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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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 18*x* 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.

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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.

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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 *Cisco Unified Communications Manager Express System Administrator Guide*.
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 *Cisco Unified Communications Manager Express System Administrator Guide*.
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.

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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.

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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 3*xx* class of responses. Redirect processing is active by default, which means that SIP gateways handle incoming 3*xx* 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.

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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.

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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.

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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.