Data Sheet

Cisco IOS Unified Communications Gateways with SIP

Cisco IOS® Unified Communications Gateways provide public-switched-telephone-network (PSTN), private-branch-exchange (PBX), and service-provider Session Initiation Protocol (SIP) trunk-gateway and interconnect capabilities. You can deploy these gateways to effectively meet today's unified communications networking needs and take advantage of emerging new applications.

The Cisco® Unified Communications System of voice and IP communications products and applications can enable your organization to communicate more effectively—by helping you streamline business processes, reach the right resource the first time, and increase profitability. The Cisco Unified Communications portfolio is an important part of the Cisco Business Communications Solution—an integrated solution for organizations of all sizes that also includes network infrastructure, security, and network management products; wireless connectivity; and a lifecycle services approach, along with flexible deployment and outsourced management options, end-user and partner financing packages, and third-party communications applications.

Cisco IOS Unified Communications Gateways provide media termination and signal translation between the PSTN and IP networks using the SIP signaling protocol for voice and video traffic. Applications include PBX interconnect, SIP trunking, IP Centrex, unified communications managed services, and residential voice. Cisco IOS Unified Communications Gateways adhere to IETF industry standards for SIP and are designed to work with Cisco call agents, third-party call agents, Cisco SIP applications, and third-party SIP applications. These gateways provide a complete platform for integration into branch-office, enterprise, campus, and service-provider networks, including configuration and management facilities, survivable failover and fall-back, session border controller (SBC), Call Admission Control (CAC), quality of service (QoS), security, conferencing, and transcoding. Cisco IOS Unified Communications Gateway routers also support H.323 and Media Gateway Control Protocol (MGCP) as well as H.323-to-SIP and SIP-to-SIP protocol interworking.

Cisco IOS Unified Communications Gateways support a wide range of time-division multiplexing (TDM) voice and video interfaces. TDM signaling support includes T1/E1 Primary Rate Interface (PRI), T1 channel associated signaling (CAS), E1-R2, T1/E1 QSIG Protocol, T1 Feature Group D (FGD), Basic Rate Interface (BRI), foreign exchange office (FXO), ear and mouth (E&M), and foreign exchange station (FXS). You can configure the following Cisco Unified Communications Gateways to support 2 to 720 channels:

- Cisco IAD2400 Series Integrated Access Devices
- Cisco VG224, VG204, and VG202 Analog Voice Gateways
- Cisco 1800 Series Integrated Services Routers
- Cisco 2800 and 3800 Series Integrated Services Routers
- Cisco 2900, 3900, and 3900E Series Integrated Services Routers
- Cisco AS5000 Series Universal Gateways
- Cisco VGD 1T3 Voice Gateway

Key Features and Benefits

The Cisco IOS Unified Communications Gateways offer the following advantages:

- **Industry-leading technology with SIP IETF standards:** Support for industry standards maximizes interoperability and protects investment. SIP provides the opportunity to bring together data, voice, and video in a single call and use a unified dial plan. Momentum has built around SIP as a signaling protocol used internally to the enterprise as well as for SIP trunk PSTN access that takes advantage of Internet technology and supports rapid application development for new and customizable applications.
- **QoS and CAC:** Features that you can use to help ensure voice quality include differentiated-services-code-point (DSCP) packet marking, IP Precedence, Low Latency Queuing (LLQ), Class-Based Weighted Fair Queuing (CBWFQ), IP service-level agreements (IP-SLAs), resource availability checks, and a range of CAC features, including support for call-counting and load-monitoring methods as well as bandwidth-based techniques such as Resource Reservation Protocol (RSVP).
- **Session border controller (SBC):** The Cisco Unified Border Element (UBE) available on Cisco Integrated Services Routers (ISRs) and Cisco AS5000 Series Universal Gateways provides a toolkit of SBC functions. These functions include interworking between SIP and H.323, topology hiding with address and port translations, billing and call-detail-record (CDR) normalization, network demarcation features, QoS and bandwidth management, rich signaling control using Tool Command Language (Tcl) and VoiceXML, media interworking for dual-tone multifrequency (DTMF) and codec translation and filtering, and a range of SIP trunk security features, including firewall and denial-of-service (DoS) protection.
- **Survivability:** If a connection is lost to the primary call agent, SIP proxy, or back-to-back user agent, the fall-back capability supports PSTN telephony interfaces on the branch-office router for the duration of the loss. You can combine this capability with Cisco Unified Survivable Remote Site Telephony (SRST) to enable IP phone endpoint call processing. Additionally, if a Cisco Unified Communications Manager fails over to a tertiary server, the Cisco IOS Unified Communications Gateway uses the next available server.
- **Multiprotocol support:** Cisco IOS Unified Communications Gateway routers support SIP, MGCP, and H.323. Support for multiple protocols maximizes flexibility in network design and simplifies protocol migrations. In addition, the Cisco Unified Border Element provides protocol translation between SIP and H.323 where needed.
- **Cisco Unified Communications Manager Express call processing:** Cisco Unified Communications Manager Express (UCME) embedded in Cisco IOS Software on the Cisco ISRs provides a rich set of voice- and video-processing features for small and medium-sized offices and branch offices. Cisco Unified Communications Manager Express delivers a low-cost, reliable, feature-rich unified communications solution—all within a single router platform.
- **ISR platform design:** A broad set of platforms, interface combinations, and features is available with the Cisco ISRs to address varying network design requirements. Enabling multiple features within a single ISR simplifies overall management and reduces costs.
- **Configuration and management:** Cisco IOS Unified Communications Gateways are configured using the familiar Cisco IOS Software command-line interface (CLI). A variety of management tools are available, including the CiscoWorks Family of products.

