



White Paper

Session Initiation Protocol Support in Cisco Unified Communications Products

Session Initiation Protocol (SIP) provides a standards-based approach to enabling IP communications for numerous devices and applications. This paper describes current and planned SIP support in the Cisco® Unified Communications system. Encompassing IP voice, data, and video communications products and applications, the Cisco Unified Communications system helps organizations communicate more effectively, streamline business processes, and positively affect financial results.

Cisco Systems® delivers SIP technology through a comprehensive roadmap that includes support for SIP RFCs (requests for comments) on a wide variety of Cisco products. The Cisco roadmap also integrates the SIP RFCs with the enhanced capabilities of the Cisco Unified Communications system. Organizations that adopt SIP-compliant Cisco products can reap the benefits of greater scalability, extensibility, and modularity for their communications today. They also establish a solid foundation for the unique and exciting capabilities associated with SIP technology as it extends in the future to additional communications types, devices, and applications.

WHY CONSIDER SIP?

Every year, technical innovation introduces an array of new devices with new addresses and methods for reaching people. In many cases, people find that the intended convenience of these new devices is compromised by the complexity of managing all these tools separately while juggling different—and perhaps overlapping—addresses, messages, and features among them. Access to new devices also raises questions about the most appropriate way to reach another person: “Should I send an e-mail or a voicemail? If voicemail, which one should I use? His office or cell phone voicemail?” Sharing communications among devices can be cumbersome, but a well-designed network based on a common signaling protocol can reduce this complexity substantially.

By implementing SIP technologies, enterprises can overcome these challenges and gain a powerful and flexible platform for creating innovative communications applications. In a SIP environment, all devices act as part of one system—users are reachable through a small number of addresses, and they can easily move communications from one device to another.

Defined by IETF (Internet Engineering Task Force) RFCs, SIP is a peer-to-peer, multimedia signaling protocol that integrates with other Internet services to deliver rich communications. SIP enables communication among compliant endpoints such as IP phones, desktop and notebook PCs, personal digital assistants (PDAs), and mobile phones. Systems that support SIP can enable innovative services, including Web-based communications; user mobility, presence, and preferences; and multiparty collaboration.

The benefits of SIP will become an integral part of IP communications systems and intelligent networks. Across a business or organization, SIP brings together many diverse applications, devices, and communications processes to deliver powerful new capabilities and features.

Implementing SIP technologies in a converged IP network offers numerous benefits, including increased value and user productivity from new and customizable applications, vendor independence for greater choice of applications and endpoints, and the potential to reduce costs for equipment and the management of communications services. Cisco is committed to supporting devices that have implemented standard SIP-based communications in order to offer customers maximum investment protection, interoperability, and deployment options.

CISCO EXPERTISE AND SUPPORT FOR SIP

Cisco Systems is uniquely positioned to take advantage of all capabilities offered by SIP. Cisco employees have been instrumental in defining SIP standards and have been at the forefront of developing SIP technology. Additionally, Cisco employees have authored or co-authored more SIP-related drafts and RFCs than employees of any other company. For in-depth background information about SIP and Cisco's involvement in developing the SIP standards, refer to the white paper, [SIP: The Next Step in Converged IP Communications](#).

With the largest SIP-enabled product line in the industry, Cisco is applying its own SIP experience to its enterprise solutions. The company is also working with other companies to help ensure their SIP-enabled applications integrate well with the Cisco infrastructure.

