Feature Information for Cisco UBE (Enterprise) SIP Support

Table 1 lists the release history for this chapter.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which 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 software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Table 1 Feature Information for Cisco UBE (Enterprise) SIP Support Features

Feature Name	Releases	Feature Information
Transparent Tunneling of QSIG and Q.931 over SIP TDM Gateway and SIP-SIP Cisco Unified Border Element	Cisco IOS XE Release 3.1S	This 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. This feature uses no new or modified commands.
SIP-to-SIP Supplementary Services for Session Border Controller	Cisco IOS XE Release 3.1S	The SIP-to-SIP Supplementary Services for Session Border Controller feature enhances terminating and re-originating signaling and media between VoIP and Video networks This feature uses no new or modified commands.
SIP-to-SIP Basic Functionality for Session Border Controller	Cisco IOS XE Release 3.1S, Cisco IOS XE Release 3.3S	The SIP-to-SIP Basic Functionality for Session Border Controller (SBC) and for Cisco Unified Border Element (Cisco UBE) feature provides termination and reorigination of both signaling and media between VoIP and video networks using SIP signaling in conformance with RFC 3261. This feature uses no new or modified commands.
SIP-to-SIP Basic Functionality for Session Border Controller	Cisco IOS XE Release 3.1S	The SIP-to-SIP Basic Functionality for Session Border Controller (SBC) for Cisco Unified Border Element (Cisco UBE) feature provides termination and re-origination of both signaling and media between VoIP and video networks using SIP signaling in conformance with RFC 3261. This feature uses no new or modified commands.

Table 1 Feature Information for Cisco UBE (Enterprise) SIP Support Features (continued)

Feature Name	Releases	Feature Information
SIP—INFO Method for DTMF Tone Generation	Cisco IOS XE Release 2.5	The SIP—INFO Method for DTMF Tone Generation feature uses the Session Initiation Protocol (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 (UA) or proxy server. The SIP INFO message is sent along the signaling path of the call. The following command was introduced: **show sip-ua**.
SIP—Core SIP Technology Enhancements	Cisco IOS XE Release 2.5	Compliance to RFC 2543-bis-04 adds enhanced SIP support and ensures smooth interoperability and compatibility with multiple vendors. The following commands were modified: **debug ccsip messages, show sip-ua map**, **show sip-ua statistics**.
SIP Parameter Modification	Cisco IOS XE Release 2.5	Allows users to change the standard SIP messages sent from the Cisco SIP stack for better interworking with different SIP entities. This feature introduces or modifies the following commands: **voice class sip-profiles**, **voice-class sip profiles**
SIP History-info Header	Cisco IOS XE Release 3.1S	The SIP History-info feature provides the capability for the SIP TDM gateway to generate History-info messages in the INVITE dialog for calls that are forwarded or transferred. Cisco Unified Border Element platforms transparently pass the History-info across SIP legs. The receiving application uses the history-info header information to determine how and why the call has reached it. The following commands were introduced or modified: **history-info** and **voice-class sip history-info**
SIP Diversion Header Enhancements	Cisco IOS XE Release 3.1S	The SIP Diversion Header Enhancements feature enables time-division multiplex (TDM) gateways and Cisco Unified Communications Manager Express to populate the SIP Diversion Header with a domain name. This feature also provides choice of transparent pass through or application of address hiding to the SIP Diversion Header on Cisco UBE platforms. This feature modifies the following commands: **host-registrar,** and **registrar**
SIP 181 Call is Being Forwarded Message	Cisco IOSXE Release 3.1S	This feature allows users to configure support for SIP 181 Call is Being Forwarded messages either globally or on a specific dial-peer. This feature includes the following new or modified commands: **bock**, **voice-class sip block**, **map resp-code**, **voice-class sip map**.

Table 1 Feature Information for Cisco UBE (Enterprise) SIP Support Features (continued)

Feature Name	Releases	Feature Information
SIP - Session Timer Support	Cisco IOS XE Release 2.5	The SIP Session Timer Support feature adds the capability to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITEs, are sent during an active call leg to allow user agents (UAs) or proxies to determine the status of a SIP session. The following commands were introduced or modified: **min-se (SIP)** and **show sip-ua min-se**.
SIP - Enhanced 180 Provisional Response Handling	Cisco IOS XE Release 2.5	The Session Initiation Protocol (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 following commands were introduced or modified: **disable-early-media 180** and **show sip-ua status**.
Selective Filtering of Outgoing Provisional Response on the Cisco Unified Border Elements	Cisco IOS XE Release 3.1S	This feature adds support on Cisco UBE for selective filtering of outgoing provisional responses, including 180-Alerting, and 183-Session In Progress responses. Selective filtering can be further based on the availability of media information in the received provisional response. This feature introduces the following commands: **block** and **voice-class sip block**.
RFC 4040-Based Clear Channel Codec Negotiation for SIP Calls	Cisco IOS XE Release 3.1S	This feature adds support for RFC 4040-based clear channel codec Negotiation for SIP calls. The following commands were modified: **encap clear-channel standard** and **voice-class sip encap clear-channel**
Reporting End-of-Call Statistics in SIP BYE Message	Cisco IOS XE Release 3.3S	Allows users to send call statistics to remote ends when a call terminates. These statistics are sent as a new header in a BYE message or in the 200 OK message. The following commands were introduced or modified: **request**, **response**.

