



## Product Overview

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## What is Session Initiation Protocol?

SIP is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points.

Like other VoIP protocols, SIP is designed to address the functions of signaling, device control, and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Device control* allows an intelligent entity to control a device with limited intelligence. *Session management* provides the ability to control the attributes of an end-to-end call.

SIP provides the capabilities to:

- Determine the location of the target end point—SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target end point—SIP determines the “lowest level” of common services between the end points. Conferences are established using only the media capabilities that can be supported by all end points.
- Determine the availability of the target end point—If a call cannot be completed because the target end point is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. It then returns a message indicating why the target end point was unavailable.
- Establish a session between the originating and target end point—If the call can be completed, SIP establishes a session between the end points. SIP also supports mid-call changes, such as the addition of another end point to the conference.
- Handle the transfer and termination of calls—SIP supports the transfer of calls from one end point to another. During a call transfer, SIP simply establishes a session between the transferee and a new end point (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

The conferences can consist of two or more users and can be established using multicast or multiple unicast sessions.

**Note**

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The term *conference* means an established session (or *call*) between two or more end points. In this document, the terms conference and call are used interchangeably.

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## Components of SIP

SIP is a peer-to-peer protocol. The peers in the sessions are called User Agents (UAs). These user agents can be grouped into two categories:

- User agent clients (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

Often a SIP end point will act as both a UAC and a UAS, one for sending requests the other for receiving.

From an architecture standpoint, the physical components of a SIP network can also be grouped into two categories: clients and servers. Figure 1-1 illustrates the architecture of a SIP network.

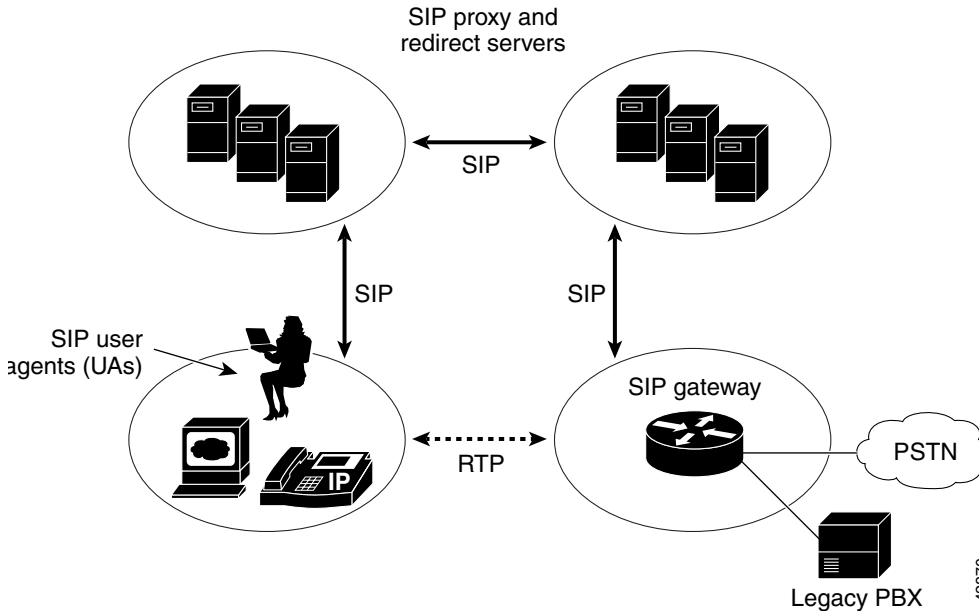
**Note**

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In addition, the SIP servers can interact with other application services, such as Lightweight Directory Access Protocol (LDAP), a database application, or an extensible markup language (XML) application. These application services provide back-end services such as directory, authentication, and billing services.

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Figure 1-1 SIP Architecture



## SIP Clients

SIP clients include:

- **Phones**—Can act as either a UAS or UAC. Softphones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.
- **Gateways**—Provide call control. Gateways provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway also translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side.
- **Media gateway controllers (MGC)**—Provide control of media and signaling gateways. Media gateways provide an interface between data networks and PSTN or ISDN networks.

## SIP Servers

SIP servers include:

- Proxy server—Acts as both a UAS and a UAC. The proxy server is an intermediate device that receives SIP requests from a client and then initiates requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, and security.
- Redirect server—Receives SIP requests, strips out the address in the request, checks its address tables for any other addresses that may be mapped to the one in the request, and then returns the results of the address mapping to the client. Basically, redirect servers provide the client with information about the path that a message should take and then the client contacts the appropriate SIP server directly.
- Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.
- Location server—Acts only as a UAS. The location server processes requests (typically from a redirect or proxy server) to provide information about the possible location of a target end user. Location servers can be co-located with a redirect or proxy server.