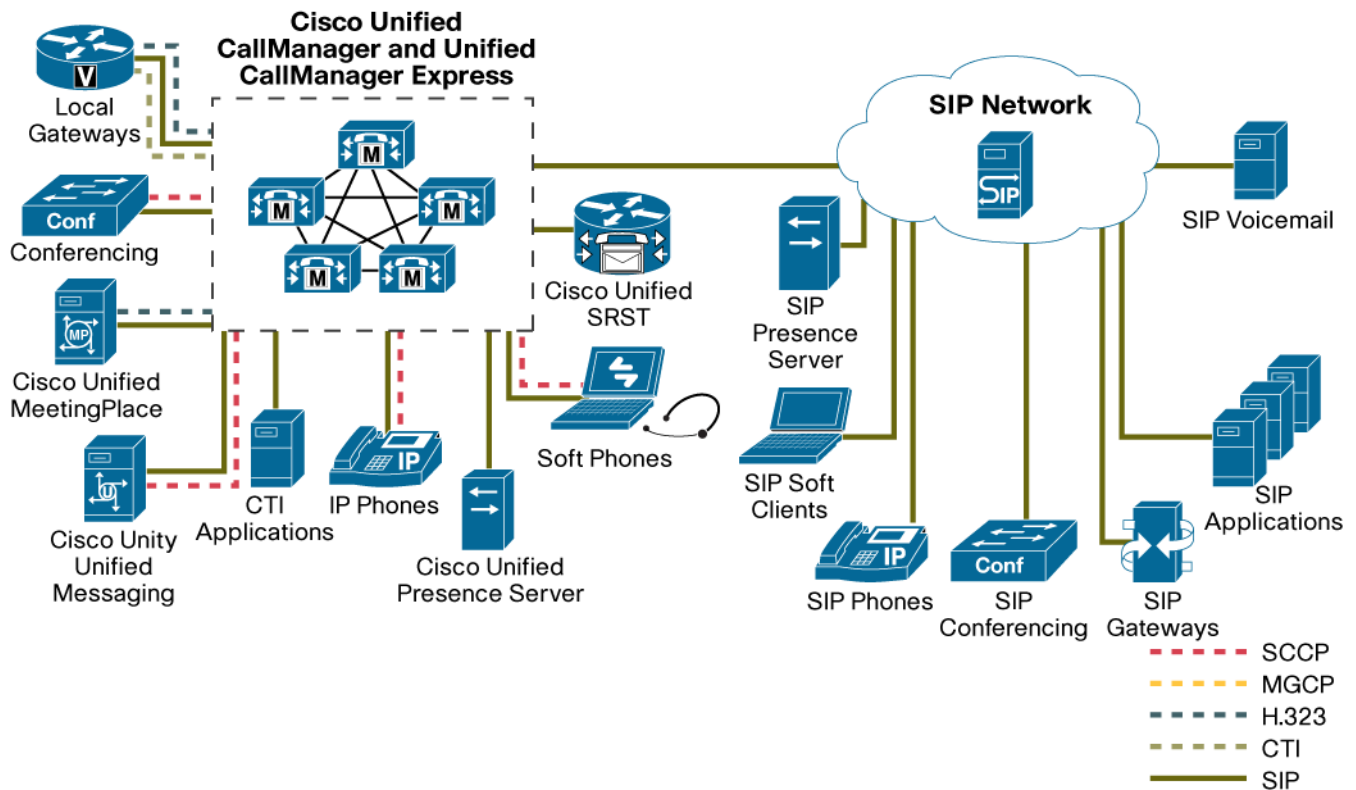
Customers can consider a migration to SIP with the confidence that Cisco has a proven strategy for integrating SIP technology and protecting customer investments in networks and communications systems.

The remainder of this document describes SIP features and support in the relevant products of the Cisco Unified Communications system.

CALL-PROCESSING SYSTEMS

Call-processing systems extend enterprise telephony features and functions to devices such as IP phones, media-processing devices, voice gateways, and multimedia applications. A call-processing system also integrates separate systems for unified messaging, video, and multimedia conferencing (Figure 1). The Cisco Unified Communications call-processing systems, gateways, session border controllers, and IP phones all support the SIP IETF RFC 3261 standard.

Figure 1. An Overview of Cisco SIP Support in an Enterprise Call-Processing Solution



Cisco Unified CallManager

Cisco Unified CallManager 5.0 provides a single call-processing solution with integrated, native SIP capabilities to support both call control and registration for SIP endpoints. Cisco Unified CallManager 5.0 offers native SIP protocol support for direct connection of SIP endpoints and SIP networks without additional equipment or software. The Cisco integration of SIP into the IP private-branch-exchange (PBX) environment is unique because it protects customer investments, supports a vast feature set for SIP endpoints, and allows customers to migrate to SIP applications and endpoints at their own pace.

Enterprises with a Cisco Unified CallManager can use Cisco Unified IP phones with either SIP or the company-developed Skinny Call Control Protocol (SCCP). These protocols coexist within the Cisco Unified CallManager environment, allowing gradual migration to SIP while protecting investments in existing devices. In addition to supporting SIP on numerous Cisco Unified IP Phone models, Cisco Unified CallManager supports registration and features for any endpoint compliant with RFC 3261.