Figure 1 shows a variety of applications using SIP Cisco IOS Unified Communications Gateways, including Cisco Unified Communications Manager Express, IP Centrex, residential voice over broadband, PBX interconnect, Cisco Unified SRST, and enterprise SIP trunk.

Figure 1. SIP Network

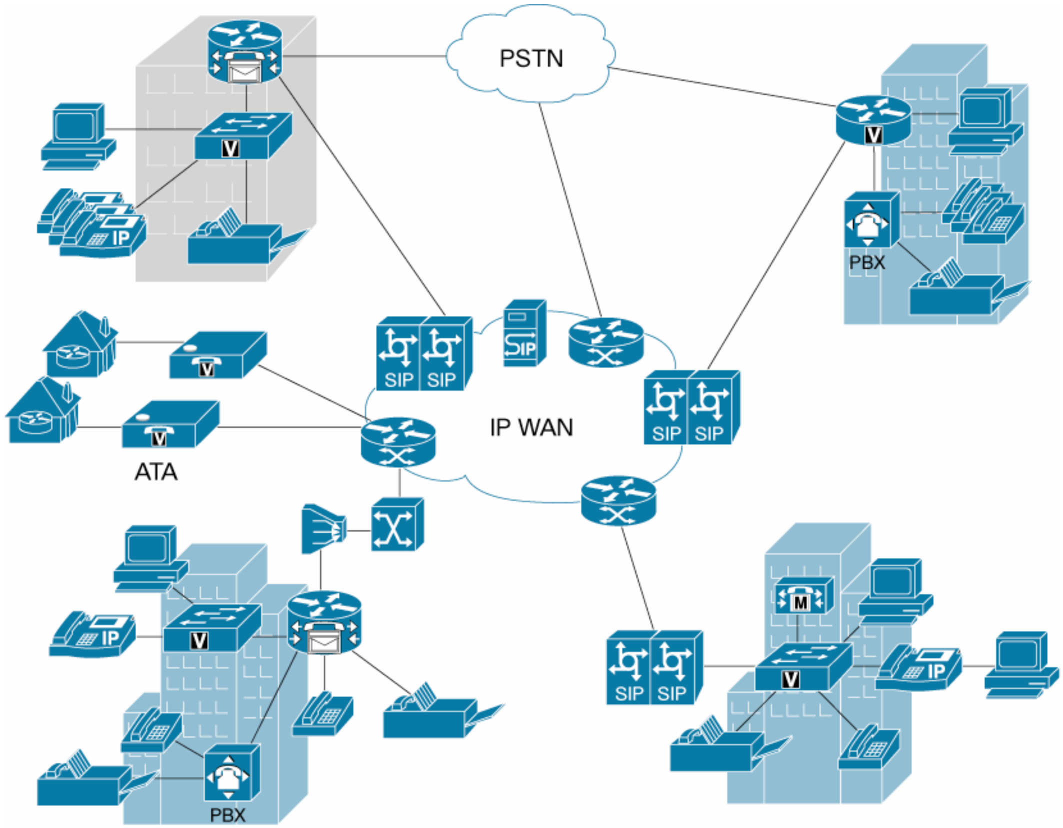

Table 1 lists the IETF standard features that are available in Cisco IOS Software on the Cisco IOS Unified Communications Gateways.

