Data Sheet

Cisco IOS Voice Gateways with Session Initiation Protocol (SIP)

Cisco® IOS® voice gateways provide PSTN-to-SIP trunk gateway facilities. The gateways can be deployed to effectively meet today's voice networking needs and to take advantage of emerging new applications.

The Cisco Unified Communications system of voice and IP communications products and applications enables organizations to communicate more effectively—helping them to streamline business processes, reach the right resource the first time, and impact the top and bottom line. The Cisco Unified Communications portfolio is a key part of the Cisco Business Communications Solution—an integrated solution for organizations of all sizes which 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 voice gateways provide media termination and signal translation between the public switched telephone network (PSTN) and IP networks using the SIP signaling protocol. Applications include private branch exchange (PBX) interconnect, SIP trunking, IP Centrex, and residential voice. Cisco IOS voice gateways adhere to IETF industry standards and are designed to work with both Cisco call agents and third-party call agents. Cisco IOS voice gateways provide a complete platform for integration into branch, enterprise, and service provider networks, including configuration and management facilities, survivable failover and fall-back, session border controller, call admission control, quality of service (QoS), security, conferencing, and transcoding. Cisco IOS voice gateways routers also support H.323 and Media Gateway Control Protocol (MGCP) as well as protocol interworking.

Cisco IOS voice gateways support a wide range of packet telephony-based voice interfaces. 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), E&M, and foreign exchange station (FXS). The following Cisco voice gateway routers can be configured to support 2 to 2688 voice channels:

- Cisco 1700 Series modular access routers
- Cisco 2600 Series multiservice platforms
- Cisco 2800 Series integrated services routers
- Cisco 3700 Series multiservice access routers
- Cisco 3800 Series integrated services routers
- Cisco 5000 Series universal gateways

KEY FEATURES AND BENEFITS

The Cisco IOS voice 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 that takes advantage of Internet technology and supports rapid application development for new and customizable applications.
- **QoS and call admission control**—Features that can be used 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), Service Assurance Agent, Response Time Reporter, Resource Availability Checks, and Resource Reservation Protocol (RSVP).

- **Session border controller**—The Cisco Multiservice IP-to-IP Gateway available on Cisco IOS voice gateways provides a toolkit of session border controller functions. These include signal interworking between SIP and H.323, topology hiding with address and port translations, billing and call detail record normalization, QoS and bandwidth management, rich signaling using Tool Command Language (TCL) and Voice Extensible Markup Language (VXML), media interworking for dual tone multifrequency (DTMF) and codec translation, and security with firewall and denial of service protection.
- **Survivability**—If a connection is lost to the primary 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. This capability can be combined with Cisco Unified Survivable Remote Site Telephony to enable call processing. Additionally, in the case of a Cisco Unified CallManager failover to a tertiary server, the Cisco IOS voice gateway will use the next available server.
- **Multiprotocol support**—Cisco IOS voice 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 Multiservice IP-to-IP Gateway provides protocol translation between SIP and H.323 where needed.
- **Cisco Unified CallManager Express call processing**—Cisco Unified CallManager Express embedded in Cisco IOS Software on the Cisco integrated services and multiservice routers provides a rich set of call processing features for small and medium-sized offices and branch offices. Cisco Unified CallManager Express delivers a low-cost, reliable, feature-rich telephony solution all within a single router platform.
- **Integrated services router platform design**—A broad set of platforms, interface combinations, and features are available with the Cisco integrated services routers to address varying network design requirements. Enabling multiple features within a single integrated services router simplifies overall management and reduces costs.
- **Configuration and management**—Cisco IOS voice gateways are configured using the familiar Cisco 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 IOS voice gateways including, Cisco Unified CallManager Express, IP Centrex, residential voice over broadband, PBX interconnect, Cisco Unified Survivable Remote Site Telephony, and enterprise SIP trunk.