The Cisco implementation of SIP is based on RFCs for implementing core telephony features. In order to deliver many of the SCCP-based features available today, the Cisco implementation supports additional SIP features, some of which are based on pending RFCs. The system also detects and communicates user presence information with the Cisco Unified Presence Server.

Customers will appreciate the consistency for using calling features delivered on SIP and SCCP phones. For example, SIP and SCCP phones can have appearances of the same extension. In addition, a call can be parked from a SIP phone and picked up from a SCCP phone—and conversely.

On the line side, SIP running on the Cisco Unified IP Phone models 7905G, 7911G, 7912G, 7940G, 7941G, 7941G-GE, 7960G, 7961G, 7961G-GE, 7970G, and 7971G-GE helps customers participate in systemwide, SIP-directed features such as presence and interoperability with third-party SIP services. In addition, line-side SIP support helps enable Cisco Unified CallManager 5.0 to provision, register, and manage specialty and third-party SIP phones.

The SIP trunk interface on Cisco Unified CallManager 5.0 is compliant with RFC 3261 and supports many other SIP RFCs for integrating a range of applications with voice and video calls over the SIP trunk. Interoperability on the Cisco Unified CallManager SIP trunk interface improves conferencing and application support experiences when used with Cisco Unity® Unified Messaging systems and Cisco Unified MeetingPlace® rich-media conferencing systems. By using SIP trunk-side and network-side interfaces, service providers can also deliver new SIP and presence-enabled services to their enterprise customers.

Because native SIP support is integrated into Cisco Unified CallManager, customers enjoy a single solution that delivers enterprise-class features and capabilities, increases interoperability, and improves support for innovation with SIP applications. Upgrades to the SIP capabilities in Cisco Unified CallManager 5.0 are available at no charge to current Cisco customers with the appropriate support contract.

For detailed product information, visit: <http://www.cisco.com/go/voice>

Cisco Unified CallManager Express

Cisco Unified CallManager Express supports the use of SIP trunks with either SCCP or SIP phones at local sites. SIP trunking helps customers use new SIP trunking services from service providers and take advantage of other services in the Cisco Unified Communications system.

Cisco Unified CallManager Express Version 3.4 operates as a registrar and back-to-back user agent (B2BUA), where SIP phones on the local network register to Cisco Unified CallManager Express. This capability allows network designs that support a broad range of deployment options. For example, the Cisco Unified CallManager Express sites can signal directly to other Cisco Unified CallManager Express sites or to a Cisco Unified CallManager site. Alternatively, Cisco Unified CallManager Express can signal to a higher-level SIP proxy server that provides directory calling features to simplify site-to-site dialing over an enterprise or service provider network.

A bulk registration with digest authentication capability simplifies administration and management for registrations from Cisco Unified CallManager Express to the higher-level SIP proxy server. The B2BUA provides supplementary services such as call transfer, hold, and forwarding.

Cisco has notable experience with SIP trunking, which provides benefits for service providers offering SIP-managed services and for organizations planning SIP migrations. The ability to use either SCCP or SIP endpoints at local sites provides customers with options to take advantage of innovations and capabilities in endpoints and applications as needed, with the assurance of interoperability for hybrid deployments and technology migrations.

The function Register to Network SIP Proxy with Digest Authentication, which is a secure authentication process to prevent rogue third-party phones from registering to the network SIP proxy, is also supported in Cisco Unified CallManager Express Version 3.4.

For detailed product information, visit: <http://www.cisco.com/go/voice>

Cisco RSVP Agent

The Cisco RSVP Agent feature in the Cisco IOS® Software uses the IETF standards-based Resource Reservation Protocol (RSVP) to provide Call Admission Control (CAC) and quality-of-service (QoS) features together with Cisco Unified CallManager Version 5.0. This feature helps Cisco Unified CallManager harness intelligence in the network to deliver networkwide CAC and QoS—capabilities that can significantly improve the end-to-end quality of communications services, including voice and video.

CAC and QoS also provide dynamic adaptability and resilience in changing network environments, meshed networks, and redundant network links. The physical and logical design of data, voice, and video is unified in the Cisco RSVP Agent. Because the RSVP CAC intelligence is placed in the network, there is no dependence on the end device.

Operating on all Cisco Integrated Services Router models, Cisco RSVP Agent supports endpoints using the SIP, H.323, Media Gateway Control Protocol (MGCP), and SCCP signaling protocols. Cisco leads the industry in supporting RSVP and in using this protocol to take advantage of network capabilities for enterprise call processing.

