



SIP Features Roadmap

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

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.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
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.	“Configuring SIP Support for PAI” section on page 267 of 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.	“Configuring SIP Support for SDP” section on page 473 of this guide.
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.	“Configuring SIP Support for SRTP” section on page 483 of 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.	“Configuring Outbound Proxy Support for the SIP Gateway” section on page 264 of this guide.



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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.	SIP Features for Unified CME 4.1, Unified SRST 4.1, and Voice Gateways.
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.	SIP Features for Unified CME 4.1, Unified SRST 4.1, and Voice Gateways.
12.4(11)T	SIP Support for Hookflash	Configures IP Centrex supplementary services on SIP-enabled, Foreign Exchange Station (FXS) lines.	“Configuring SIP Support for Hookflash” section on page 497 of 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.	“Configuring SIP DTMF Features” section on page 445 of 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.	“Configuring SIP MWI Features” section on page 633 of 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.	“SIP KPML-Based Out-of-Band DTMF Relay” section on page 454 of 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.	“Configuring SIP Session Timer Support” section on page 248 of this guide.
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.	“SIP Gateway Support for SDP Session Information and Permit Hostname Command” section on page 190 of 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.	“Configuring SIP ISDN Features” section on page 521 of this guide.

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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.	“Configuring SIP Message, Timer, and Response Features” section on page 163 of 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.	“Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” on page 373 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP MWI Features” on page 633 of 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.	“Achieving SIP RFC Compliance” on page 67 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of this guide.
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.	“Configuring SIP AAA Features” on page 319 of 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.	“Configuring SIP ISDN Features” on page 521 of this guide.
12.3(4)T	ISDN Calling Name Display	Provides end-to-end calling name display in SIP networks.	“Configuring SIP ISDN Features” on page 521 of this guide.

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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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP Bind Features” on page 433 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP DTMF Features” on page 445 of 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.	“Basic SIP Configuration” on page 39 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of this guide.
12.3(4)T	SIP: RFC 3261 Enhancements (RFC 3261)	Provides compliance with RFC 3261.	“Achieving SIP RFC Compliance” on page 67 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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).	“Configuring SIP ISDN Features” on page 521 of 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.	“Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” on page 373 of this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
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.	“Configuring SIP QoS Features” on page 573 of this guide.
12.2(15)T	SIP: ISDN Suspend/Resume Support	Supports ISDN and ISDN User Part (ISUP) signaling basic functions, Suspend and Resume.	“Configuring SIP ISDN Features” on page 521 of 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.	“Configuring SIP Call-Transfer Features” on page 117 of 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.	“Configuring SIP QoS Features” on page 573 of this guide.
12.2(13)T	SIP - Call Transfer Enhancements Using the Refer Method	Enhances the Refer method for call transfer.	“Configuring SIP Call-Transfer Features” on page 117 of 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).	“Configuring SIP Connection-Oriented Media, Forking, and MLPP Features” on page 373 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of this guide.
12.2(13)T	SIP: Core SIP Technology Enhancements (RFC 2543)	Provides compliance with RFC 2543 and RFC 2543-bis-04.	“Achieving SIP RFC Compliance” on page 67 of 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.	“Configuring SIP DTMF Features” on page 445 of 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.	“Configuring SIP QoS Features” on page 573 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP AAA Features” on page 319 of this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
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.	“Configuring SIP Call-Transfer Features” on page 117 of 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.	“Configuring SIP ISDN Features” on page 521 of 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.	“Configuring SIP DTMF Features” on page 445 of this guide.
12.2(11)T	SIP Session Timer Support	Enables the periodical refresh of SIP sessions by sending repeated INVITE requests.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP AAA Features” on page 319 of 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.	“Configuring SIP DTMF Features” on page 445 of 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.	“Basic SIP Configuration” on page 39 of this guide.
12.2(8)T	SIP - DNS SRV RFC 2782 Compliance (RFC 2782)	Provides compliance with RFC 2782 in appending protocol labels.	“Achieving SIP RFC Compliance” on page 67 of 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.	“Configuring SIP AAA Features” on page 319 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP QoS Features” on page 573 of this guide.

Release (latest to earliest)	Features in That and Later Releases	Feature Description	Feature Documentation
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.	“Basic SIP Configuration” on page 39 of this guide.
12.2(8)T	SIP INVITE Request with Malformed Via Header	Allows the enabling of a response to an INVITE even if the Via header becomes malformed and cannot deliver the required information.	“Configuring SIP Message, Timer, and Response Features” on page 163 of 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.	“Configuring SIP Message, Timer, and Response Features” on page 163 of this guide.

