)
- [Configuring Voice-Mail Server Settings on the UA, page 7](#)
- [Configuring the Voice-Mail Server for Unsolicited, page 8](#)
- [Enabling MWI Under an FXS Voice Port, page 9](#)
- [Verifying MWI Settings, page 10](#)
- [Troubleshooting Tips, page 11](#)



Note

- Before you perform a procedure, familiarize yourself with the following information:
 - [“Prerequisites for SIP MWI”](#) section on page 2
 - [“Restrictions for SIP MWI”](#) section on page 2
- For help with a procedure, see the verification and troubleshooting sections listed above.

Configuring SIP MWI NOTIFY - QSIG MWI Translation

This section contains information for configuring SIP MWI NOTIFY - QSIG MWI Translation on a gateway.

**Note**

All configuration for this feature is done on the gateway or Cisco router.

Configuring the Gateway

To configure SIP MWI NOTIFY - QSIG MWI Translation on a gateway, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port**
4. **mwi**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice-port <i>slot/port</i> Example: Router(config)# voice-port 2/2	Enters voice-port configuration mode for the specified PRI or BRI voice port.
Step 4	mwi Example: Router (config-voiceport)# mwi	Enables MWI on this voice port. Note If the voice port is not configured for MWI, the gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server. If multiple dial peers are associated with the same voice port, multiple subscriptions are sent to the voice-mail server.
Step 5	exit Example: Router(config-dial-peer-voice)# exit	Exits the current configuration mode.

Configuring Voice-Mail Server Settings on the UA

To configure voice-mail server settings on the UA, perform the following steps.



Note

This configuration initiates the capability of a UA or voice gateway to indicate voice-mail status changes. One voice-mail server is configured per voice gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP user-agent configuration mode.

Command or Action	Purpose
<p>Step 4</p> <pre>mwi-server {ipv4:destination-address dns:host-name} [expires seconds] [port port] [transport {tcp udp}] [unsolicited]</pre> <p>Example: Router(config-sip-ua)# mwi-server dns:test.example.com expires 86000 port 5060 transport udp unsolicited</p>	<p>Configures voice-mail server settings on a voice gateway or UA. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • ipv4:destination-address—IP address of the voice-mail server. • dns:host-name—Host device housing the domain name server that resolves the name of the voice-mail server. The argument should contain the complete hostname to be associated with the target address; for example, dns:test.example.com. • expires seconds—Subscription expiration time, in seconds. Range is from 1 to 999999. Default is 3600. • port port—Port number on the voice-mail server. Default is 5060. • transport—Transport protocol to the voice-mail server. Valid values are tcp and udp. Default is UDP. • unsolicited—Requires the voice-mail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
<p>Step 5</p> <pre>exit</pre> <p>Example: Router(config-sip-ua)# exit</p>	<p>Exits the current mode.</p>

Configuring the Voice-Mail Server for Unsolicited

To configure the Cisco Unity voice-mail server to be unsolicited, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server ipv4:x.x.x.x unsolicited**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Router(config)# sip-ua	Enters SIP-user-agent configuration mode.
Step 4	mwi-server ipv4:x.x.x.x unsolicited Example: Router (config-sip-ua)# mwi-server ipv4:192.0.10.150 unsolicited	Configures the specified voice-mail (MWI) server to be unsolicited. (That is, requires the server to send a SIP notification message to the voice gateway or user agent if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.)
Step 5	exit Example: Router(config-sip-ua)# exit	Exits the current configuration mode.

Enabling MWI Under an FXS Voice Port

To enable MWI under the specified FXS voice port, perform the following steps.

**Note**

If the voice port does not have MWI enabled, the voice gateway returns a 481 Call Leg/Transaction Does Not Exist message to the voice-mail server.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port**
4. **cptone**
5. **mwi**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice-port port Example: Router(config)# voice-port 2/2	Enters voice-port configuration mode. To find the <i>port</i> argument for your router, see the Cisco IOS Voice Command Reference , Release 12.3T.
Step 4	cptone locale Example: Router(config-voiceport)# cptone us	Specifies a regional analog voice-interface-related tone, ring, and cadence setting for a specified FXS voice port.
Step 5	mwi Example: Router(config-voiceport)# mwi	Enables MWI for a specified FXS voice port.
Step 6	exit Example: Router(config-voiceport)# exit	Exits the current mode.