Cisco Unified Survivable Remote Site Telephony

The Cisco Unified Survivable Remote Site Telephony software operates on the Cisco Integrated Services Router models to provide communications failover, redundancy, and security capabilities for remote sites that connect to a central Cisco Unified CallManager system. With Cisco Unified Survivable Remote Site Telephony Version 3.4, the router can act as a temporary SIP call-processing service (B2BUA) for the duration of the WAN link failure.

In conjunction with Cisco Unified CallManager Version 5.0, customers can configure Cisco Unified Survivable Remote Site Telephony to provide backup telephony features (compliant with RFC 3261) for both SCCP and SIP phones at a remote site. For detailed product information, visit: <http://www.cisco.com/go/srst>

Cisco Emergency Responder

Cisco Emergency Responder dynamically identifies the location of 911 callers in an emergency and directs Cisco Unified CallManager to route emergency calls to the appropriate Public Safety Answering Point (PSAP) based on the caller's location. Location tracking supports Cisco SIP phones, SCCP, third-party SIP phones, and wireless IP phones. For smaller offices, Cisco Unified CallManager also supports E-911 services through configuration and connection to the public switched telephone network (PSTN).

In the future, Cisco Unified IP phones will learn their locations from the wired or wireless LAN infrastructure, and include their location information in SIP call signaling for emergency calls. Interconnect services will be available to connect SIP emergency calls to virtually any emergency call center—whether or not IP-enabled—and provide the associated location information. It will even become possible to track and report changes in the location of a wireless IP phone while an emergency call is in progress or after it has concluded.

For detailed product information, visit: <http://www.cisco.com/go/voice>

Cisco Voice Gateways

Cisco 1700, 2600XM, 2800, 3700, and 3800 and Cisco Catalyst® 6000 series voice gateway routers use SIP to provide media termination and signal translation between the PSTN and IP networks. Applications for this capability include PBX interconnect, SIP trunking, IP Centrex, and residential voice services. Providing comprehensive adherence to IETF standards, these Cisco SIP gateways are designed to work with call agents from Cisco and other vendors. Cisco voice gateways also support the H.323 protocol and MGCP as well as protocol interworking.

Cisco Session Border Controller

As customers build more complex voice and video networks, session border controllers (SBCs) are becoming a critical network component. A SBC allows customers to consolidate interconnect points, provide and enforce QoS metrics, interwork between protocols (for example, H.323 and SIP), and provide a more secure connection for intersite calling. The Cisco Multiservice IP-to-IP Gateway, the Cisco IOS Software Session Border Controller product, is used to extend SIP functions to enhance terminating and reoriginating call flows in conformance with SIP RFCs.

For customizing applications, the Cisco Multiservice IP-to-IP Gateway software supports Cisco Unified CallManager Version 5.0 functions as well as Tool Command Language (TCL) and voice Extensible Markup Language (VXML). These customized applications can overlay the signaling and media termination and reorigination functions for applications such as enforcing long-distance access codes.

In the future, the Cisco IOS Software Session Border Controller will offer new functions for Hosted Network Address Translation (NAT) Traversal to support an IP Address Hiding feature in all SIP messages without any configuration on remote gateways.

For detailed product information about the Cisco Multiservice IP-to-IP Gateway, visit: <http://www.cisco.com/go/voice>

Cisco Security Solutions

As a rendezvous protocol, SIP presents new challenges for network security because devices can communicate without any previously established security relationship. In addition, SIP deals with multiple intermediaries and endpoints with different trust policies.

SIP security features for products in the Cisco Unified Communications system are a natural extension of the Cisco Self-Defending Network strategy, which offers an integrated, collaborative, and adaptive approach to security that allows the network to respond to new threats as they arise. Today, security features in Cisco IOS Software, the Cisco PIX® security appliance, and security modules on Cisco switches and routers support protection for many voice-over-IP (VoIP) protocols, including SIP.

In addition, SIP-specific security features are supported on products in the Cisco Unified Communications system. SIP authentication is supported with the HTTP Digest Authentication feature on Cisco 1700, 2600XM, 2800, 3700, and 3800 and Cisco Catalyst 6000 series voice gateway routers, Cisco Unified CallManager, Cisco Unified CallManager Express, proxy servers, and SIP phones. The Cisco Remote-Party-ID (RPID) feature gives detailed descriptions of caller identity information in RPID headers.