Table 1. **Standards Support**

IETF RFC	Features and Benefits	Cisco IOS Software Release
1889	The Real-Time Transport Protocol (RTP) supports real-time applications.	12.0M
2131 and 2132	The Cisco IOS Unified Communications Gateways support Dynamic Host Control Protocol (DHCP) FORCERENEW and use the DHCP framework to acquire SIP parameters from a DHCP server.	12.4(22)YB
2246	Transport Layer Security (TLS) provides signal encryption.	12.4(6)T
2327	The Session Description Protocol (SDP) defines the format for communication of information to set up a call.	12.2(11)T
2617	Basic and Digest Access Authentication (HTTP authentication) allows for challenge and response authentication of a user agent request.	12.4(4)T
2782	A Domain Name System Resource Record (DNS RR) for specifying the location of DNS services (DNS SRV) allows administrators to use several servers for a single domain, to move services from host to host, and to designate some hosts as primary servers for a service and others as backups.	12.2(8)T
2806	URLs for telephone calls (tel:URL) describe a terminal that can be contacted using the telephone network. The description includes the subscriber (telephone) number of the terminal and the necessary parameters to successfully connect to that terminal.	12.4M[1]
2833	RTP Payload for DTMF digits, telephony tones, and telephony signals describes how to carry DTMF signaling and other tone signals and telephony events in RTP packets.	12.2(8)T
2976	The SIP INFO Method is used to carry optional application-level information along the session signaling path.	12.4M
3182	The Cisco IOS Unified Communications Gateways support RSVP preconditions for audio and video.	12.4(22)T[2]
3203	The Cisco IOS Unified Communications Gateways support DHCP reconfigure extension.	12.4(22)YB
3204	The Cisco IOS Unified Communications Gateways support Multipurpose Internet Mail Extensions (MIME) types for application and QSIG objects for use in QSIG and tunneling through SIP.	12.4(20)T[2] 12.4(22)T[3]

3261	SIP is the standard (evolved from RFC 2543) application layer control (signaling) protocol for creating, modifying, and terminating sessions.	12.3(8)T[4]
3262	The Provisional Response Acknowledgement (PRACK) method provides reliable provisional response messages on the progress of request processing.	12.4M
3263	To locate SIP servers, SIP uses DNS procedures to allow a client to resolve a SIP Uniform Resource Identifier (URI) into the IP address, port, and transport protocol of the next hop to contact. It also uses DNS to allow a server to send a response to a backup client.	12.4M[5]
3264	The Offer/Answer Model defines a mechanism by which two entities can use SDP to arrive at a common view of a multimedia session between them. In the model, participant A offers participant B a description of the desired session from participant A's perspective, and participant B answers with the desired session from participant B's perspective.	12.4M
3265	Specific Event Notification provides a way for entities in the network to subscribe to resource or call state for various resources or calls in the network, and those entities (or entities acting on their behalf) can send notifications when those states change.	12.4(2)T
3311	The UPDATE Method allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no effect on the state of a dialog. This method is typically used to provide updated session information before a final response to the initial INVITE request is generated.	12.4M[2,11]
3312	The Cisco IOS Unified Communications Gateways support RSVP end-to-end preconditions for audio and video with symmetric QoS strength (mandatory or optional).	12.4(22)T[10,2]
3323	The Privacy header provides a mechanism to safeguard personal identity information.	12.4(15)T
3325	The Cisco IOS Unified Communications Gateways support private extensions to SIP for asserted identity within trusted networks.	12.4(15)T[2]
3515	The REFER Method allows the party sending the REFER to be notified of the outcome of the referenced request. You can use this method to enable many applications, including call transfer.	12.4(22)YB[3] 12.4M
3326	The Reason Header field defines a way to provide information about why a SIP request was issued.	12.4M
3361	The Cisco IOS Unified Communications Gateways support DHCP (IPv4) for SIP servers.	12.4(22)YB
3455	Private header (P-Header) extensions to SIP for the Third-Generation Partnership Project (3GPP) support P-Associated-URI and P-Called-Party-ID headers.	12.4(22)YB
3608	A SIP extension header field for service route discovery during registration architecture offers route support using the path header and service route.	12.4(22)YB[3]
3665	SIP Basic Call Flow is supported.	12.3(8)T
3666	Best-practice call flows for PSTN interworking are supported.	12.3(8)T
3711	The Cisco IOS Unified Communications Gateways support Secure RTP (SRTP) for media authentication and encryption on the unified communications router.	12.4(15)T
3725	Best practices for Third-Party Call Control (3PCC) are defined to allow one entity to set up and manage a communications relationship between two or more other parties.	12.4(22)YB[9]
3842	A Message Summary and Message Waiting Indication Event Package describes a SIP event package to carry message-waiting status and message summaries from a messaging system to an interested user agent.	12.3(8)T
3856	A Presence Event Package for SIP supports an idle status prompt with a customizable message and line-status subscription, which provides presence with authorization and authentication. Line-status subscription is for registered, in-service, idle, in-use, and busy.	12.4(4)T[6]
3863	The Cisco IOS Unified Communications Gateways support Presence Information Data Format (PIDF) as a common presence data format for Common Profile for Presence (CPP)-compliant Presence protocols. PIDF also defines a new media type "application/pidf+xml" to represent the XML MIME entity for PIDF.	12.4(15)T 12.4(11)XJ
3891	The Replaces header is used to logically replace an existing SIP dialog with a new SIP dialog. One use of the Replaces header is to replace one participant with another in a conversation.	12.4(15)T 12.4(11)XJ
3892	The Referred-By Mechanism allows the referrer to provide information about the REFER request to the refer target using the referee as an intermediary. This information includes the identity of the referrer and the URI to which the referrer referred. Call transfer is an example of where this mechanism can be used.	12.4(4)T[7]
4032	The Cisco IOS Unified Communications Gateways support this update to the SIP preconditions framework.	12.4(4)T[8]
4244	The Cisco IOS Unified Communications Gateways support this extension to SIP for request history information.	12.4(22)T[2]
4480	This extension of RFC 3863 adds optional elements to the Presence Information Data Format. These extensions provide additional information about the "presentity" and its contacts. The support is compliant with a pre-RFC draft: draft-ietf-simple-rpid-07.	12.4(22)T