Verifying MWI Settings

To verify MWI settings, perform the following step.

SUMMARY STEPS

1. **show sip-ua mwi**

DETAILED STEPS

Step 1 **show sip-ua mwi**

Use this command to display SIP MWI settings from the voice-mail server. The command displays endpoint status as OFF if a message is deleted or if no message is waiting. The endpoint status changes to ON when a message is waiting.

The following sample output shows endpoint status as OFF if a message is deleted or if no message is waiting. The endpoint status changes to ON when a message is waiting.

```
Router# show sip-ua mwi
```



```

MWI type: 2
MWI server: dns:unity-vm.example1.com
MWI expires: 60
MWI port: 5060
MWI transport type: UDP
MWI unsolicited
MWI server IP address:
C801011E
0
0
0
0
0
0
0
0
0
MWI ipaddr cnt 1:
MWI ipaddr idx 0:
MWI server: 192.168.1.30, port 5060, transport 1
MWI server dns lookup retry cnt: 0
endpoint 8000 mwi status ON
endpoint 8000 mwi status ON
endpoint 8001 mwi status OFF

```

Troubleshooting Tips



Note

For general troubleshooting tips and a list of important **debug** commands, see the [Verifying and Troubleshooting SIP Features](#) chapter in the *Cisco IOS SIP Configuration Guide*.

- Use the **debug ccsip messages** command for debugging purposes.

Following is sample output for this command:

- [Sample Output for the debug ccsip messages Command, page 11](#)

Sample Output for the debug ccsip messages Command

The following sample output is from the perspective of a SIP UA acting on the behalf of an analog FXS phone. The output shows that when the phone connected to the UA is called and the line is busy, the caller leaves a message. The UA, connected to the voice-mail server, receives notification and provides a tone to the user. The user listens to the message and deletes it.

```

Router# debug ccsip messages
00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:78002@csp-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK24E9
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csp-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER

```

```

CSeq: 101 INVITE
Max-Forwards: 70
Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK24E9
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
CSeq: 101 INVITE
Content-Length: 0

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 407 Proxy Authentication Required
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK24E9
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=5ea400de-695763f1
CSeq: 101 INVITE
Proxy-Authenticate: DIGEST realm="example.com", nonce="40871b34", qop="auth",
algorithm=MD5
Content-Length: 0

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:78002@csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK24E9
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=5ea400de-695763f1
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Max-Forwards: 70
CSeq: 101 ACK
Content-Length: 0

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:78002@csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612

```

```

To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq: 102 INVITE
Max-Forwards: 70
Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Proxy-Authorization: Digest
username="user1",realm="example.com",uri="sip:192.168.1.37",response="df92654ce55d734
6398013442919e7fc",nonce="40871b34",cnonce="2AEBD5CD",qop=auth,algorithm=MD5,nc=00000001
Content-Type: application/sdp
Content-Length: 234

v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
CSeq: 102 INVITE
Content-Length: 0

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:78002@192.168.1.174:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1
Record-Route:
<sip:78001.474b6083-19c218c7-16e9de49-93b83d71@192.168.1.174:5060;maddr=192.168.1.37>
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 3659524871-1844515286-2148452871-566800187
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
CSeq: 102 INVITE
Max-Forwards: 69

```

```

Remote-Party-ID: "SIPMWI-1" <sip:78001@192.168.1.174>;party=calling;screen=no;privacy=off
Timestamp: 1022206059
Contact: <sip:78001@192.168.1.174:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 234

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 5421 615 IN IP4 192.168.1.174
s=SIP Call
c=IN IP4 192.168.1.174
t=0 0
m=audio 16818 RTP/AVP 18 19
c=IN IP4 192.168.1.174
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1,SIP/2.0/UDP
192.168.1.174:5060;re
ceived=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Timestamp: 1022206059
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 INVITE
Allow-Events: telephone-event
Content-Length: 0

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 486 Busy here
Via: SIP/2.0/UDP
192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1,SIP/2.0/UDP
192.168.1.174:5060;re
ceived=192.168.1.174;branch=z9hG4bK612
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Timestamp: 1022206059
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 INVITE
Allow-Events: telephone-event
Reason: Q.850;cause=17
Content-Length: 0

```

```

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:78002@192.168.1.174:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.37:5060;branch=474b6083-19c218c7-16e9de49-93b83d71-1
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A843C-187B