Table 1 Feature Information for Cisco UBE (Enterprise) SIP Support Features (continued)

Feature Name	Releases	Feature Information
PAID, PPID, Privacy, PCPID, and PAURI Headers on the Cisco Unified Border Element	Cisco IOS XE Release 3.1S	This feature enables Cisco UBE platforms to support: • P-Preferred Identity (PPID), P-Asserted Identity (PAID), Privacy, P-Called Party Identity (PCPID), in INVITE messages • Translation of PAID headers to PPID headers and vice versa • Translation of From: or RPID headers to PAID or PPID headers and vice versa • Configuration and/or pass through of privacy header values • PCPID header to route INVITE messages • Multiple PAURI headers in the response messages (200 OK) it receives to REGISTER messages • P-Preferred Identity and P-Asserted Identity Headers The following commands were introduced: **call-route p-called-party-id, privacy-policy, random-contact, random-request-uri validate, voice-class sip call-route p-called-party-id, voice-class sip privacy-policy, voice-class sip random-contact, and voice-class sip random-request-uri validate**.
Multiple Registrars on SIP Trunks on a Cisco Unified Border Element, on Cisco IOS SIP TDM Gateways, and on Cisco Unified Communications Manager Express feature	Cisco IOS XE Release 3.1S	This feature provides support for multiple registrars on SIP trunks on Cisco IOS SIP TDM gateways, Cisco Unified CME, and Cisco UBEs. This feature allows for a redundant registrar for each SIP trunk and enables registrar redundancy across multiple service providers. This feature includes the following new or modified commands: **credentials**, **localhost**, **registrar**, **voice-class sip localhost**.
Hiding the Internal Topology Information Embedded Within the History-info Header at the Cisco UBE	Cisco IOS XE Release 3.3S	This feature enables privacy across the enterprise domain by hiding internal topology information by stripping topology information from the history-info header. The following command was introduced or modified: **privacy policy, voice class sip privacy policy**.
Expires Timer Reset on Receiving or Sending SIP 183 Message	Cisco IOS XE Release 3.1S	This feature enables support for resetting the Expires timer upon receipt of SIP 183 messages on Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE). The following commands were modified in this feature: **reset timer expires** and **voice-class sip reset timer expires**
Configuring SIP Message, Timer, and Response Features	Cisco IOSXE Release 3.1S	This features allows SIP TDM gateway to generate History INFO message in the INVITE dialog for calls that are forwarded or transferred and Cisco UBE transparently passes History INFO across SIP legs.

Table 1 Feature Information for Cisco UBE (Enterprise) SIP Support Features (continued)

Feature Name	Releases	Feature Information
Configurable Hostname in Locally Generated SIP Header	Cisco IOS XE Release 2.5	This feature allows you to configure the hostname in locally generated SIP headers in global and dial-peer-specific configuration modes. The following commands were introduced or modified: **localhost dns** and **voice-class sip localhost dns**
Configurable Pass-through of SIP INVITE Parameters	Cisco IOS XE Release 3.1S	This feature enables the Cisco Unified Border Element platform to pass through end-to-end headers at a global or dial-peer level, that are not processed or understood in a SIP trunk to SIP trunk scenario. The pass through functionality includes all or only a configured list of unsupported or non-mandatory SIP headers, and all unsupported content/MIME types. This feature introduces the following commands: **pass-thru** and **voice-class sip pass-thru**.
Configuring an Error Response Code upon an Out-of-Dialog OPTIONS Ping Failure	Cisco IOS XE Release 3.1S	This feature provides option to configure the error response code when a dial peer is busied out because of an Out-of-Dialog OPTIONS ping failure. The following commands were introduced or modified in this release: **error-code-override options-keepalive failure**, **voice-class sip error-code-override options-keepalive failure**.
SIP-to-SIP Extended Feature Functionality for Session Border Controllers	Cisco IOS XE Release 3.1S, Cisco IOS XE Release 3.3S	The SIP-to-SIP Extended Feature Functionality for Session Border Controllers (SBCs) enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP user agents. The following commands were introduced or modified: **modem passthrough (dial-peer)**; **modem passthrough (voice-service)**; **show call active voice voice**; **show call history voice voice**; **show dial-peer voice**; **voice service** .
Call Routing logic on Cisco UBE using the History-info Header	Cisco IOS XE Release 3.3S	The call-routing logic on the Cisco UBE is enhanced by utilizing the information embedded in the history-info header of SIP requests that are retargeted or routed across domains. This is done by utilizing the information in the history-info header to bypass retargeting dial-peer entries that are indicated in the history-info header. The following command was introduced or modified: **call-route history-info, voice clas sip call-route history-info**.

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