## What is the Cisco SIP IP Phone 7960?

Cisco SIP IP phones 7960s (hereafter referred to as Cisco SIP IP phones) are full-featured telephones that can be plugged directly into an IP network and used very much like a standard private branch exchange (PBX) telephone. The Cisco SIP IP phone is an IP telephony instrument that can be used in VoIP networks.

The Cisco SIP IP phone model terminals can attach to the existing in place data network infrastructure, via 10BaseT/100BaseT interfaces on an Ethernet switch. When used with a voice-capable Ethernet switch (one that understands Type of Service [ToS] bits and can prioritize VoIP traffic), the phones eliminate the need for a traditional proprietary telephone set and key system/PBX.

The Cisco SIP IP phone complies with RFC 2543.

Figure 1-2 illustrates physical features of the Cisco SIP IP phone:

**Figure 1-2 Cisco SIP IP Phone Physical Features**

- LCD screen—Displays information about your Cisco SIP IP phone, such as the number dialing out or calling in, online help, functions of soft keys and on-screen mode buttons, and so on.
- Line or speed dial buttons—Provides access to a new line on the screen.
- Footstand adjustment—Adjusts the angle of the phone base.
- Soft keys—Activates the feature described by the text message directly above on the LCD screen.
- Information (*i*) button—Provides online help for selected keys or features and network statistics about the active call. This feature will be available in a future release.

- On-screen mode buttons—Retrieves information about current settings, recent calls, available services, and voice mail messages. Only the settings button is available in this release of the Cisco SIP IP phone.
- Volume buttons—Adjusts the volume of the handset, headset, speaker, ringer and adjusts the brightness contrast settings on the LCD screen.
- Function toggles—Includes these options:
  - Headset and speaker—Toggles these functions enabling you to answer the phone using a headset or speakerphone.
  - Mute—Stops or resumes voice transmission.
- Scroll key—Enables you to move among different soft key options displayed on LCD screen.
- Dialing pad—Press the dial pad buttons to dial a phone number. Dial pad buttons work exactly like those on your existing telephone.
- Handset—Lift the handset and press the dial pad numbers to place a call, review voice mail messages, answer a call, and so on.

## Supported Features

In addition to the physical features illustrated in Figure 1-2, the Cisco SIP IP phone also provides the following:

- An adjustable ring tone
- A hearing-aid compatible handset
- Headset compatibility
- An integrated two-port Ethernet switch that allows the telephone and a computer to share a single Ethernet jack
- A direct connection to a 10BaseT or 100BaseT Ethernet (RJ-45) network (half- or full-duplex connections are supported)
- A large (4.25 x 3 in.) display with adjustable contrast
- G.711 (u-law and a-law) and G.729a audio compression
- IP address assignment—Dynamic Host Configuration Protocol (DHCP) client or manually configured via a local setup menu
- In-band dual-tone multifrequency (DTMF) support for touch-tone dialing

- Out-of-band DTMF signaling for codecs that do not transport the DTMF signaling correctly (for example, G.729 or G.729A)
- Local or remote (using the SIP 183 Ringing message) call progress tone
- Network startup via DHCP and Trivial File Transfer Protocol (TFTP)
- The following call options:
  - Call hold—Allows the Cisco SIP IP phone user (user A) to place a call (from user B) on hold. When user A places user B on hold, the 2-way RTP voice path between user A and user B is temporarily disconnected but the call session is still connected. When user A takes user B off hold, the 2-way RTP voice path is reestablished.
  - Call transfer—Allows the Cisco SIP IP phone user (user A) to transfer a call from one user (user B) to another user (user C). User A places user B on hold and calls user C. If user C accepts the transfer, a session is established between user B and user C and the session between user A and user B is terminated.
  - Secondary directory number—Allows the Cisco SIP IP phone to have an additional directory number that is logically assigned.
  - Call waiting—Plays an audible tone to indicate that an incoming call is waiting. The user can then put the existing call on-hold and accept the other call. The user can alternate between the two calls. If the user disconnects from one of the calls, they are automatically connected with the other call.
  - URL dialing—Provides the ability to place a call using an email address instead of a phone number.