```

CSeq: 102 ACK
Content-Length: 0

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 180 Ringing
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 102 INVITE
Content-Length: 0

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK612
Record-Route:
<sip:7200@example1.com:5060;maddr=192.168.1.37>,<sip:78002@csps-release.example1.com:5060;
maddr=192.168.1.37>
Contact: sip:7200@192.168.1.30:5060
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 102 INVITE
Content-Length: 166
Content-Type: application/sdp

v=0
o=192.168.1.30 7542610 7542610 IN IP4 192.168.1.30
s=No Subject
c=IN IP4 192.168.1.30
t=0 0
m=audio 22840 RTP/AVP 18
a=rtpmap:18 G729/8000
a=fmtp:18 annex=no

00:11:29: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:78002@csps-release.example1.com:5060;maddr=192.168.1.37 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK10EF
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
Route: <sip:7200@example1.com:5060;maddr=192.168.1.37>,<sip:7200@192.168.1.30:5060>
Max-Forwards: 70
CSeq: 102 ACK
Proxy-Authorization: Digest
username="user1",realm="example.com",uri="sip:192.168.1.37",response="631ff1eec9e21b0
2fcbdbe932c9f7b5b",nonce="40871b34",cnonce="81C16CF6",qop=auth,algorithm=MD5,nc=00000002
Content-Length: 0

00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
REGISTER sip:csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK171F
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:48 GMT
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B

```
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1022206068
CSeq: 14 REGISTER
Contact: <sip:78002@192.168.1.174:5060>
Expires: 60
Content-Length: 0
```

```
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK171F
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 14 REGISTER
Content-Length: 0
```

```
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK171F
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 14 REGISTER
WWW-Authenticate: DIGEST realm="example.com", nonce="40871b3d", qop="auth", algorithm=MD5
Content-Length: 0
```

```
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
REGISTER sip:csps-release.example1.com:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK21B5
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
Date: Fri, 24 May 2002 02:07:48 GMT
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1022206068
CSeq: 15 REGISTER
Contact: <sip:78002@192.168.1.174:5060>
Expires: 60
Authorization: Digest
username="user2", realm="example.com", uri="sip:192.168.1.37", response="134885a71dd969037019
6
089e445e955", nonce="40871b3d", cnonce="7446932B", qop=auth, algorithm=MD5, nc=00000001
Content-Length: 0
```

```
00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK21B5
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 15 REGISTER
Content-Length: 0
```

```

00:11:38: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK21B5
Call-ID: 6CD62112-6DF011D6-8006CA07-21C8AF3B
From: "user2" <sip:78002@192.168.1.174>;tag=AA7F4-1F83
To: <sip:78002@csps-release.example1.com>
CSeq: 15 REGISTER
Contact: <sip:78002@192.168.1.174:5060>;expires=60
Content-Length: 0

00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
BYE sip:78002@csps-release.example1.com:5060;maddr=192.168.1.37 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.174:5060;branch=z9hG4bK79A
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Date: Fri, 24 May 2002 02:07:39 GMT
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Route: <sip:7200@example1.com:5060;maddr=192.168.1.37>, <sip:7200@192.168.1.30:5060>
Timestamp: 1022206074
CSeq: 103 BYE
Reason: Q.850;cause=16
Proxy-Authorization: Digest
username="user1", realm="example.com", uri="sip:192.168.1.37", response="dfc15fe72d26b9
3d78162852a1a341", nonce="40871b34", cnonce="AF9FD85E", qop=auth, algorithm=MD5, nc=00000003
Content-Length: 0

00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK79A
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
CSeq: 103 BYE
Content-Length: 0

00:11:44: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
From: "SIPMWI-1" <sip:78001@192.168.1.174>;tag=A842C-2612
To: <sip:78002@csps-release.example1.com>;tag=A59035E8274E4600A8F3D15C3DAB9631
Via: SIP/2.0/UDP 192.168.1.174:5060;received=192.168.1.174;branch=z9hG4bK79A
Call-ID: DBAC09D2-6DF111D6-8011CA07-21C8AF3B@192.168.1.174
CSeq: 103 BYE
Content-Length: 0

```

Configuration Examples for SIP MWI

The following example shows that SIP MWI is configured on the gateway.

```

Router# show running-config

Building configuration...