Cisco Unified CallManager and Cisco Unified IP phones also support media and signaling encryption using secure Real-Time Transport Protocol (SRTP) and Transport Layer Security (TLS) for privacy, integrity, and confidentiality of voice conversations. Digitally signed IP phone images on Cisco Unified IP phones help prevent configuration tampering.

For a detailed white paper on security concerns relative to SIP, visit: <http://www.cisco.com/go/ipsecurity>

TELEPHONES AND COMMUNICATIONS DEVICES

SIP offers a standards-based means of connecting compatible phones and other communications devices to an enterprise telephone system. Users of these devices also benefit from the SIP presence capabilities, which help users define their own availability and determine the status of people they want to reach.

Cisco Unified IP Phones

The Cisco Unified IP Phone models 7905G, 7911G, 7912G, 7940G, 7941G, 7941G-GE, 7960G, 7961G, 7961G-GE, 7970G, and 7971G-GE are compliant with SIP RFCs. Cisco Unified IP phones support inline power, QoS, and security for both SCCP and SIP as well as automatic phone configuration and authentication.

Provisioning, registering, and managing Cisco Unified IP Phone models 7905G, 7912G, 7940G, and 7960G for SIP in Cisco Unified CallManager and Cisco Unified CallManager Express is as easy as managing these phones for SCCP. This simplicity protects existing phone investments as customers migrate to SIP.

For the enhanced Cisco Unified IP phones—models 7911G, 7941G, 7941G-GE, 7961G, 7961G-GE, 7970G, and 7971G-GE—the SIP user interface closely matches the SCCP interface, and feature parity will increase with future releases of Cisco Unified CallManager and Cisco Unified CallManager Express. Customers using the enhanced Cisco Unified IP phones will enjoy both investment protection and future-proofing as their SIP deployments evolve.

When selecting IP phones for use with Cisco Unified CallManager, customers should consider advanced call features, manageability, security, serviceability, and support as well as price. Cisco is committed to providing the best overall value in SIP-compatible phones. Cisco Unified IP phones that are compatible with SIP offer features defined by the standards, as well as additional capabilities developed by Cisco.

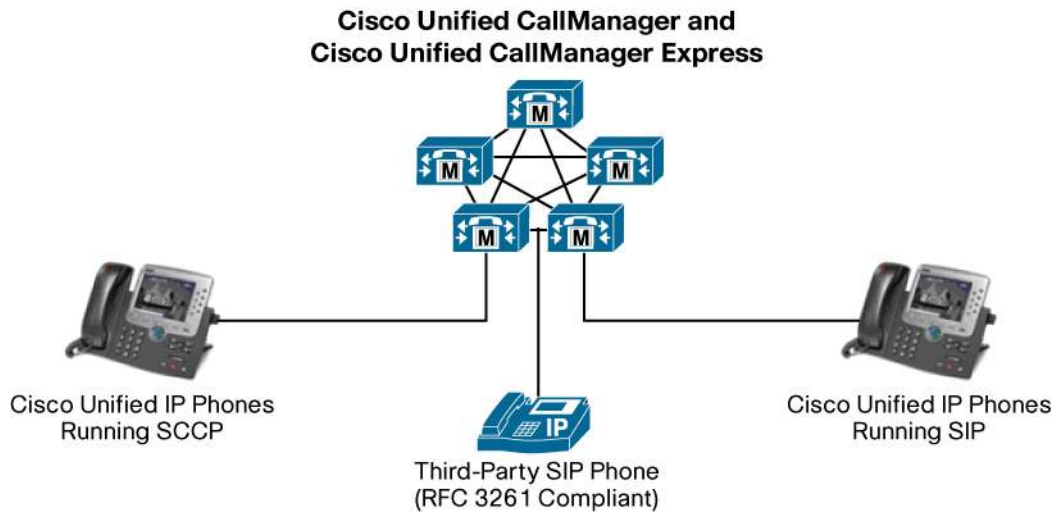
For detailed product information, visit: <http://www.cisco.com/go/ipphones>

Third-Party SIP Phones

Cisco Unified CallManager 5.0, Cisco Unified CallManager Express 3.4, and Cisco Unified Survivable Remote Site Telephony 3.4 support devices compliant with SIP RFCs (Figure 2).

Cisco will identify selected third-party SIP phones that have been accepted in the Cisco Technology Developer Program (refer to description later in this document). These phones will also go through external testing to certify interoperability with the Cisco Unified Communications system.