4497	Best practices for interworking between SIP and QSIG through mapping between SIP and QSIG messages is supported.	12.4(15)XZ
Draft-levy-sip-diversion-06.txt	Diversion Indication provides the ability for the called SIP user agent to identify the person from whom the call was diverted and why it was diverted.	12.4(4)T
draft-ietf-sip-resource-priority-03	Communications Resource Priority headers are used to communicate and accept resource priority for SIP user agents, such as telephone gateways and IP telephones and SIP proxies.	12.4(4)T
draft-ietf-mmusic-sdescriptions-09.txt	The Cisco IOS Unified Communications Gateways support SDP Security descriptions for media streams. These descriptions define an SDP cryptographic attribute to signal SRTP on unicast media streams (supported on TDM-IP gateway).	12.4(4)T

[1] Tel:URL support only; Fax:URL and modem:URL are not supported
[2] Supported on TDM IP gateway router
[3] Supported on Cisco Unified Border Element
[4] TLS support introduced with 12.4(6)T
[5] A and SRV DNS records supported; NAPTR DNS records are not supported
[6] Message-summary supported; Msg-account, msg-summary-line, Account-URI, and opt-msg-headers are not supported
[7] Invite with Replace implementation is limited to transfer.
[8] Supported for call-transfer; 429 response is not supported
[9] Support for SRTP-to-RTP interworking on Cisco Unified Border Element
[10] Mid-call QoS negotiation does not use provisional responses, and new parameters take effect immediately after offer-answer, even when strength is mandatory.
[11] UPDATE without SDP is supported on Cisco Unified Border Element for Remote-Party-ID (RPID) update.

Cisco Unified Communications Services and Support

Using the Cisco Lifecycle Services approach, Cisco and our partners offer a broad portfolio of end-to-end services. These services are based on proven methodologies for deploying, operating, and optimizing IP communications solutions. Initial planning and design services, for example, can help you meet aggressive deployment schedules and minimize network disruption during implementation. Operate services reduce the risk of communications downtime with expert technical support, and Optimize services enhance solution performance for operational excellence. Cisco and our partners offer a system-level service and support approach that can help you create and maintain a resilient, converged network that meets your business needs.

Americas Headquarters
Cisco Systems, Inc.
San Jose, CA

Asia Pacific Headquarters
Cisco Systems (USA) Pte. Ltd.
Singapore

Europe Headquarters
Cisco Systems International BV
Amsterdam, The Netherlands

Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at www.cisco.com/go/offices.

Printed in USA C78-562664-01 03/10