```

```

Current configuration : 14146 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
no service dhcp
!
boot-start-marker
boot system flash:c2430-is-mz.mwi_dns
boot-end-marker
!
card type e1 1
logging buffered 9000000 debugging
!
username all
network-clock-participate E1 1/0
network-clock-participate E1 1/1
no aaa new-model
no ip subnet-zero
!
ip domain name example1.com
ip name-server 192.168.1.1
ip dhcp excluded-address 172.16.224.97
!
isdn switch-type primary-qsig
!
trunk group Incoming
!
voice-card 0
!
voice service voip
  fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none
  h323
  sip
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g726r32
!
voice hpi capture buffer 100000
voice hpi capture destination flash:t1.dat
!
voice translation-rule 1
  rule 1 /.*/ /8005550100/
!
voice translation-profile Out
  translate calling 1
!
controller E1 1/0
  linecode ami
  pri-group timeslots 1-31
!
controller E1 1/1
  linecode ami
  pri-group timeslots 1-10,16
!
interface FastEthernet0/0
  ip address 192.168.1.172 255.255.255.0
  no ip mroute-cache
  duplex half
  speed auto
!

```



```
interface FastEthernet0/1
 ip address 10.2.141.19 255.255.0.0
 no ip mroute-cache
 duplex auto
 speed auto
 !
 ip http server
 ip classless
 ip route 0.0.0.0 0.0.0.0 192.168.1.2
 !
 ip rtcp report interval 30000
 !
 control-plane
 !
 ! Enable MWI on voice ports 2/0 and 2/1.
 !
 voice-port 2/0
  mwi
  timeouts ringing 30
  station-id name SIPUser1
  station-id number 8000
  caller-id enable
 !
 voice-port 2/1
  mwi
  timeouts ringing 30
  station-id name SIPUser2
  station-id number 8001
  caller-id enable
 !
 dial-peer cor custom
 !
 ! Configure dial peers.
 !
 dial-peer voice 1 pots
  preference 5
  destination-pattern 8000
  port 2/0
 !
 dial-peer voice 2 pots
  preference 5
  destination-pattern 8001
  port 2/1
 !
 dial-peer voice 3 voip
  destination-pattern .T
  voice-class codec 1
  session protocol sipv2
  session target sip-server
  dtmf-relay rtp-nte
 !
 dial-peer voice 7 pots
  trunkgroup Incoming
  destination-pattern 789...
 !
 dial-peer voice 8 pots
  trunkgroup Incoming
  destination-pattern 789...
 !
 dial-peer voice 22 voip
  destination-pattern 7232
  session protocol sipv2
  session target sip-server
  dtmf-relay rtp-nte
```

```

    codec g711ulaw
    !
gateway
    timer receive-rtcp 5
    timer receive-rtp 1200
    !
    ! Configure the voice-mail server settings on the gateway with the mwi-server command.
    !
sip-ua
    authentication username user1 password password1 realm example.com
    mwi-server dns:test.example.com expires 60 port 5060 transport udp unsolicited
    registrar dns:csps-release.test.example.com expires 3600
    sip-server dns:csps-release.test.example.com
    !
telephony-service
    max-dn 100
    max-conferences 4
    !
ephone-dn 1
    !
line con 0
    exec-timeout 0 0
    password 7 password2
    transport preferred all
    transport output all
line aux 0
    transport preferred all
    transport output all
line vty 0 4
    password 7 password3
    login
    transport preferred all
    transport input all
    transport output all
    !
end

```

Configuration Example for SIP MWI NOTIFY - QSIG MWI Translation

The following example shows a sample configuration of the SIP MWI NOTIFY - QSIG MWI Translation feature on a SIP gateway.

```

dial-peer voice 1000 voip
    destination-pattern .T
    session protocol sipv2
    session target ipv4:10.120.70.10
    incoming called-number .T
    dtmf-relay rtp-nte
    !
sip-ua
    !
    !
line con 0
    exec-timeout 0 0
line aux 0
line vty 0 4
    login
    !
end

```

Additional References

General SIP References

- “[SIP Features Roadmap](#)” chapter—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- “[Basic SIP Configuration](#)” chapter—Describes underlying SIP technology; also lists related documents, standards, MIBs, RFCs, and how to obtain technical assistance.