Figure 1. SIP Network

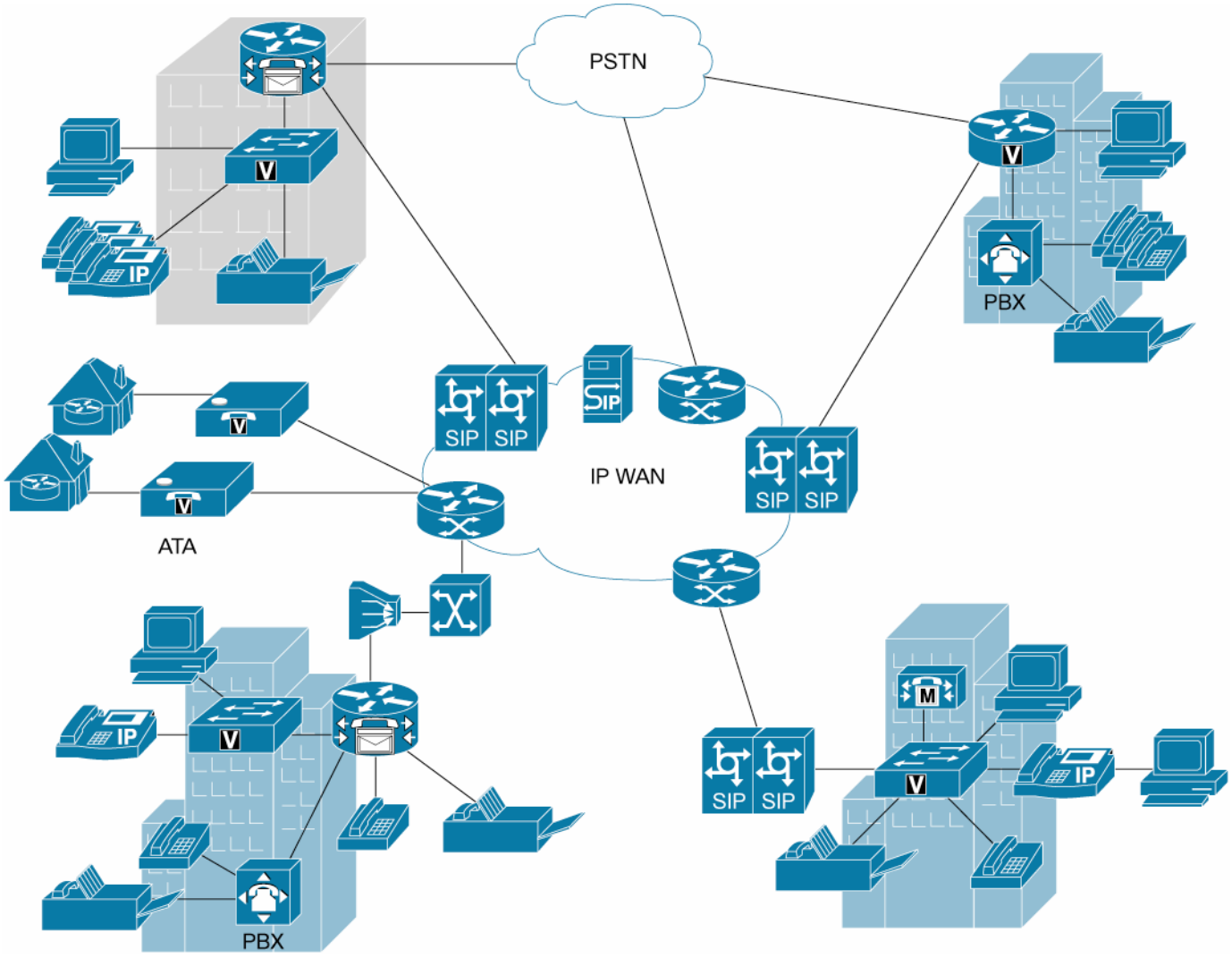

Table 1 lists the IETF standard features that are available in Cisco IOS Software on the Cisco IOS voice gateways.

Table 1. Standards Support

IETF RFC	Features and Benefits	Cisco IOS Software Release
1889	Real-Time Protocol (RTP) supports real-time applications.	12.0M
2246	Transport Layer Security (TLS) provides signal encryption.	12.4(6)T
2327	Session Description Protocol (SDP) defines the format for communication of information to set up a call.	12.2(11)T
2617	HTTP Authentication: Basic and Digest Access Authentication allows for challenge and response authentication of a User Agent request.	12.4(4)T

IETF RFC	Features and Benefits	Cisco IOS Software Release
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*
2833	RTP Payload for DTMF digits, telephony tones, and telephony signals describes how to carry dual-tone multifrequency (DTMF) signaling and other tone signals and telephony events in RTP packets	12.2(8)T
2976	SIP INFO Method is used to carry optional application-level information along the session signaling path.	12.4M
3261	SIP is the standard (evolved from RFC 2543) application-layer control (signaling) protocol for creating, modifying, and terminating sessions.	12.3(8)T**
3262	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***
3264	The Offer/Answer Model defines a mechanism by which two entities can make use of the Session Description Protocol (SDP) to arrive at a common view of a multimedia session between them. In the model, one participant offers the other a description of the desired session from their perspective, and the other participant answers with the desired session from their 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 impact on the state of a dialog. This is typically used to provide updated session information before a final response to the initial INVITE request is generated.	12.4M
3515	The REFER Method provides a mechanism allowing the party sending the REFER to be notified of the outcome of the referenced request. This can be used to enable many applications, including call transfer.	12.4M
3326	The Reason Header Field defines a way to provide information about why a SIP request was issued.	12.4M
3665	SIP Basic Call Flow examples are defined.	12.3(8)T
3666	Best-practice call flows are defined for PSTN interworking.	12.3(8)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.	12.3(8)T
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.	12.4(4)T****
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(4)T*****
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 can be used.	12.4(4)T******

IETF RFC	Features and Benefits	Cisco IOS Software Release
Draft-levy-sip-diversion-06.txt	The Diversion Indication provides the ability for the called SIP user agent to identify from whom the call was diverted and why the call 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

* Tel:URL support only. Fax:URL and modem:URL are not supported.

** TLS support introduced with 12.4(6)T.

*** A and SRV DNS records supported. NAPTR DNS records are not supported.

**** Message-summary is supported. Msg-account, msg-summary-line, Account-URI, and opt-msg-headers are not supported.

***** Invite with Replace implementation is limited to transfer.

****** Supported for call-transfer. 429 response is not supported.

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