Figure 2. Cisco Support for Line-Side SIP Gives Customers Numerous Choices for Telephones



Cisco Unified Presence Server

The Cisco Unified Presence Server is a repository for collecting, aggregating, and communicating user capabilities and attributes to enable services such as presence-enabled directories, click-to-communicate, and personal communications preferences and management. These capabilities are made possible by taking advantage of industry-standard SIP and SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) technologies. Integrated with the Cisco Unified Communications system, the Cisco Unified Presence Server provides a standards-based peering environment for any SIP- or SIMPLE-enabled application or network.

The Cisco Unified Presence Server delivers both the SIP presence and SIP proxy function. The SIP Presence Engine provides the infrastructure for user and device status information such as busy, idle, away, presence engine, or available. It also stores information about the user's communications capabilities (for example, voice, video, and Web collaboration). Various communications applications and features, developed by Cisco and third parties, take advantage of this presence information to provide services that help users communicate with greater responsiveness and efficiency while still controlling their own accessibility and privacy (Figure 3).

The core SIP proxy function allows for efficient and accurate routing of presence and general SIP messages throughout the network and communications applications.

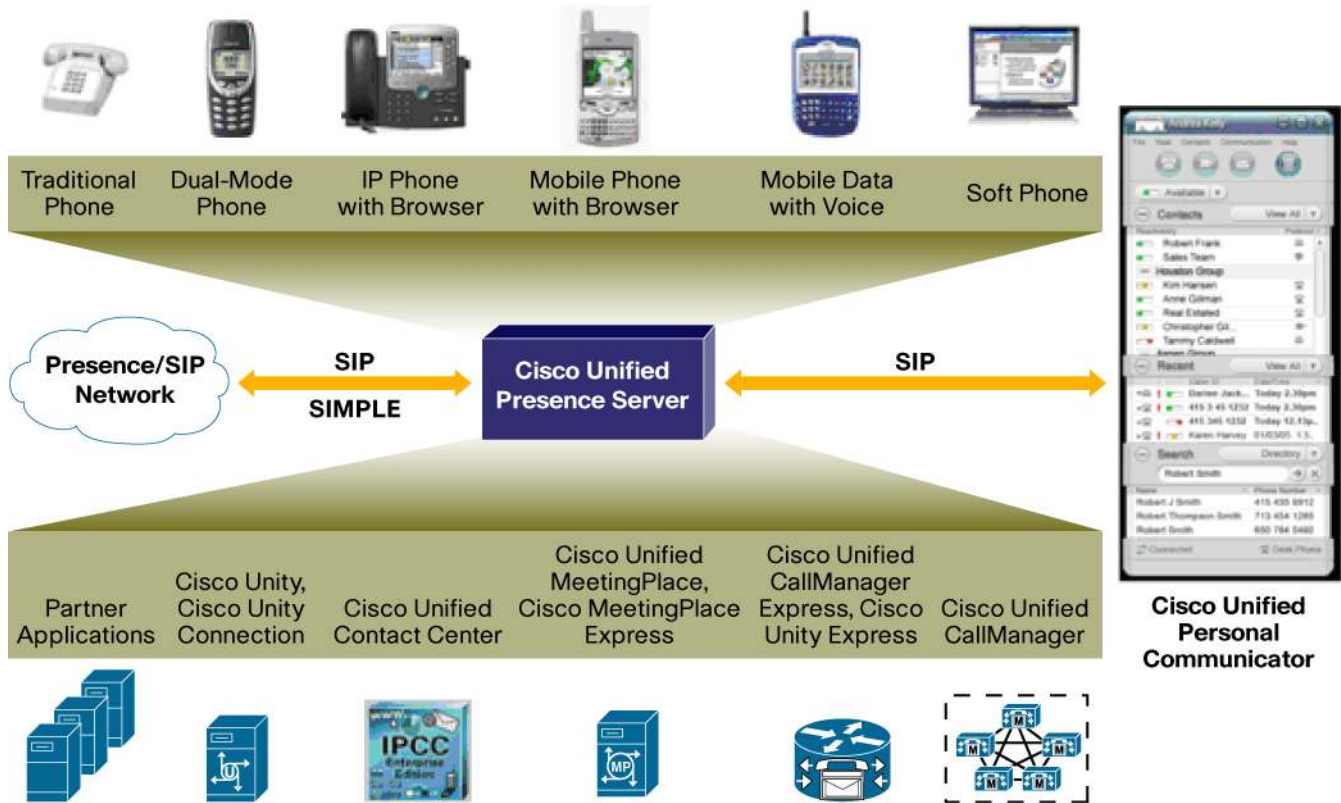
Cisco Unified Presence Server provides a common point of integration for all SIP/SIMPLE enabled application environments into the Cisco Unified Communications solution. Because the Cisco Unified Presence Server's provides a standard based interface, this allows for numerous desktop and end user applications to integrate presence from Cisco products into their services for a richer end user experience.