References Mentioned in This Chapter (listed alphabetically)

- [RFC 3842, “A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol \(SIP\)”](http://www.ietf.org/rfc/rfc3842.txt) at <http://www.ietf.org/rfc/rfc3842.txt>
- [Cisco IOS Voice Command Reference](http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html) at http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html
- [Cisco IOS Voice Configuration Library](http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm) at http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm

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Verifying and Troubleshooting SIP Features

This chapter describes how to verify and troubleshoot Cisco SIP features.

Contents

- [Basic Troubleshooting Procedures, page 1](#)
- [Using show Commands, page 2](#)
- [Using debug Commands, page 6](#)
- [Additional References, page 7](#)

Basic Troubleshooting Procedures

Cisco routers provide numerous integrated commands to assist you in monitoring and troubleshooting your internetwork:

- **show** commands help you monitor installation behavior and normal network behavior, and isolate problem areas.
- **debug** commands help you isolate protocol and configuration problems.
- **ping** commands help you determine connectivity between devices on your network.
- **trace** commands provide a method of determining the route by which packets reach their destination.

This chapter discusses use of **show** and **debug** commands.



Note

Under moderate traffic loads, **debug** commands produce a high volume of output. We therefore recommend that, as a general rule, you use **show** commands first and use **debug** commands with caution.

Generally, you should proceed as follows:

1. Determine whether or not VoIP is working.
2. Determine whether or not you can make a voice call.



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3. Verify that SIP-supported codecs are used. Support for codecs varies on different platforms; use the **codec ?** command to determine the codecs available on a specific platform.
4. Isolate and reproduce the failure.
5. Collect relevant information from **show** and **debug** commands, configuration files, and protocol analyzers.
6. Identify the first indication of failure in protocol traces or internal **debug** command output.
7. Look for the cause in configuration files.

**Note**

General troubleshooting of problems affecting basic functionality such as dial peers, digit translation, and IP connectivity is beyond the scope of this chapter. For links to additional troubleshooting help, see the [“Additional References” section on page 7](#).

Using show Commands

To verify SIP gateway status and configuration, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show sip service**
2. **show sip-ua register status**
3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**

DETAILED STEPS

Step 1 **show sip service**

Use this command to display the status of SIP call service on a SIP gateway.

The following sample output shows that SIP call service is enabled:

```
Router# show sip service

SIP Service is up
```

The following sample output shows that SIP call service was shut down with the **shutdown** command:

```
Router# show sip service

SIP service is shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop** command:

```
Router# show sip service

SIP service is shut
under 'voice service voip', 'sip' submode
```

The following sample output shows that SIP call service was shut down with the **shutdown forced** command:

```
Router# show sip service

SIP service is forced shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop forced** command:

```
Router# show sip service

SIP service is forced shut
under 'voice service voip', 'sip' submode
```

Step 2 show sip-ua register status

Use this command to display the status of E.164 numbers that a SIP gateway has registered with an external primary SIP registrar.

```
Router# show sip-ua register status

Line peer expires(sec) registered
4001 20001 596 no
4002 20002 596 no
5100 1 596 no
9998 2 596 no
```

Step 3 show sip-ua statistics

Use this command to display response, traffic, and retry SIP statistics, including whether call redirection is disabled.

The following sample shows that four registers were sent:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
  Trying 0/0, Ringing 0/0,
  Forwarded 0/0, Queued 0/0,
  SessionProgress 0/0
Success:
  OkInvite 0/0, OkBye 0/0,
  OkCancel 0/0, OkOptions 0/0,
  OkPrack 0/0, OkPreconditionMet 0/0,
  OkSubscribe 0/0, OkNOTIFY 0/0,
  OkInfo 0/0, 202Accepted 0/0
  OkRegister 12/49
Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
  MultipleChoice 0, MovedPermanently 0,
  MovedTemporarily 0/0, UseProxy 0,
  AlternateService 0
Client Error:
  BadRequest 0/0, Unauthorized 0/0,
  PaymentRequired 0/0, Forbidden 0/0,
  NotFound 0/0, MethodNotAllowed 0/0,
  NotAcceptable 0/0, ProxyAuthReqd 0/0,
  ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
  ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
  UnsupportedMediaType 0/0, BadExtension 0/0,
  TempNotAvailable 0/0, CallLegNonExistent 0/0,
  LoopDetected 0/0, TooManyHops 0/0,
```