**Note**

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For information on how to use the standard telephony features and URL dialing, refer to the *Getting Started Cisco IP Phone 7960* and *Quick Reference Cisco IP Phone 7960* documents that shipped with the phone.

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## Supported Protocols

The Cisco SIP IP phone supports the following standard network protocols:

- Internet Protocol (IP)

IP is a network layer protocol that sends datagram packets between nodes on the Internet. IP also provides features for addressing, type-of-service (ToS) specification, fragmentation and reassembly, and security.

The Cisco SIP IP phone supports IP as it is defined in RFC 791.

- User Datagram Protocol (UDP)

UDP is a simple protocol that exchanges data packets without acknowledgments or guaranteed delivery. SIP can use UDP as the underlying transport protocol. If UDP is used, retransmissions are used to ensure reliability.

The Cisco SIP IP phone supports UDP as it is defined in RFC 768 for SIP signaling.

- Internet Control Message Protocol (ICMP)

ICMP is a network layer Internet protocol that enables hosts to send error or control messages to other hosts. ICMP also provides other information relevant to IP packet processing.

The Cisco SIP supports ICMP as it is documented in RFC 792.

- Real-Time Transport Protocol (RTP)

RTP transports real-time data (such as voice data) over data networks. RTP also the ability to obtain Quality of Service (QoS) information.

The Cisco SIP IP phone supports RTP as a media channel.

- Trivial File Transfer Protocol (TFTP)

TFTP allows files to be transferred from one computer to another over a network.

The Cisco SIP IP phone uses TFTP to download configuration files and software updates.

- Dynamic Host Control Protocol (DHCP)

DHCP is used to dynamically allocate and assign IP addresses. DHCP allows you to move network devices from one subnet to another without administrative attention. If using DHCP, you can connect Cisco SIP IP phones to the network and become operational without having to manually assign an IP address and additional network parameters.

The Cisco SIP IP phone complies with the DHCP specifications documented in RFC 2131. By default, Cisco SIP IP phones are DHCP-enabled.
- Domain Name System (DNS)

DNS is used in the Internet for translating names of network nodes into addresses. SIP uses DNS to resolve the host names of end points to IP addresses.

## Prerequisites

For the Cisco SIP IP phone to successfully operate as a SIP endpoint in your network, your network must meet the following requirements:

- A working IP network is established.

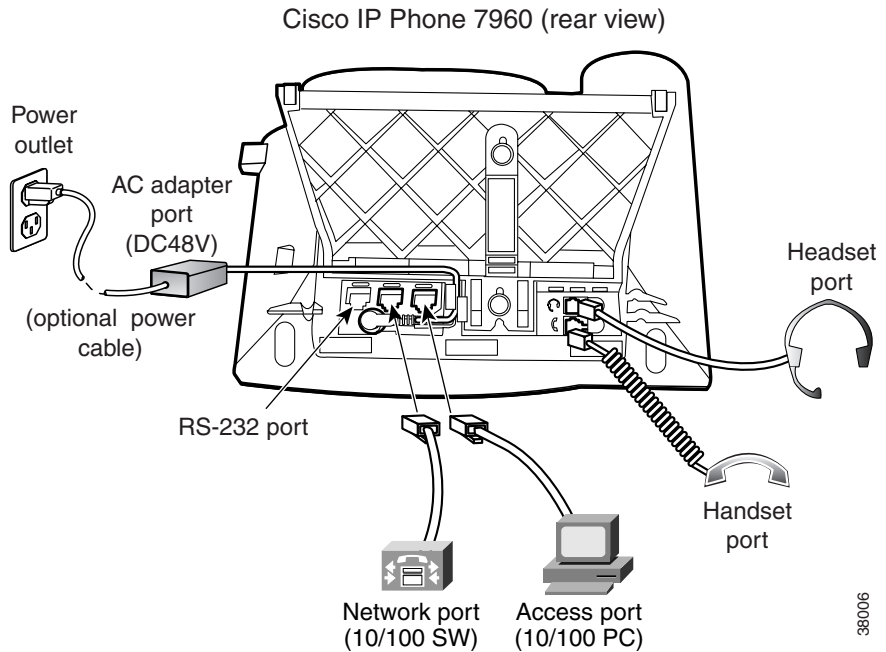
For more information about configuring IP, refer to *Cisco IOS IP and IP Routing Configuration Guide*.
- VoIP is configured on your Cisco routers.