The Cisco Unified Presence Server also allows for interoperability with Microsoft Live Communications Server 2005 and the Microsoft Office Communicator client for any Cisco Unified IP phone connected to a Cisco Unified CallManager. This interoperability includes support for click-to-dial and phone-control capabilities, as well as presenting the telephony status of Cisco Unified IP phones within a Microsoft Office Communicator environment.

Many other communications vendors rely on an external product to provide presence information. In contrast, the integrated Cisco Unified Presence Server allows Cisco to offer more robust and diverse features for presence. The Cisco approach helps enable more valuable communications applications and services, whether they are developed by Cisco, the enterprise, or third parties.

For detailed product information, visit: <http://www.cisco.com/go/presenceserver>

Figure 3. Presence Services Help Users Control Their Communications Availability on Varied Devices While also Distributing That Status Information to Many Communications Systems and Applications



Cisco Unified Personal Communicator

The new Cisco Unified Personal Communicator software client transparently integrates a wide variety of rich communications applications and services, including voice, video, call management, presence, and Web conferencing. It operates as a SIP client on the Cisco Unified CallManager system.

Cisco Unified Personal Communicator uses SIP and SIMPLE standards to exchange presence information with the Cisco Unified Presence Server and uses SIP for audio and video soft phone call signaling. The presence information allows users to view the status and availability of colleagues from the Cisco Unified Personal Communicator client on a desktop computer.

For detailed product information, visit: <http://www.cisco.com/go/unifiedpersonalcomm>

User Location Features

Although presence indicates a person's availability to receive calls, it does not indicate that person's current physical location. For industries such as logistics, hospitality, transportation, and healthcare, user location data supplements presence information to help people and organizations communicate more efficiently.

In the Cisco Unified Communications system, location can be determined through the switch port accessed by a Cisco Unified IP phone, triangulation on an access point for wireless phones, or through global positioning system (GPS) data collected from supporting devices. The Cisco Wireless Location Appliance supports the triangulation capability for Cisco Aironet® lightweight access points, while GPS data can be obtained from the third-party solutions. In addition, Cisco Emergency Responder provides location data for both wired Cisco Unified IP phones

(using SIP or SCCP) and wireless access points. This location data can also be used by solutions, such as the Cisco Clinical Connection Suite, that are tailored for specific industries.

In the future, the Cisco Unified Communications system will provide a flexible framework to potentially use and integrate GPS information that may be received from endpoints, which will support additional business applications and integrated workflow processes.

Cisco Technology Developer Program

The Cisco Technology Developer Program helps enable select third-party products to integrate with Cisco solutions. This program extends to SIP applications and devices, including SIP phones that complement the Cisco product portfolio.

Although SIP is an open standard, many advanced features supported by Cisco Unified CallManager require extensions to SIP or other value-added technologies that will be available to third-party products only through the Cisco Technology Developer Program. Moreover, SIP is a large and complex set of standards, with many options and other areas open to interpretation.

Through the Cisco Technology Developer Program, the company will make available to third-party developers the know-how gained through years of experience in successfully implementing SIP for a wide range of products.

For more information about the Cisco Technology Developer Program, visit: <http://www.cisco.com/web/partners/pr46/tdp>. To find applications approved for this program, visit: <http://www.cisco.com/go/apps>

MESSAGING, CONFERENCING, AND VIDEO

Because SIP was designed to support multimedia environments, the SIP RFCs can be applied to easily improve integration of systems for unified messaging, multimedia conferencing, and video communications. With SIP, enterprises can more easily adapt multimedia services to facilitate and streamline everyday activity.

Cisco Unity, Cisco Unity Connection and Cisco Unity Express

Cisco Unity Unified Messaging systems (Version 4.0 and above) include a SIP interface that supports voice calls, message waiting indication, and other features for voicemail and unified messaging deployments with Cisco Unified CallManager 5.0 and Cisco Unified CallManager Express 4.0, delivering the same rich voicemail and unified messaging features that have traditionally been supported for SCCP phones.