```

    AddrIncomplete 0/0, Ambiguous 0/0,
    BusyHere 0/0, RequestCancel 0/0,
    NotAcceptableMedia 0/0, BadEvent 0/0,
    SETooSmall 0/0
Server Error:
    InternalError 0/0, NotImplemented 0/0,
    BadGateway 0/0, ServiceUnavail 0/0,
    GatewayTimeout 0/0, BadSipVer 0/0,
    PreCondFailure 0/0
Global Failure:
    BusyEverywhere 0/0, Decline 0/0,
    NotExistAnywhere 0/0, NotAcceptable 0/0
Miscellaneous counters:
    RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0,
    Subscribe 0/0, NOTIFY 0/0,
    Refer 0/0, Info 0/0
    Register 49/16

Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
Register 4

```

```

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0
Last time SIP Statistics were cleared: <never>

```

The following sample output shows the `RedirectResponseMappedToClientError` status message. An incremented number indicates that 3xx responses are to be treated as 4xx responses. When call redirection is enabled (default), the `RedirectResponseMappedToClientError` status message is not incremented.

```

Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
Informational:
    Trying 0/0, Ringing 0/0,
    Forwarded 0/0, Queued 0/0,
    SessionProgress 0/0
Success:
    OkInvite 0/0, OkBye 0/0,
    OkCancel 0/0, OkOptions 0/0,
    OkPrack 0/0, OkPreconditionMet 0/0,
    OKSubscribe 0/0, OkNotify 0/0,
    202Accepted 0/0
Redirection (Inbound only):
    MultipleChoice 0, MovedPermanently 0,
    MovedTemporarily 0, UseProxy 0,
    AlternateService 0
Client Error:
    BadRequest 0/0, Unauthorized 0/0,
    PaymentRequired 0/0, Forbidden 0/0,
    NotFound 0/0, MethodNotAllowed 0/0,
    NotAcceptable 0/0, ProxyAuthReqd 0/0,
    ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
    ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,

```



```

UnsupportedMediaType 0/0, BadExtension 0/0,
TempNotAvailable 0/0, CallLegNonExistent 0/0,
LoopDetected 0/0, TooManyHops 0/0,
AddrIncomplete 0/0, Ambiguous 0/0,
BusyHere 0/0, RequestCancel 0/0
NotAcceptableMedia 0/0, BadEvent 0/0
Server Error:
  InternalError 0/0, NotImplemented 0/0,
  BadGateway 0/0, ServiceUnavail 0/0,
  GatewayTimeout 0/0, BadSipVer 0/0,
  PreCondFailure 0/0
Global Failure:
  BusyEverywhere 0/0, Decline 0/0,
  NotExistAnywhere 0/0, NotAcceptable 0/0
Miscellaneous counters:
  RedirectResponseMappedToClientError 1,
SIP Total Traffic Statistics (Inbound/Outbound)
  Invite 0/0, Ack 0/0, Bye 0/0,
  Cancel 0/0, Options 0/0,
  Prack 0/0, Comet 0/0,
  Subscribe 0/0, Notify 0/0,
  Refer 0/0

Retry Statistics
  Invite 0, Bye 0, Cancel 0, Response 0,
  Prack 0, Comet 0, Reliable1xx 0, Notify 0

SDP application statistics:
  Parses: 0, Builds 0
  Invalid token order: 0, Invalid param: 0
  Not SDP desc: 0, No resource: 0

```

Step 4 show sip-ua status

Use this command to display status for the SIP user agent (UA), including whether call redirection is enabled or disabled.

```

Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
Redirection (3xx) message handling: ENABLED

```

Step 5 show sip-ua timers

Use this command to display the current settings for the SIP user-agent (UA) timers.

The following sample output shows the waiting time before a register request is sent—that is, the value that is set with the **timers register** command:

```

Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rellxx 500, notify 500
refer 500, register 500

```

Using debug Commands



Note

Commands are listed in alphabetical order.

- Use the **debug aaa authentication** command to display high-level diagnostics related to AAA logins.
- Use the **debug asnl events** command to verify that the SIP subscription server is up. The output displays a pending message if, for example, the client is unsuccessful in communicating with the server.
- Use the **debug call fallback** family of commands to display details of VoIP call fallback.
- Use the **debug cch323** family of commands to provide debugging output for various components within an H.323 subsystem.
- Use the **debug ccsip** family of commands for general SIP debugging, including viewing direction-attribute settings and port and network address-translation traces. Use any of the following related commands:
 - **debug ccsip all**—Enables all SIP-related debugging
 - **debug ccsip calls**—Enables tracing of all SIP service-provider interface (SPI) calls
 - **debug ccsip error**—Enables tracing of SIP SPI errors
 - **debug ccsip events**—Enables tracing of all SIP SPI events
 - **debug ccsip info**—Enables tracing of general SIP SPI information, including verification that call redirection is disabled
 - **debug ccsip media**—Enables tracing of SIP media streams
 - **debug ccsip messages**—Enables all SIP SPI message tracing, such as those that are exchanged between the SIP user-agent client (UAC) and the access server
 - **debug ccsip preauth**—Enables diagnostic reporting of authentication, authorization, and accounting (AAA) preauthentication for SIP calls
 - **debug ccsip states**—Enables tracing of all SIP SPI state tracing
 - **debug ccsip transport**—Enables tracing of the SIP transport handler and the TCP or User Datagram Protocol (UDP) process
- Use the **debug isdn q931** command to display information about call setup and teardown of ISDN network connections (layer 3) between the local router (user side) and the network.
- Use the **debug kpml** command to enable debug tracing of KPML parser and builder errors.
- Use the **debug radius** command to enable debug tracing of RADIUS attributes.
- Use the **debug rpms-proc preauth** command to enable debug tracing on the RPMS process for H.323 calls, SIP calls, or both H.323 and SIP calls.
- Use the **debug rtr trace** command to trace the execution of an SAA operation.
- Use the **debug voip** family of commands, including the following:
 - **debug voip ccapi protoheaders**—Displays messages sent between the originating and terminating gateways. If no headers are being received by the terminating gateway, verify that the **header-passing** command is enabled on the originating gateway.
 - **debug voip ivr script**—Displays any errors that might occur when the Tcl script is run

- **debug voip rtp session named-event 101**—Displays information important to DTMF-relay debugging, if you are using codec types g726r16 or g726r24. Be sure to append the argument *101* to the command to prevent the console screen from flooding with messages and all calls from failing.

Sample output for some of these commands follows:

- [Sample Output for the debug ccsip events Command, page 7](#)
- [Sample Output for the debug ccsip info Command, page 7](#)

Sample Output for the debug ccsip events Command

- The example shows how the Proxy-Authorization header is broken down into a decoded username and password.

```
Router# debug ccsip events

CCSIP SPI: SIP Call Events tracing is enabled

21:03:21: sippmh_parse_proxy_auth: Challenge is 'Basic'.
21:03:21: sippmh_parse_proxy_auth: Base64 user-pass string is
'MTIZNDU2Nzg5MDEyMzQ1Njou'.
21:03:21: sip_process_proxy_auth: Decoded user-pass string is '1234567890123456:.'.
21:03:21: sip_process_proxy_auth: Username is '1234567890123456'.
21:03:21: sip_process_proxy_auth: Pass is '..'.
21:03:21: sipSPIAddBillingInfoToCcb: sipCallId for billing records =
10872472-173611CC-81E9C73D-F836C2B6@172.18.192.19421:03:21: ****Adding to UAS Request
table
```

Sample Output for the debug ccsip info Command

This example shows only the portion of the debug output that shows that call redirection is disabled. When call redirection is enabled (default), there are no debug line changes.

```
Router# debug ccsip info

00:20:32: HandleUdpSocketReads :Msg enqueued for SPI with IPAddr: 172.18.207.10
:5060
00:20:32: CCSIP-SPI-CONTROL: act_sentininvite_new_message
00:20:32: CCSIP-SPI-CONTROL: sipSPICheckResponse
00:20:32: sip_stats_status_code
00:20:32: ccsip_get_code_class: !!Call Redirection feature is disabled on the GW
00:20:32: ccsip_map_call_redirect_responses: !!Mapping 302 response to 480
00:20:32: Roundtrip delay 4 milliseconds for method INVITE
```

Additional References

- “SIP Features Roadmap” on [page 1](#)—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- *Cisco IOS Debug Command Reference*, Release 12.3T at <http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123dbr/index.htm>
- *Cisco IOS Voice Troubleshooting and Monitoring Guide* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2 at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvfax_c/index.htm
- Cisco Technical Support at <http://www.cisco.com/en/US/support/index.html>

- *Troubleshooting and Debugging VoIP Call Basics* at http://www.cisco.com/warp/public/788/voip/voip_debugcalls.html
 - *VoIP Debug Commands* at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/1700/1750/1750voip/debug.htm
-

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