For more information about configuring VoIP, refer to the *Cisco IOS Release 12.1 Multiservice Applications Configuration Guide* for the appropriate access platform. For more information about configuring SIP VoIP, refer to the *Enhancements to SIP for VoIP on Cisco Access Platforms*.
- VoIP gateways are configured for SIP.
- A TFTP server is active and contains the latest Cisco SIP IP phone firmware image in its root directory.
- A proxy server is active and configured to receive and forward SIP messages.

# Cisco SIP IP Phone Connections

The Cisco SIP IP phone has connections for connecting to the data network, for providing power to the phone, and for connecting a headset to the phone. Figure 1-3 illustrates the connections on the Cisco SIP IP phone.

**Figure 1-3 Cisco SIP IP Phone Cable Connections**



## Connecting to the Network

The Cisco SIP IP phone has two RJ-45 ports that each support 10/100 Mbps half- or full-duplex Ethernet connections to external devices—network port (labeled 10/100 SW) and access port (labeled 10/100 PC). You can use either Category 3 or 5 cabling for 10 Mbps connections, but use Category 5 for 100 Mbps connections. On both the network port and access port, use full-duplex mode to avoid collisions.

**Network Port (10/100 SW)**

Use the network port to connect the phone to the network. You must use a straight-through cable on this port. The phone can also obtain inline power from the Cisco Catalyst switch over this connection. See the “Connecting to Power” section on page 1-12 for details.

**Access Port (10/100 PC)**

Use the access port to connect a network device, such as a computer, to the phone. You must use a straight-through cable on this port.

## Connecting to Power

The Cisco SIP IP phone can be powered by the following sources:

- External power source—Optional Cisco AC adaptor and power cord for connecting to a standard wall receptacle.
- WS-X6348-RJ45V 10/100 switching module—Provides inline power to the Cisco SIP IP phone when connected to a Catalyst 3500, 4000, or 6000 family 10/100BaseTX switching module.

This module sends power on pins 1 & 2 and 3 & 6.

- WS-PWR-PANEL—Power patch panel provides power to the Cisco SIP IP phone which allows the Cisco SIP IP phone to be connected to existing Catalyst 4000, 5000, and 6000 family 10/100BaseTX switching modules.

This module sends power on pins 4, 5, 7, and 8.

**Note**

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Only the network port (labeled 10/100 SW) supports inline power from the Cisco Catalyst switches.

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For redundancy, you can use the Cisco AC adapter even if you are using inline power from the Cisco Catalyst switches. The Cisco SIP IP phone can share the power load being used from the inline power and external power source. If either the inline power or the external power goes down, the phone can switch entirely to the other power source.

To use this redundancy feature you *must* set the inline power mode to auto on the Cisco Catalyst switch. Next, connect the un-powered Cisco SIP IP phone to the network. After the phone powers up, connect the external power supply to the phone.

## Using a Headset

The Cisco SIP IP phone supports a four or six-wire headset jack. Specifically, the Cisco SIP IP phone supports the following Plantronics headset models:

- Tristar H81
- Encore Binaural H91
- Encore Monaural H101

The Volume and Mute controls will also adjust volume to the earpiece and mute the speech path of the headset. The headset activation key is located on the front of the Cisco SIP IP phone.

# The Cisco SIP IP Phone with a Catalyst Switch

To function in the IP telephony network, the Cisco SIP IP phone must be connected to a networking device, such as a Catalyst switch, to obtain network connectivity.

The Cisco SIP IP phone has an internal Ethernet switch, which enables it to switch traffic coming from the phone, access port, and the network port.

If a computer is connected to the access port, packets traveling to and from the computer and to and from the phone share the same physical link to the switch and the same port on the switch.

This configuration has these implications for the VLAN configuration on the network:

- The current VLANs might be configured on an IP subnet basis, and additional IP addresses might not be available to assign the phone to a port so that it belongs to the same subnet as other devices (PC) connected to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.

You can resolve these issues by isolating the voice traffic onto a separate VLAN on each of the ports connected to a phone. The switch port configured for connecting a phone would have separate VLANs configured for carrying:

- Voice traffic to and from the Cisco SIP IP phone (auxiliary VLAN)
- Data traffic to and from the PC connected to the switch through the access port of the Cisco SIP IP phone (native VLAN)

Isolating the phones on a separate, auxiliary VLAN increases the quality of the voice traffic and allows a large number of phones to be added to an existing network where there are not enough IP addresses.

For more information, refer to the documentation included with the Cisco Catalyst switch.