Over the long term, Cisco Unity, Cisco Unity Connection, and Cisco Unity Express systems will take advantage of the SIMPLE extensions to SIP in order to access user presence information in a Cisco product-based network. Together with the Cisco Unified Presence Server and the Cisco Unified Personal Communicator, this presence information will give users easier and more natural access to messages.

For detailed product information, visit: <http://www.cisco.com/go/unity>, <http://www.cisco.com/go/unityconnection>, or <http://www.cisco.com/go/cue>

Cisco Unified MeetingPlace and Cisco Unified MeetingPlace Express Rich-Media Conferencing Solutions

Both Cisco Unified MeetingPlace and Cisco Unified MeetingPlace Express conferencing systems can connect SIP-enabled phones, from Cisco or third parties. Both systems support SIP trunking to a Cisco Unified CallManager, as well as direct inward dialing of SIP endpoints. Cisco Unified MeetingPlace Express supports SIP RFC 3261 for integration with Cisco Unified CallManager, Cisco Unified CallManager Express, and call control systems that support the same standard.

SIP support in the Cisco Unified MeetingPlace conferencing system helps enable a new feature for large enterprises: reservationless conferencing access to employees globally. This feature also helps enable a single dial-in access number for all conferences, regardless of the location of the employee or the Cisco Unified MeetingPlace server.

For detailed product information, visit <http://www.cisco.com/go/meetingplace> or <http://www.cisco.com/go/meetingplaceexpress>

Cisco Unified Videoconferencing

Cisco Unified Videoconferencing Software 3.5 and above delivers comprehensive support for SIP RFC 3261 for all video endpoints, including traditional videoconferencing room systems, Cisco Unified IP phones, and Cisco Unified Video Advantage. Each port on the Cisco Unified Videoconferencing Multipoint Control Unit (MCU) product is a dual-protocol port that provides a single dynamic pool of videoconferencing resources for all endpoints. This port design eliminates the need to segment videoconferencing resources between SIP and non-SIP endpoints.

The Cisco Unified Videoconferencing MCU provides investment protection for existing videoconferencing deployments by natively supporting H.323 and SIP on all ports, allowing newer SIP clients to connect with existing non-SIP videoconferencing endpoints (H.320, H.323, and SCCP) within the same videoconference. No additional licensing is required to implement SIP support on a Cisco Unified Videoconferencing MCU.

In the future, the Cisco Unified Videoconferencing solution will be enhanced with support for additional proxy capabilities, increased integration for Unified Communication and enhanced security.

More details about this product are available at: <http://www.cisco.com/go/ipvc>

MIGRATING TO SIP TECHNOLOGY

Adoption of SIP is a customer choice. With the Cisco Unified Communications system, customers have deployment flexibility and a complete choice of protocols, including SIP, H.323, MGCP, and SCCP.

For customers planning migrations to SIP, the migration process can be either gradual or a rapid cutover. For customers already using Cisco Unified Communications system products, migrating to SIP requires only an upgrade to the appropriate software release and reconfiguring devices and applications to use SIP. Cisco also offers the ability for customers to easily bridge networks with multiprotocol support in call-processing agents, and SIP-to-H.323 support in the Cisco Multiservice IP-to-IP Gateway.

Unlike alternative SIP solutions, Cisco customers will not need to replace systems or implement new hardware to take advantage of these technologies. For user applications, Cisco customers can choose when to deploy the Cisco Unified Presence Server and Cisco Unified Personal Communications client software to gain SIP-enabled presence capabilities.

CONCLUSION

SIP can be used to bring together many different applications, devices, and communications processes while also delivering new capabilities and features for users. The benefits of SIP for customer deployments will become integral to the Cisco Unified Communications system and Cisco product-based intelligent networks.

Cisco is leading the industry in both defining SIP capabilities and delivering advanced SIP products and technologies that are ready for deployment in enterprise networks. Cisco is committed to open SIP standards, which provide customers with maximum investment protection, interoperability, and choices for implementation. Most importantly, the Cisco implementation of SIP allows customers to determine their own strategies and timelines for migration.

**Corporate Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters

Cisco Systems, Inc.
168 Robinson Road
#28-01 Capital Tower
Singapore 068912
www.cisco.com
Tel: +65 6317 7777
Fax: +65 6317 7799

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